Control traffic overhead for VoIP over LTE

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I assure that I have done this work entirely on my own without any further assistance, except for the official support of the ComNets Research Group and KTH Royal Institute of Technology. All the literature used is listed in the bibliography.

Stockholm, July 10, 2012

(Syed Ghazanfar Salari)

ABSTRACT

With increasing technological advancements more sophisticated mobile devices are being used by end-users. Third generation (3G) mobile communication systems such as Universal Mobile Telecommunication System (UMTS) are not able to satisfy the rising demand for higher throughputs and low latencies. New standards based on Orthogonal Frequency Division Multiplexing (OFDM), such as Long Term Evolution (LTE) and Worldwide Interoperability for Microwave Access (WiMAX), have been proposed and are currently being integrated into existing mobile networks all over the world.

LTE specifications are being finalized within the 3rd Generation Partnership Project (3GPP) with the ambitious goals of increased spectral efficiency and end user throughput. Despite the introduction of several high data rate services, voice communication is still an essential part of the overall wireless wide area cellular communication market. In LTE, the core network is purely packet switched, thus voice is transmitted entirely using a Voice over Internet Protocol (VoIP). Like its predecessor standards it is desired that a large number of simultaneous VoIP calls be supported in LTE, while satisfying the desired Quality of Service (QoS) demands.

This thesis examines issues related to VoIP capacity for LTE. One of the key challenges is the limited number of schedulable voice packets per sub frame. The main goal of this thesis is to quantify the impact of this limitation. After describing basic LTE concepts, a detailed description of the control channel resource limitations for the scheduling of voice packets is presented. Consequences of these limitations are explained systematically by presenting the problem in a wider context.

Simulation results were obtained using the openWNS Simulator, an event driven system level simulation platform developed at the Communication Networks Research Group (ComNets), RWTH Aachen University Germany. Results are presented showing the impact of different scheduling strategies on VoIP capacity. These results illustrate how the limited control channel resources, specifically the Physical Downlink Control Channel (PDCCH) resources, affect the total number of schedulable VoIP user audio media streams.

SAMMANFATTNING

Med ökande tekniska framsteg mer avancerade mobila enheter som används av slutanvändarna. Tredje generationens (3G) mobila kommunikationssystem såsom Universal Mobile Telecommunication System (UMTS) inte kan tillgodose den ökande efterfrågan på högre genomströmning och låga latenser. Nya standarder som bygger på Orthogonal Frequency Division Multiplexing (OFDM), såsom Long Term Evolution (LTE) och Worldwide Interoperability för Microwave Access (WiMAX), har föreslagits och håller på att integreras I befintliga mobilnät över hela världen.

LTE specifikationer håller på att färdigställas inom 3rd Generation Partnership Project (3GPP) med de ambitiösa målen om ökad spektral effektivitet och slutanvändare genomströmning. Trots införandet av flera tjänster av hög datahastighet, är röstkommunikation fortfarande en väsentlig del av den totala Wireless Wide Area cellulär kommunikation marknaden. I LTE är kärnnätet rent paketförmedlande därmed röst överförs helt och hållet med hjälp av en Voice over Internet Protocol (VoIP). Precis som sina föregångare standarder är det önskvärt att ett stort antal samtidiga VoIP samtal få stöd i LTE, samtidigt som det uppfyller önskade Quality of Service (QoS) krav.

Denna avhandling undersöker frågor relaterade till VoIP kapacitet för LTE. En av de viktigaste utmaningarna är det begränsade antalet schemaläggningsbart röst paket per sub ram. Det huvudsakliga målet med denna avhandling är att kvantifiera effekterna av denna begränsning. Efter att ha beskrivit de grundläggande LTE begrepp, är en detaljerad beskrivning av de resurser kontroll kanal begränsningar för schemaläggning av röst paket presenteras. Konsekvenser av dessa begränsningar förklaras systematiskt genom att presentera problemet i ett större sammanhang.

Simulering resultat erhölls med hjälp av openWNS Simulator, en händelse driven systemnivå simulering som utvecklats vid Communication Networks Research Group (ComNets), RWTH Aachen University Tyskland. Resultat presenteras som visar effekterna av olika schemaläggning strategier för VoIP kapacitet. Dessa resultat illustrerar hur de begränsade kontroll kanalresurser, särskilt fysiskt Downlink (PDCCH) resurser, påverkar det totala antalet schemaläggningsbart VoIP användare ljud mediaströmmar.

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ABBREVIATIONS

3G Third generation
3GPP 3rd Generation Partnership Project
4G fourth generation
BLER Block Error Rate
BS Basestation
BSs Basestations
C-RNTI Cell-RNTI
CCEs Control Channel Elements
CEUSE Cell Edge User Spectral Efficiency
CFI Control Format Indicator
CN Core Network
ComNets Communication Networks Research Group
CQI Channel Quality Indicator
CSE Cell Spectral Efficiency
DCI Downlink Control Information
DFT Discrete Fourier Transform
DL Downlink
DMRS DeModulation Reference Signals
DwPTS Downlink Pilot Time Slot
E-UTRAN Enhanced UMTS Terrestrial Radio Access Network
eNodeBs Evolved Node B
EPC Evolved Packet Core
EPS Evolved Packet System
ESM Effective SINR Mapping
FDD Frequency Devision Duplexing
GP Guard Period
HARQ Hybrid Automatic Repeat Request
IMT-A International Mobile Telecommunication Advanced
InH Indoor Hotspot Scenario
IR Incremental Redundancy
ITU-R International Telecommunication Union Radiocommunication Sector
IP Internet Protocol
ISO International Organization for Standardization
L2S Link-to-System
LQM Link Quality Metric
LTE Long Term Evolution
MCS Modulation and Coding Schemes
MIESM Mutual Information Effective SINR Mapping
MME Mobility Management Entity
MMIB Mean Mutual Information per Bit
OFDM Orthogonal Frequency Division Multiplexing
OFDMA Orthogonal Frequency Division Multiple Access
openWNS open source Wireless Network Simulator
OSI Open System Interconnections
P-GW Packet Data Network Gateway

P-RNTI Paging-RNTI
PAR Peak to Average Ratio
PCFICH Physical Control Format Indicator Channel
PDCCH Physical Downlink Control Channel
PDSCH Physical Downlink Shared Channel
PDN Packet Data Network
PER Packet Error Rate
PHICH Physical Hybrid ARQ Indicator Channel
QoS Quality of Service
RA-RNTI Random Access-RNTI
RB Resource Block
RBP Resource Block Pair
RBIR Received Bit Information Rate
RE Resource Element
REs Resource Elements
REGs Resource Element Groups
RNTI Radio Network Temporary Identifier
RRC Radio Resource Control
S-GW Serving Gateway
SC-FDMA Single-Carrier Frequency-Division Multiple Access
SI-RNTI System Information-RNTI
SINR Signal-to-Interference-plus-Noise Ratio
SPS C-RNTI Semi Persistent Scheduling C-RNTI
TB Transport Block
TC-RNTI Temporary C-RNTI
TDD Time Division Duplexing
TD-SCDMA Time Division Synchronous Code Division Multiple Access
TPC transmit power control
TTI Transmission Time Interval
VRBs Virtual Resource Blocks
VoIP Voice over Internet Protocol
UL Uplink
UMTS Universal Mobile Telecommunication System
UpPTS Uplink Pilot Time Slot
UT User Terminal
WiMAX Worldwide Interoperability for Microwave Access

CHAPTER 1

Introduction

This chapter gives a brief overview of this Master's thesis. It describes the motivation behind this project and highlights the major problems addressed during the course of this project. Finally, the structure of this thesis is presented in Section 1.3.

1.1 Motivation

Telephony has always been the most important of all personal mobile communication services, contributing a major portion of the overall revenue of a telephony operator. An efficient telephony service supporting a large number of simultaneous calls remains a very desirable objective for mobile network operators. Moreover, the introduction of Long Term Evolution (LTE) has further raised the performance bar. LTE has been standardized within the 3rd Generation Partnership Project (3GPP) and is optimized for packet data transfer [27]. Furthermore, the core network is now solely packet switched, which means that speech is transmitted entirely through Voice over Internet Protocol (VoIP), also known as IP telephony. The future scope of VoIP is emphasized by market research studies [2] showing:

- Mobile VoIP users will increase tenfold over the next five years
- By 2013, VoIP penetration among businesses in the USA will reach 79%
- By 2014, mobile VoIP users will increase to nearly 139 million
- By 2015, annual business mobile VoIP gateway revenues will exceed 6 billion U.S. Dollars
- By 2015, annual worldwide smart phone shipments will be nearly 1 billion and IP phone shipments will surpass 40 Million

In the light of all the above predictions, the investigation of VoIP capacity in LTE is a very hot topic of research. This investigation is the focus of this thesis.

1.2 Problem statement

This thesis primarily focuses on the evaluation of LTE's VoIP capacity and addresses the following problems:

- Scheduling of large number of potentially simultaneous VoIP calls - each producing a stream of audio packets
- Smart utilization of radio control channel resources for each call
- Signal-to-Interference-plus-Noise Ratio (SINR) dependent radio resource management for both Uplink (UL) and Downlink (DL)
- Evaluation of different radio resource scheduling strategies

1.3 Structure

In Chapter 2, key features of LTE and LTE-Advanced are summarized to provide the necessary theoretical background to facilitate the reader's understanding of the following chapters. Chapter 3 deals with the role of radio control channel resources in the radio resource scheduling of voice packets by connecting the concepts described in Chapter 2 to the details of the Physical Downlink Control Channel (PDCCH). In Chapter 4, the essential concepts which play a central role in the scheduling process such as link adaptation, fitting strategies, and different radio resource scheduling strategies are described. Chapter 5 starts with a description of the open source Wireless Network Simulator (openWNS) simulator, followed by an explanation of the implemented VoIP scheduler. Chapter 6 presents the simulation results with regard to performance parameters, including user satisfaction, number of cut-off calls, allocation of aggregation formats, and the efficiency of different scheduling strategies. Finally, the thesis concludes with Chapter 7 which draws a conclusion and suggests directions for future research.

CHAPTER 2

LTE overview

This chapter presents the main features of LTE and LTE-Advanced. Moreover, some fundamental concepts are described which are necessary theoretical background for understanding the subsequent chapters. In particular section 2.2 describes the control channel region of each sub-frame. Understanding of this portion of the sub-frame is essential to understand the Physical Downlink Control Channel (PDCCH) described in Chapter 3.

2.1 LTE standard

LTE is a big step forward in the advancing series of mobile communication systems standardized within 3GPP.

2.1.1 3rd Generation Partnership Project

3GPP is a association of six telecommunication standards bodies, known as organizational partners. These organizational partners are from Asia, Europe, and North America. The aim of 3GPP is to generate a series of reports and specifications that define 3GPP technologies and that promote the successful evolution of the standards which began with GSM. The main areas covered in this standardization are the core network, the radio network, and the service architecture. These 3GPP standards are published in the form of releases with Release 10 being the latest one. Figure 2.1 shows the evolution of 3GPP standardization over time.

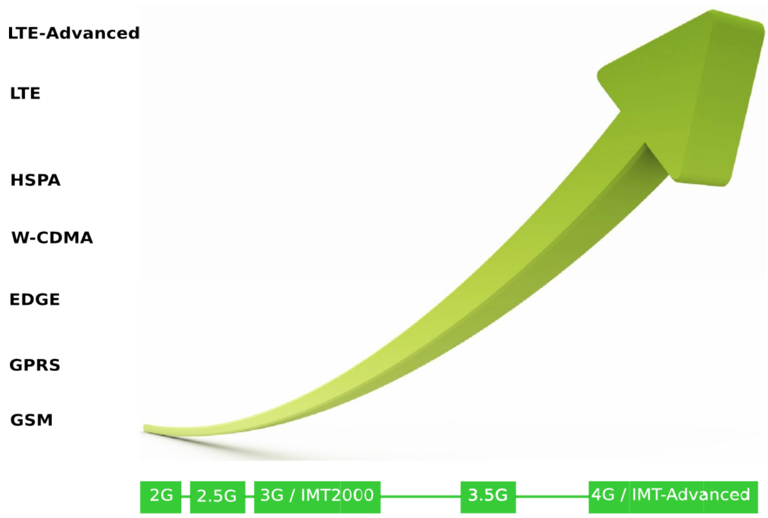


Figure 2.1: Evolution of 3GPP standardization [1]

2.1.2 Performance requirements

Enhanced system performance compared to the currently deployed systems is the most important feature of LTE and is the key reason for LTE attracting market interest. LTE is expected to offer a significantly improved user experience when transferring large amounts of data as compared to a transfer via a Third generation (3G) mobile communication systems. Some of the main performance metrics of LTE are summarized in Table 2.1.

Table 2.1: Main performance targets for LTE (data take from table 1.1 on page 8 of Chapter 1: Introduction and Background by Thomas Salzer and Matthew Baker in [29])

	Parameters	Requirements
Downlink	Peak transmission rate	> 100 Mbps
	Peak spectral efficiency	> 5 bps / Hz
	Average cell spectral efficiency	> 1.6 - 2.1 bps / Hz / user
	Cell edge spectral efficiency	> 0.04 - 0.06 bps / Hz / user
	Broadcast spectral efficiency	> 1 bps / Hz
Uplink	Peak transmission rate	> 50 Mbps
	Peak spectral efficiency	> 2.5 bps / Hz
	Average cell spectral efficiency	> 0.66 - 1.0 bps / Hz / user
	Cell edge spectral efficiency	> 0.02 - 0.03 bps / Hz / user
System	User plane latency (two way radio delay)	< 10 ms
	Connection set-up latency	< 100 ms
	Operating bandwidth	1.4 - 20 MHz
	VoIP capacity	Next Generation Target Mobile Networks (NGMN) goal expressed is > 60 sessions / MHz / cell

2.1.3 Architectural overview

In contrast to its predecessor technologies, LTE is optimized for packet data transfer only, in order to provide seamless Internet Protocol (IP) connectivity between a User Terminal (UT) and a Packet Data Network (PDN) while supporting terminal mobility. The LTE standard defines the Evolved Packet System (EPS) consisting of a radio access network, also known as the Enhanced UMTS Terrestrial Radio Access Network (E-UTRAN) encompassing the radio aspects and the Core Network (CN), called the Evolved Packet Core (EPC) that deals with non-radio aspects of the LTE system [5]. The LTE system is comprised of three key components: the EPC, the E-UTRAN, and the UT. The IP Connectivity Layer is depicted in Figure 2.2. The name of this layer is based upon the fact that the top of the protocol stack shown in Figure 2.2 on the UT is located just below the IP layer.

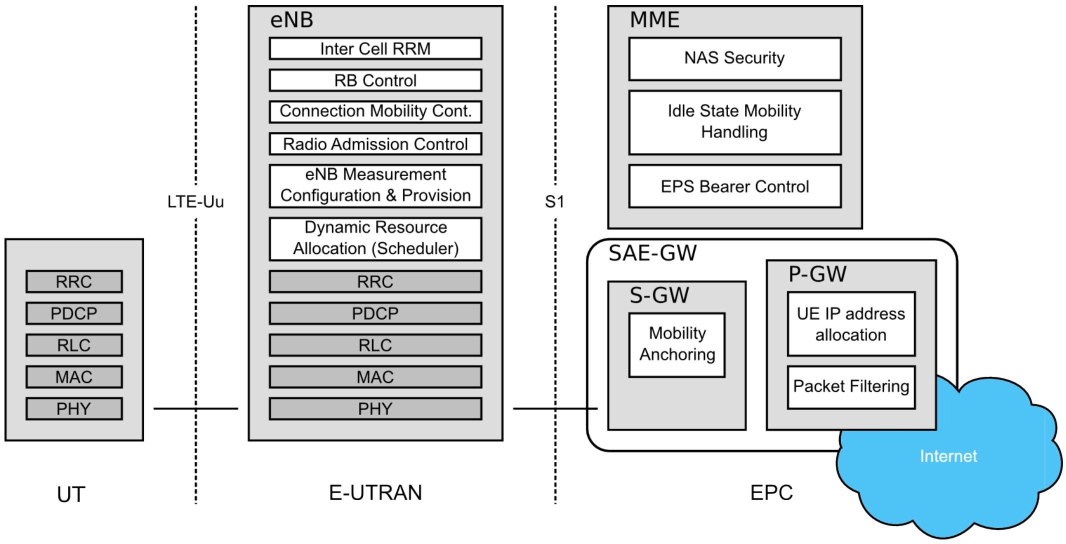


Figure 2.2: Functional split among E-UTRAN, EPC and UT [5]

2.1.3.1 Evolved Packet Core

The Evolved Packet Core (EPC) is responsible for the overall control of the UT and data bearers. Bearers are used by EPS to route IP traffic from the PDN to the UT. The main logical nodes of the EPC [23] are:

Packet Data Network Gateway (P-GW): The P-GW provides connectivity between UTs and external packet data networks by acting as the entry and exit point for UT traffic. The P-GW is also responsible for flow-based charging, packet filtering and screening, lawful interception, and policy enforcement. The P-GW is a user-plane node.

Serving Gateway (S-GW): The S-GW is responsible for routing and forwarding of user data traffic. User data is buffered at the S-GW while the UT connects to another eNB or during the paging process. Replication of user traffic is also done by the S-GW in the case of lawful interception. The S-GW is a user-plane node.

Mobility Management Entity (MME): The MME is the only control-plane node in the EPC and it handles the signalling between UT and CN using Non Access Stratum (NAS) protocols. Other functions of the MME include bearer and connection management.

2.1.3.2 Enhanced UMTS Terrestrial Radio Access Network

The access network, E-UTRAN is comprised of a network of Evolved Node B (eNodeBs). The E-UTRAN architecture is flat due to the absence of a centralized controller for unicast traffic (while there may be a centralized controller in the case of broadcast traffic). The eNodeBs are interconnected by the X2 interface and they are connected to the MME and S-GW by the S1-MME and S1-U interfaces respectively. The E-UTRAN is responsible for radio resource management, header compression, security, and connectivity to the EPC. Figure 2.3 illustrates the architecture of the E-UTRAN in detail.

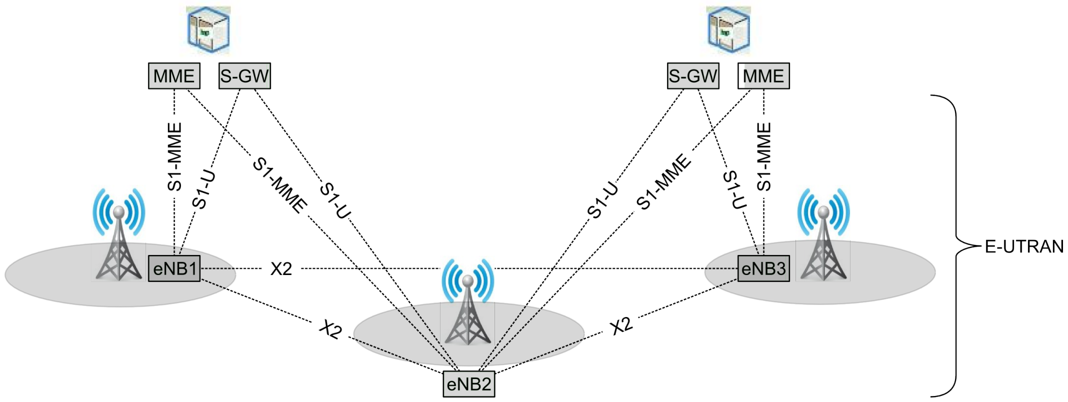


Figure 2.3: E-UTRAN architecture

2.1.3.3 User Terminal

Each UT is connected to the access network by an LTE air interface (Uu). This interface defines both a physical layer and a data link layer. In this thesis the terms UT and User Equipment (UE) are used interchangeably.

2.1.4 Transmission schemes

To achieve the high data rates targeted by the LTE standard requires wider radio bandwidth along with smarter modulation and multiple- access techniques. Different schemes are used for the uplink and the downlink. These schemes are described in the paragraphs below.

2.1.4.1 Uplink

Single carrier transmission based on Discrete Fourier Transform (DFT)-spread Orthogonal Frequency Division Multiplexing (OFDM) is used for the LTE uplink. This method is known as Single-Carrier Frequency-Division Multiple Access (SC-FDMA). In SC-FDMA the information of one symbol is spread over all the available subcarriers giving advantages such as robustness against multi-path propagation and increased frequency diversity. Moreover, the low Peak to Average Ratio (PAR) of power for SC-FDMA makes it an appropriate choice for the uplink as it facilitates conservation of the UT battery life.

2.1.4.2 Downlink

For LTE downlink transmission Orthogonal Frequency Division Multiple Access (OFDMA) was selected to implement a multiuser communication system. The channel bandwidth is subdivided into a number of non-frequency selective narrow band parallel sub channels. OFDMA allows several UTs to share the same bandwidth by assigning a subset of these sub-channels to individual UTs, resulting in several parallel low data rate streams. Lower complexity receiver implementation and high spectral efficiency are major advantages of OFDMA. Due to these reasons, OFDMA was adopted as the basis for both the 3GPP LTE and IEEE 802.16e (Mobile WiMAX) standards.

A comparison between OFDMA and SC-FDMA is shown in Figure 2.4.

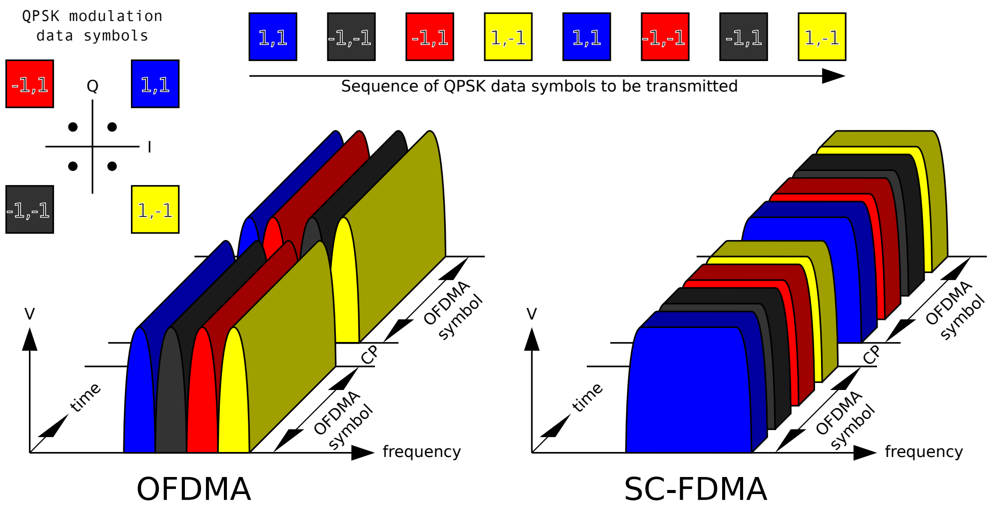


Figure 2.4: OFDMA and SC-FDMA [30]

2.1.5 Transmission resource structure

The transmission resource structure in LTE has three dimensions: time, frequency, and space. These represent three spaces in which a signal can be encoded.

2.1.5.1 Space domain

The spatial dimension is present due to the use of multiple antennas. Every added antenna port adds a new spatial dimension. These spatial channels can be exploited by using multiple input-multiple output (MIMO) techniques.

2.1.5.2 Time domain

In the time domain, a radio frame is 10 ms long. A radio frame is further divided into ten 1 ms subframes, each of which is split into two 0.5 ms slots. Every slot has six or seven OFDM symbols depending on whether a normal cyclic prefix or an extended cyclic prefix is used.

LTE is designed to support a large number of simultaneous voice calls. Additionally, LTE is optimized for carrying data traffic, thus the system must correctly prioritize different traffic classes and schedule the multiplexed transmission of the different types of IP layer traffic into the available radio frames. An LTE radio frame structure is of two modes [15]: Time Division Duplexing (TDD) and Frequency Devision Duplexing (FDD) mode.

TDD mode: In TDD mode the LTE radio frame has a duration of 10 ms and is divided into two 5 ms half-frames. Every half-frame slot is divided into 5 sub-frames as shown in Figure 2.5. In TDD mode the sub-frames can be either standard sub-frames or special sub-frames. The special sub-frames contain three fields: Downlink Pilot Time Slot (DwPTS), Guard Period (GP), and Uplink Pilot Time Slot (UpPTS). These special fields are used in Time Division Synchronous Code Division Multiple Access (TD-SCDMA). Optionally, these fields can have different lengths, but the combined duration of all of them must be 1 ms.

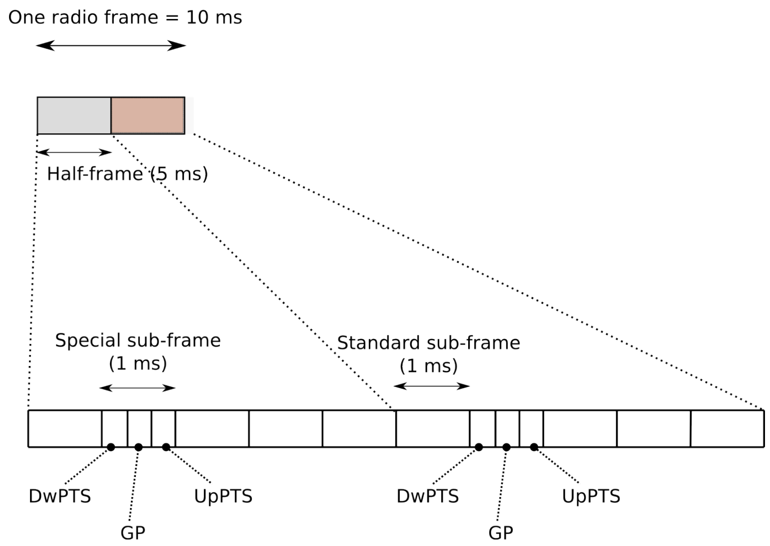


Figure 2.5: TDD mode

FDD mode: In FDD mode, the total radio frame length is 10 ms. There are 10 sub-frames of 1 ms. The overall structure of the radio frame is summarized in Figure 2.6.

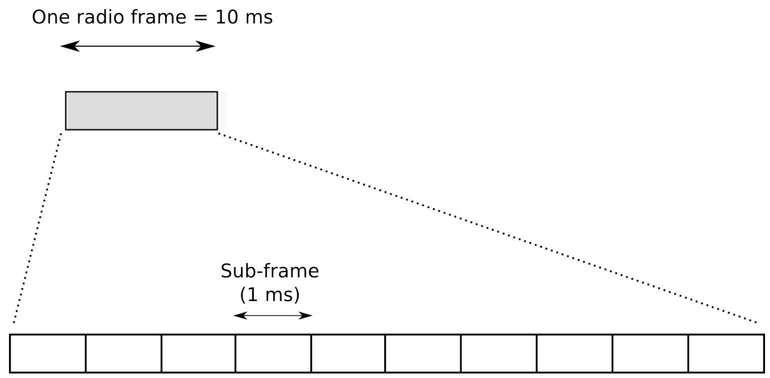


Figure 2.6: FDD mode

In this thesis FDD mode with a normal cyclic prefix will be considered in all the calculations and analysis. FDD mode was chosen because of its wide use in Germany and elsewhere in Europe.

2.1.5.3 Frequency domain

The basic frequency unit is a subcarrier which has a bandwidth of 15 kHz. These subcarriers are further grouped into blocks of 12 subcarriers occupying a total of 180 kHz.

2.1.5.4 Resource structure units

A Resource Element (RE) is the smallest resource unit and consists of one subcarrier for a duration of one OFDM symbol. The next bigger unit is a Resource Block (RB) which spans 12 subcarriers for a duration of one slot. A RB is thus comprised of 84 REs in the case of the normal cyclic prefix length and and 72 REs in the case of an extended cyclic prefix length. Generally, a Resource Block Pair (RBP) is considered the basic unit for scheduling when user data scheduling is performed. Furthermore, one or more consecutive RBs form a Transport Block (TB).

The detailed transmission resource structure with all the three dimensions for normal cyclic prefix length is shown in Figure 2.7.

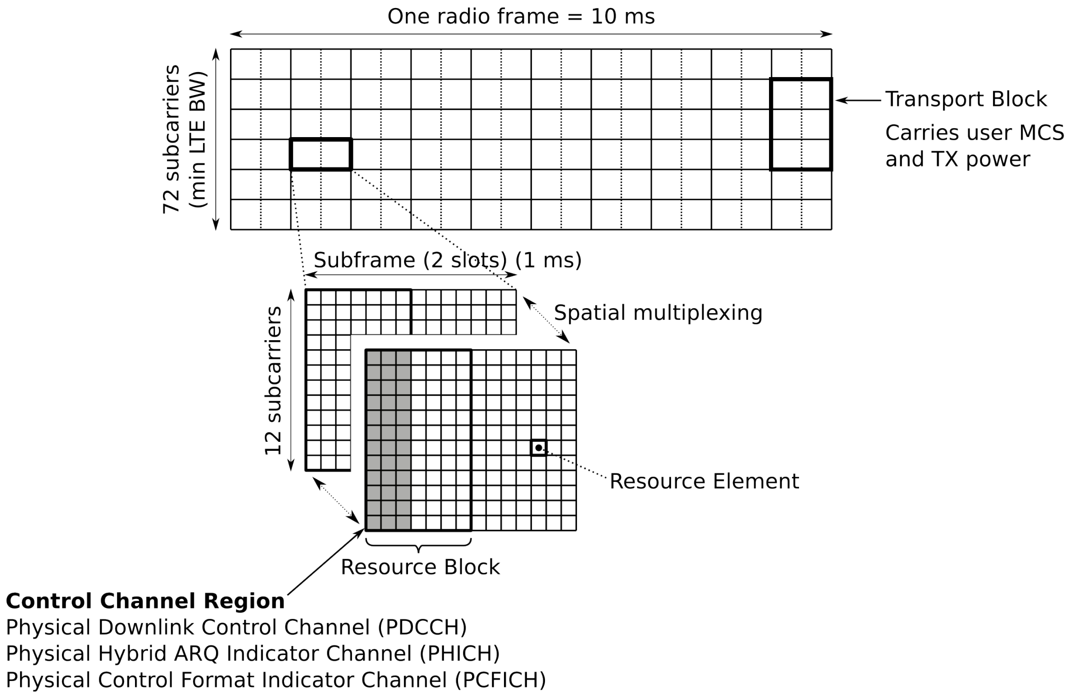


Figure 2.7: Transmission resource structure [29]

2.2 Control channel region of each sub-frame

The first three symbols of every sub-frame constitute the control channel region. As shown in Figure 2.7 this region basically consists of three channels [19]:

- Physical Control Format Indicator Channel (PCFICH)
- Physical Hybrid ARQ Indicator Channel (PHICH)
- Physical Downlink Control Channel (PDCCH)

The *Physical Control Format Indicator Channel (PCFICH)* carries a Control Format Indicator (CFI). This CFI contains information about the number of OFDM symbols (1, 2, or 3) used per sub frame for the downlink control channel region. The *Physical Hybrid ARQ Indicator Channel (PHICH)* carries Hybrid Automatic Repeat Request (HARQ) feedback. This feedback reports the success of uplink transmissions. The third channel, *Physical Downlink Control Channel (PDCCH)* will be the main focus of this thesis and it will be discussed in detail in the next chapter.

CHAPTER 3

Physical Down link Control Channel (PDCCH)

This chapter presents PDCCH in detail in order to establish a solid background for the calculations and simulations carried out in subsequent chapters. In E-UTRAN, the PDCCH carries the scheduling grants of both the uplink and the downlink. These scheduling grants indicate whom has been allocated channel capacity for the current sub-frame. The message carried by PDCCH is also known as Downlink Control Information (DCI) [29]. This DCI message is shown in Figure 3.1.

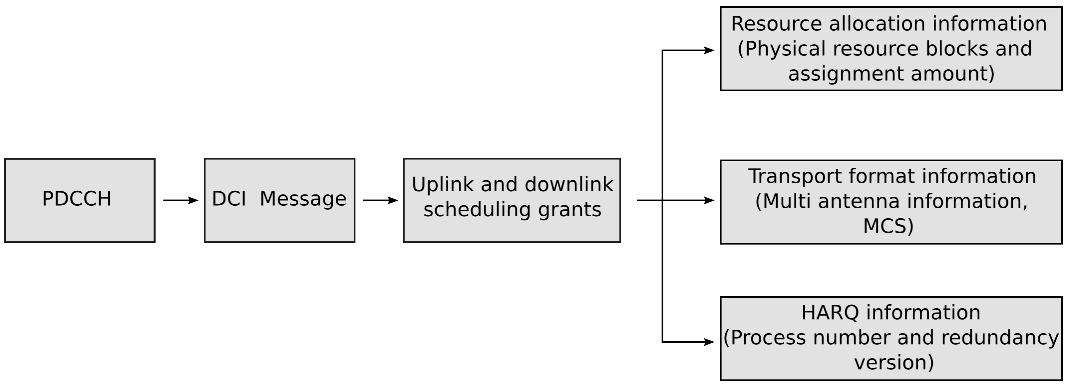


Figure 3.1: DCI message

3.1 DCI formats

Depending on the specific system deployed and its operation there are different DCI Message formats. These are shown in Figure 3.2. These message formats are:

- PUSCH resource allocation information: DCI Format 0 and 4
- PDSCH information with one codeword: DCI Format 1 and its variants (1A, 1B, 1C, and 1D)
- PDSCH information with two codewords: DCI Format 2 and its variants (2A, 2B, and 2C)
- Uplink power control information: DCI Format 3 and its variants (3A)

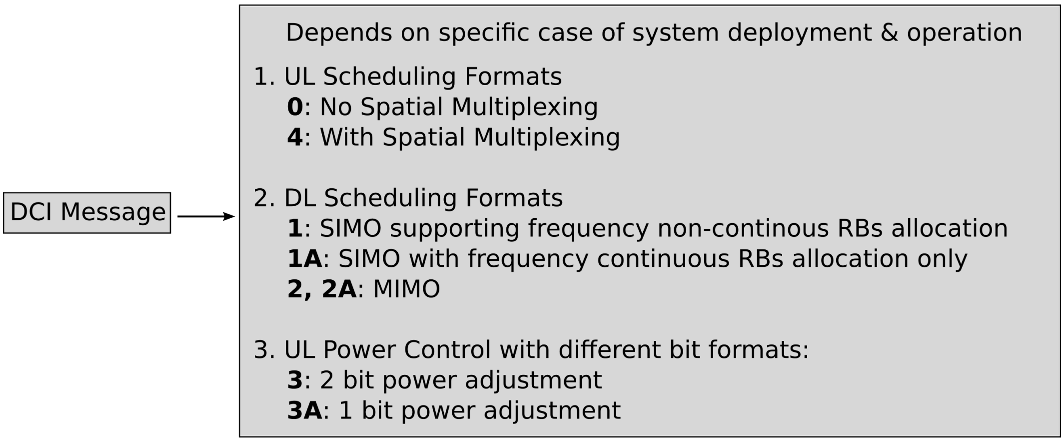


Figure 3.2: Formats of DCI message

3.1.1 DCI Formats for specific situations

There are two major factors that determine the DCI format of a PDCCH for a specific situation: the Radio Network Temporary Identifier (RNTI) Type [17] and the Transmission Mode.

The **RNTI** is used by the eNodeB to indicate that a particular PDCCH is for a specific UT. This is done by scrambling the 16 bit RNTI with the 16 bit CRC field of a DCI message. This process has been elaborately explained in 3GPP TS 36.213 [6]. There are several types of RNTIs: Cell-RNTI (C-RNTI), Semi Persistent Scheduling C-RNTI (SPS C-RNTI), System Information-RNTI (SI-RNTI), Temporary C-RNTI (TC-RNTI), Random Access-RNTI (RA-RNTI), and Paging-RNTI (P-RNTI).

Transmission mode refers to the Physical Downlink Shared Channel (PDSCH) mode selected for a UT corresponding to a PDCCH. The PDSCH is the main data carrying downlink channel. There are 7 types of transmission modes, numbered 1-7 [29]. These different types of transmission modes and their uses are shown in Table 3.1.

Table 3.1: Transmission modes

Mode	Scenario
1	Transmission from single antenna port
2	Transmit diversity with two or four antenna ports
3	Open-loop spatial multiplexing
4	Closed-loop spatial multiplexing
5	Multiple Input Multiple Output (MIMO)
6	Closed loop with recoding
7	Transmission using UE specific reference signals

3.1.2 DCI Formats in the situation that will be considered in this thesis: 25 RB (5 MHz), Single Input Multiple Output (SIMO)

In this thesis the case of a LTE system using 25 RBs totaling 5 MHz and single input multiple output (SIMO) antennas (the UT will have a single antenna and the eNodeB will have multiple antennas) is considered. In this case the RNTIs used are: C-RNTI for dynamic scheduling and SPS C-RNTI for semi persistent scheduling. For transmission, Mode 1 is used, indicating a single antenna port. DCI formats 1 and 1A are the desired formats for carrying downlink scheduling grants and DCI format 0 is the desired format for carrying uplink scheduling grants.

3.1.3 DCI Formats

Details of the formats that are used is given in the following paragraphs.

3.1.3.1 Downlink scheduling DCI formats 1 and 1A

DCI formats 1 and 1A are used for downlink SIMO and uplink power control. The bit format of these DCI format 1 and 1A messages are shown in Figure 3.3. The values shown in each of the fields indicate the bit length of the corresponding field.

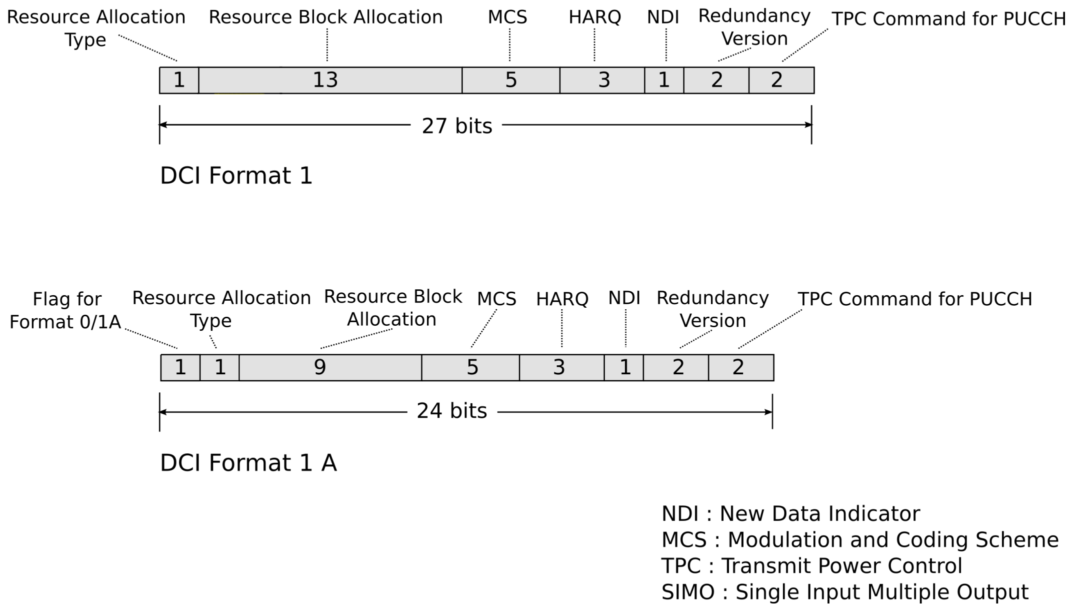


Figure 3.3: DCI format fields [8] (number of bits is for a 5MHz system bandwidth)

According to 3GPP TS 36.212 [8], the payload sizes listed in Figure 3.4 are ambiguous sizes, hence a zero bit has to be appended to the payload to produce the DCI field that will be transmitted. As format 1A turns out to be 24 bits in length, one padding zero is appended to make it 25 bits in length.

{12, 14, 16 ,20, 24, 26, 32, 40, 44, 56}

Figure 3.4: Ambiguous sizes

DCI message fields of format 1 and 1A [12] are defined below.

DCI format 0/1A indication (1 bit): Used to differentiate between format 0 and 1A (as both have same message size). This field only exists in format 0 and 1A messages.

Resource Allocation header (1 bit): In the case of Format 1, this field indicates whether the resource allocation is of type 0 or type 1. For format 1A this field tells whether it is a localized or distributed virtual RB assignment of type 2.

RB allocation(13 or 9 bits): Indicates the position of the RBs assigned to the UT within a RB domain. There are three types of resource allocations: type 1, type 2, and type 3. Format 1 supports type 0 and 1, while Format 1A supports type 2. These are shown in Figure 3.5 and described below:

Type 0 → contiguous RBs forming Resource Block Groups (RBGs). The number of RBGs is a function of the downlink bandwidth as shown in Table 3.2. The field points to these RBGs instead of individual RBs. If the field is 13 bits, then all of bits represent a single RBG which is a pair of two adjacent RB; in this case the UT will search for its resources in only the RBGs which are assigned to it.

Type 1 → total number of RBs are divided into subsets. This type supports non-continuous RB assignment. In the case of a 13 bit field, 11 bits represent a single RBG and the remaining 2 bits are for subset and offset selection in order to enable Distributed Mapping. Note that the UT will search for its resources only in those RBGs which are assigned to it.

Type 2 → Does not rely on a bitmap, but rather encodes the resource allocation as a start position and number of RBs allocated. This type is used to support continuous RB assignment. If the field is 9 bits, then RBs are not directly allocated - but rather Virtual Resource Blocks (VRBs) are allocated which are then mapped to RBs. The VRBs are the scheduling units. These VRBs are further divided into two types: localized and distributed.

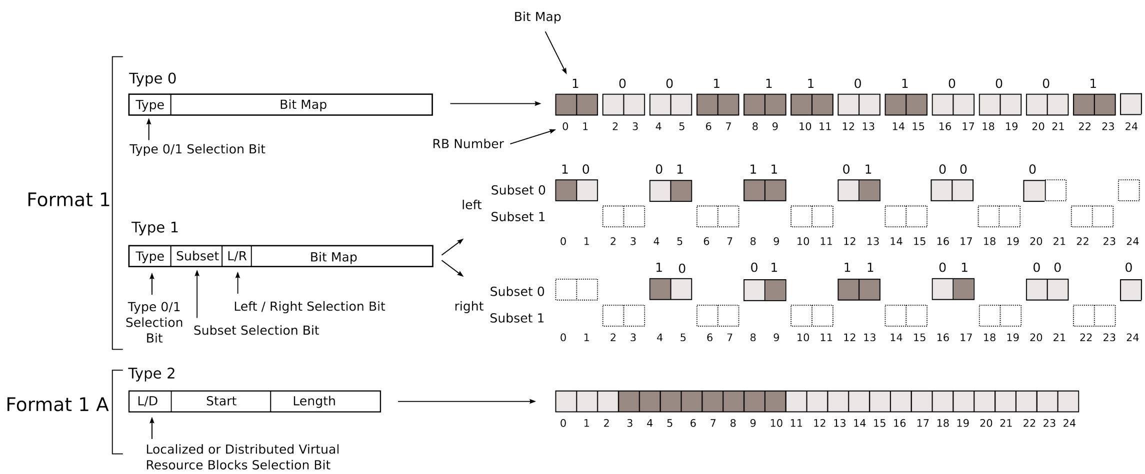


Figure 3.5: Downlink Resource Block allocation types for 25 RBs (here RBG is 2)

Modulation and Coding Schemes (MCS) (5 bits): This field indicates the modulation scheme, code rate (number of allocated PRBs), and transport block size.

HARQ (3 bits): This field informs the UT about the HARQ process used for soft combining. In this approach an incorrectly received data packet is stored and is used together with a re-transmitted packet to help produce a successful decoding.

In LTE downlink HARQ transmission is asynchronous so this requires that the HARQ process number be communicated - so that the receiver can know which HARQ process is being referred to. Without the process number information, the UT might confuse

Table 3.2: Resource Block Groups for different downlink bandwidths

Number of RBs	RBG size
<10	1
11-26	2
27-63	3
64-110	4

different processes and combine the wrong data [7]. The number of HARQ processes is fixed for both uplink and downlink to 8.

Redundancy Version (2 bits): This field contains an HARQ parameter which is used with Incremental Redundancy (IR) to tell which re-transmission version is used [14].
Every re-transmission contains some different information from the previously transmitted version. Multiple sets of coded bits are generated each representing the same set of information bits. Every time re-transmission occurs a different set of coded bits are used with a different redundancy version. Thus when ever a re-transmission occurs the decoder gains some extra information which helps in decoding.

New Data Transmission NDI (1 bit): This field indicates whether the packet is a retransmission or new transmission.

TPC Command for PUCCH (2 bits): The transmit power control (TPC) command is used for the transmission power adaptation for the PUCCH on the uplink.
The difference between format 1 and format 1A is that format 1 allows frequency non-continuous PDSCH RBs allocation; where as format 1A allows only frequency continuous PDSCH RBs allocation. With non-contiguous allocation the advantages of frequency diversity can be exploited. On the other hand, contiguous allocation reduces the payload size of the DCI message at the cost of reduced allocation flexibility.

3.1.3.2 Uplink scheduling DCI format 0

The DCI formats 0 is used for uplink SIMO and uplink power control. The bit format of this DCI format 0 message is shown in Figure 3.6. The values shown in each of the fields indicate the bit length of the corresponding field.

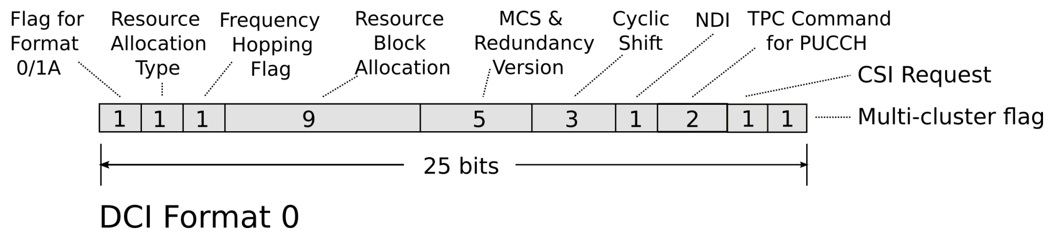


Figure 3.6: DCI Format 0 (the indicated numbers of bits are for a 5MHz system bandwidth)

If number of information bits in format 0 mapped to a certain search space, is less than the payload size of format 1A for the same cell and search space, then zeros shall be appended until the payload size of format 0 becomes equal to that of format 1A. The DCI message fields of format 0 which are different from the fields of format 1 and 1A are described below.

CSI Request (1 bit): This bit requests the transmission of an aperiodic Channel Quality Indicator (CQI) report. The eNodeB can ask for an aperiodic CQI report to be transmitted on the Uplink Schared Channel (UL-SCH) by setting this bit of the uplink grant in a DCI message.

NDI (1 bit): This bit is intended for synchronizing the scheduling commands with HARQ ACK and NACK message status. It tells the UT whether transmission of a new transport block or re-transmission of the previous transport block is granted.

Muti-cluster flag (1 bit): The bit indicates whether one or two clusters of RBs are used for the uplink transmission. This field and capability was first introduced in 3GPP Release 10.

The basic resource assignment scheme in the uplink is single cluster assignment where the RBs are continuous in the frequency domain. However, in Release 10, support for up to 2 clusters on a single component carrier has been added.

Cyclic Shift (3 bits): Phase rotation of the uplink DeModulation Reference Signals (DMRS) is used to support MIMO. By assigning different reference-signal phase rotations to scheduled terminals in the same time-frequency resources, the eNodeB can estimate the uplink channel response from each terminal and suppress the inter-terminal interface by signal processing.

The requirements of LTE-Advanced are quite ambitious, especially for the uplink. These requirements are difficult to achieve with a single transmit antenna from the UT. Therefore multiple transmit antennas can be supported for an LTE-Advanced uplink. However, we will not consider the case of a UT with multiple antennas in this thesis – this is left as future work.

RB Allocation + Freq Hopping Flag (9+1): The traditional uplink resource allocation scheme has always been a single-cluster allocation. So continuous RBs in a single frequency are assigned. In LTE release 10 for the first time the option of multi cluster uplink transmission has been added. The uplink RB allocation for 25 RBs is shown in Figure 3.7.

Type 0 → represents the basic single cluster resource allocation. This approach is similar to the type 2 approach in the downlink resource allocation. The hopping flag indicates whether two consecutive RBs in the first half and in second half of a sub frame are used together or if frequency diversity is applied to them.

Type 1 → represents the multi cluster resource allocation scheme in which 2 clusters are supported. The starting and the ending positions in the frequency domain of both clusters are encoded in the index. Frequency hopping is not allowed in type 1. Also more bits are required to signal 2 clusters of resources so the frequency hopping flag can be merged in order to use it for the cluster allocation. Despite the increase with the merger of the frequency hopping flag, the total field still does not have enough bits to signal 2 clusters for all bandwidths. So in this case an approach similar to downlink resource allocation is applied and cluster allocation is indicated by the starting and the ending RBG (here RBG

= 2 as we are considering 25 RBs).

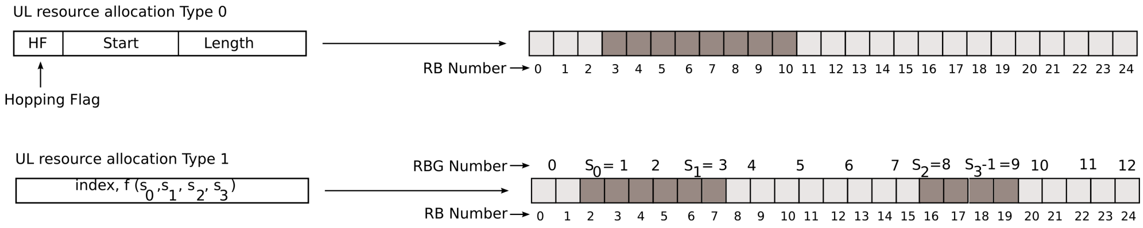


Figure 3.7: Uplink Resource Block allocation types for 25 RBs

3.1.4 From DCI Message to PDCCH pay load Formation

The overall process of filling the PDCCH payload is shown in Figure 3.8. Each of these steps will be described in more detail in the following paragraphs.

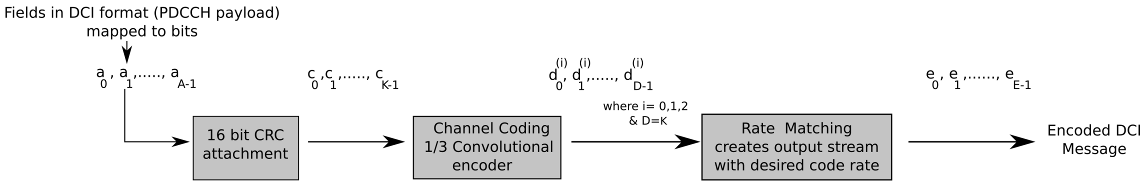


Figure 3.8: Processing of a DCI message [8]

3.1.4.1 Step 1: 16 bit CRC attachment

Performing the 16 bit CRC attachment to the DCI formats is shown in Figure 3.9.

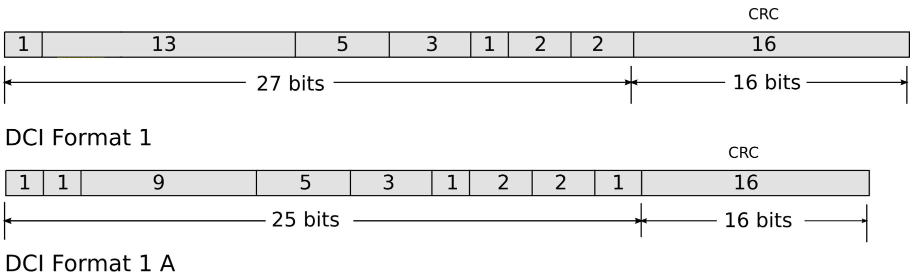


Figure 3.9: CRC attached to the DCI messages

3.1.4.2 Channel coding

The CRC that is attached to a DCI message undergoes 1/3 convolution coding. Convolutional coding is a form of forward error correction. Channel coding improves the channel goodput, since the redundant information makes it easier to recover the messages and avoids the time required for a request for retransmission and for the retransmission. This convolution coding is shown in Figure 3.10.

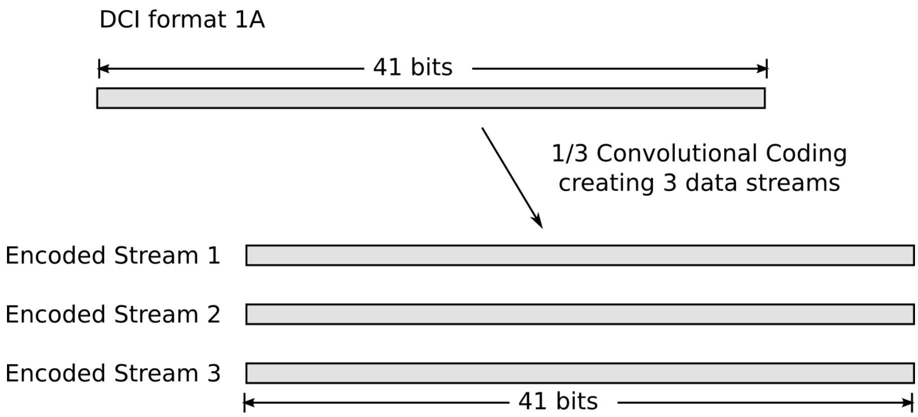


Figure 3.10: Channel coding

3.1.4.3 Rate matching

Rate matching creates an output bit stream with a desired code rate. The three bit streams (40 bits long) from the convolution encoder are first appended with 24 nulls, then interleaved to form a 64 bit stream. A circular buffer (192 bits) is formed by concatenating these three 64 bit streams.

Finally bits are selected and pruned from the circular buffer to create an output bit stream with our desired code rate [20] [8]. This is done by sequentially outputting the bits in the circular buffer from f(start) (looping back to f(start) after f(last) bit). Nulls bits are discarded. This process is continued until the length of the output is z times the input length. In this way we achieve a code rate of 1/z. The rate matching process is shown in Figure 3.11.

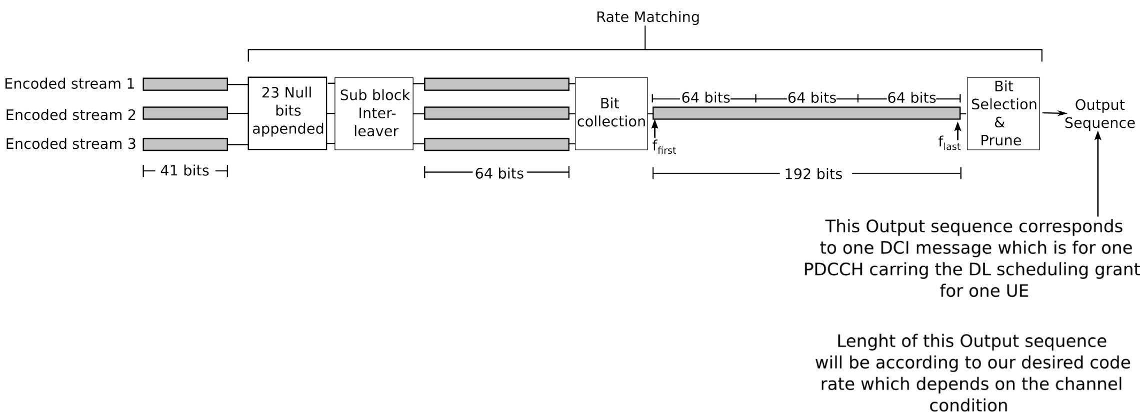


Figure 3.11: Rate Matching

3.1.4.4 Scheduling-type indication for UT

A UT can be scheduled for dynamic transmission, semi-persistent transmission, or semi-persistent re-transmission depending upon the MAC-ID (RNTI) and NDI field [6]. The RTNI and NDI field are used to indicate these different scheduling schemes, as shown in Figure 3.12. The periodicity (after how many TTIs) of semi-persistent scheduling of a UT is configured by Radio Resource Control (RRC) signaling.

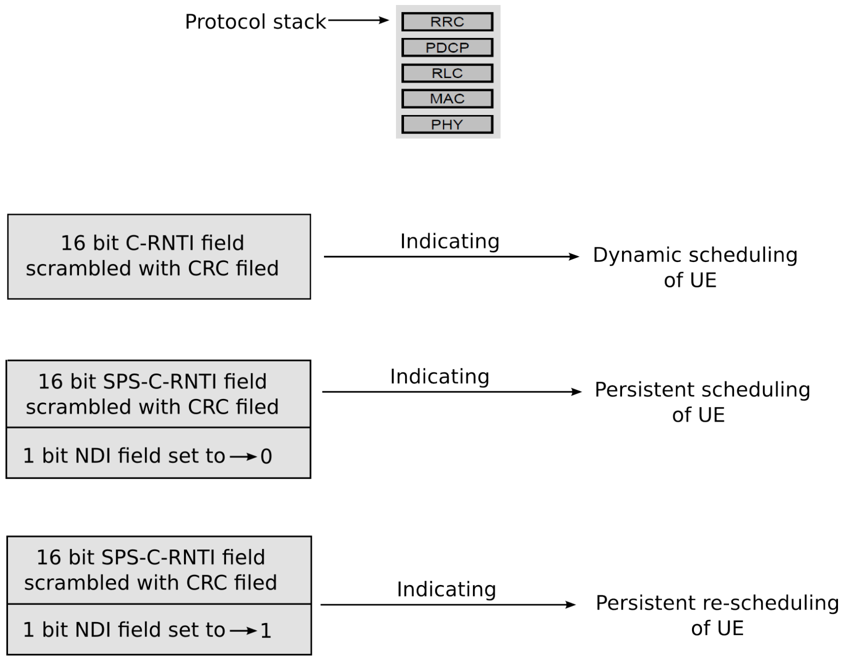


Figure 3.12: RNTI along with NDI indicating type of scheduling

Upon reception of a PDCCH, the UT will check the CRC field of the DCI message using its own RNTI. If the CRC is found to be valid, then the message was correctly received and is intended for this UT, otherwise it is discarded. In this way the MAC-ID for a UT is encoded in the CRC and transmitted implicitly instead of being transmitted explicitly.

3.2 Control Channel Element

Every PDCCH carries a scheduling grant (either uplink or downlink) for one UT. Individual PDCCHs are transmitted using *Control Channel Elements (CCEs)*. A CCE is the minimum resource unit allocated to a PDCCH for control signalling. Each CCE consists of nine Resource Element Groups (REGs), where each REG corresponds to four physical Resource Elements (REs). Four QPSK symbols are mapped to each REG. So in total a CCE has 36 REs which are distributed over a time duration of 3 OFDM symbols and whole frequency bandwidth (this is 25 RBs in the case considered in this thesis).

3.2.1 PDCCH aggregation formats

There are four CCE aggregation formats. The difference between the aggregation formats and the DCI formats must be kept in mind. The PDCCH formats are shown in Table 3.3 and the PDCCH formats and sizes are shown in Figure 3.13.

Table 3.3: PDCCH formats [13]

PDCCH Aggregation format	Number of CCEs	Number of REGs	Number of bits	QPSK Modulation (effective coding rates)
0	1	9	72	2/3
1	2	18	144	1/3
2	4	36	288	1/6
3	8	72	576	1/12

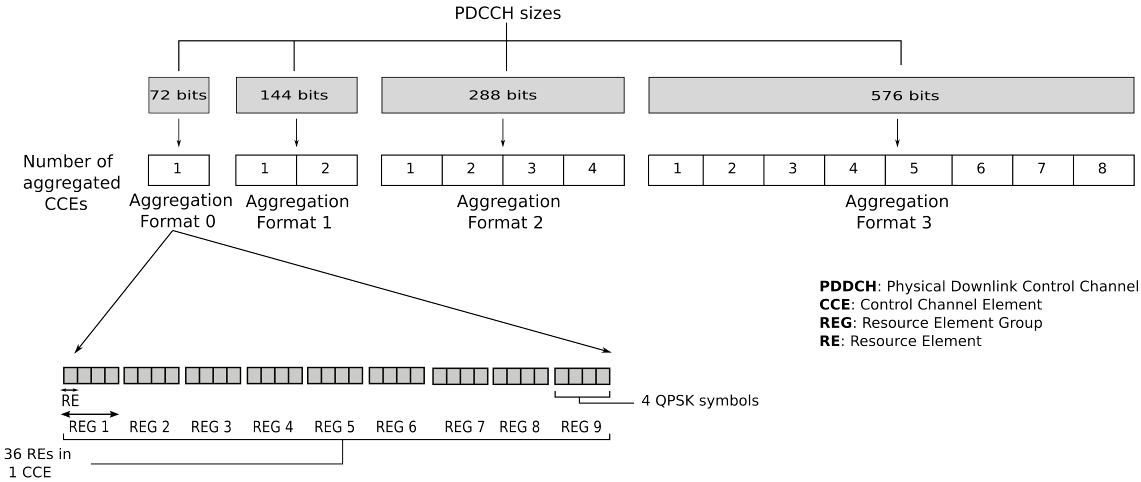


Figure 3.13: PDCCH formats and sizes

3.2.2 PDCCH format depends on channel conditions

The number of CCEs required for the transmission of a particular PDCCH is determined by the eNodeB according to the channel conditions. A UT in favourable conditions may only require a single CCE, whereas a UT with a bad channel may require multiple CCEs for the transmission of a single PDCCH for this specific UT.

3.2.3 Mapping of CCEs

CCEs are mapped to the control region as shown in Figure 3.14. The number of schedulable UTs per sub frame will change (increase or decrease) based upon:

- Employing different scheduling strategies: If semi-persistent scheduling is applied, then more UTs can be scheduled as PDCCH resources are not needed every time a semi-persistently scheduled UT is allocated resources in the PDSCH.
- By increasing the number of allocated CCEs per PDCCH: If channel conditions are bad for a UT, then one CCE might not be enough for a PDCCH which is carrying the scheduling grant for this particular UT. In this case, this PDCCH might need to be allocated more than one CCE (2, 4, or 8) depending upon the current conditions of the channel, thus the number of schedulable UTs per sub frame will decrease–as some UTs will require more CCEs–, hence reducing the number of CCEs remaining which can be allocated to other UTs for their scheduling.

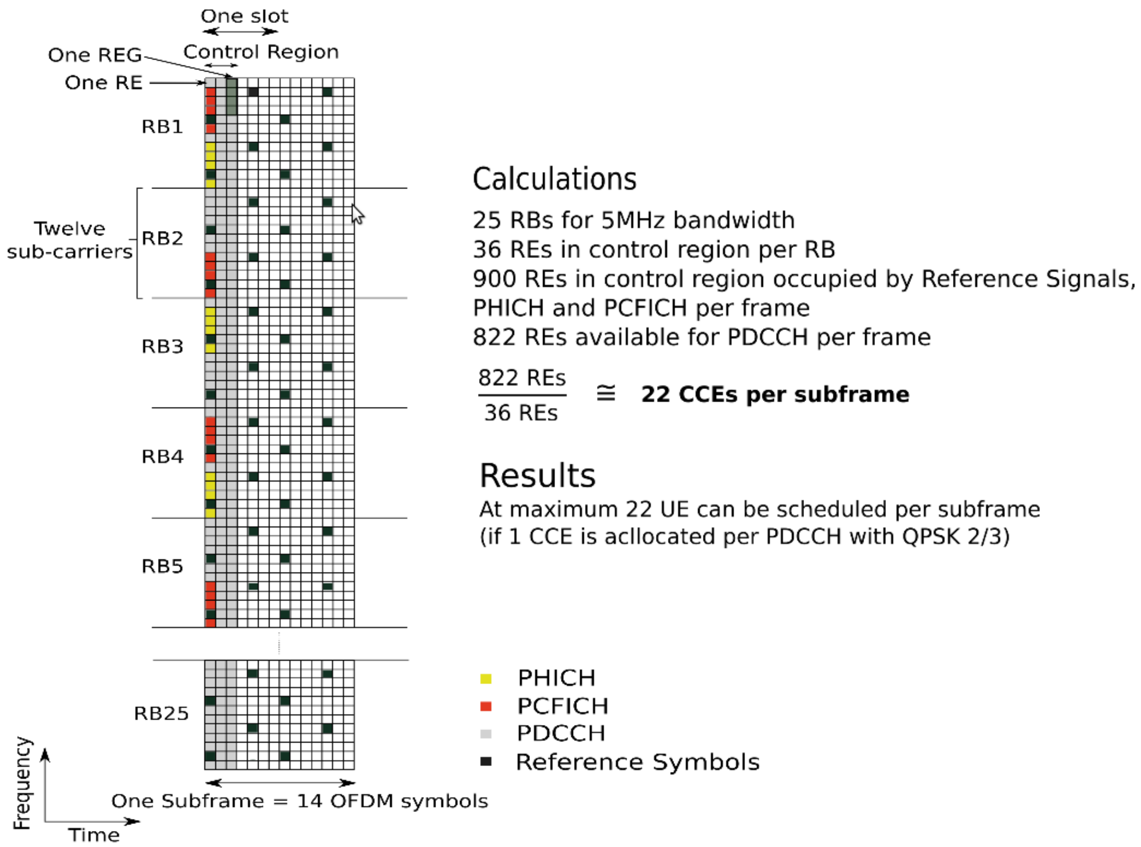


Figure 3.14: Resource mapping

3.3 Difference between uplink and downlink resource management

The most important question associated with resource management is, does UT voice call scheduling require equal control channel resources per sub-frame for uplink and downlink? A user deployment scenario is shown in Figure 3.15.

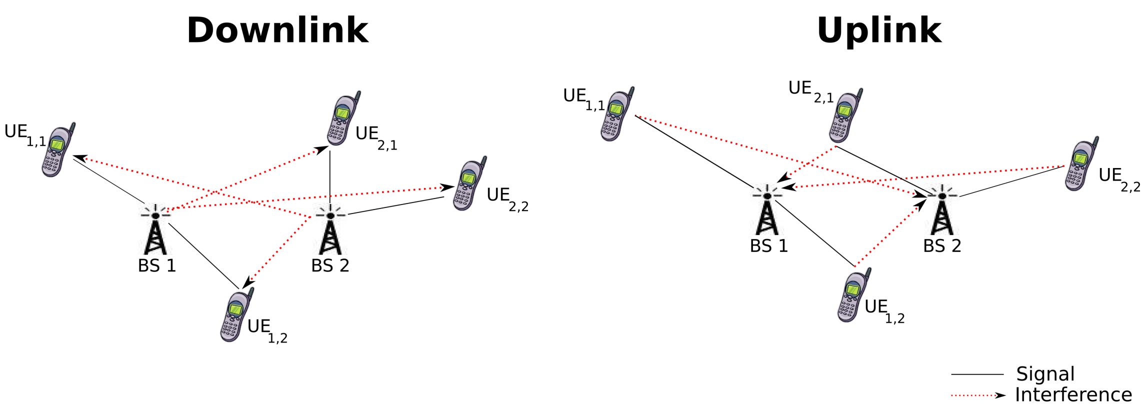


Figure 3.15: Downlink and uplink scenarios

3.3.1 Downlink

In the downlink, the influence of SINR depends on a number of factors such as channel attenuation to the serving Basestation (BS) and channel attenuation to interfering BSs. These two factors depend on channel fading, and the position and speed of the UT. However, even more important is the occupation of RBs in the interfering BS, i.e., is the interfering BS actually using this RB for scheduling of any UT or not. Furthermore, by shifting the RBs of different UTs, the channel conditions of the corresponding UT can become better or worse resulting in a requirement for an increase or decrease in RBs as shown in Figure 3.16.

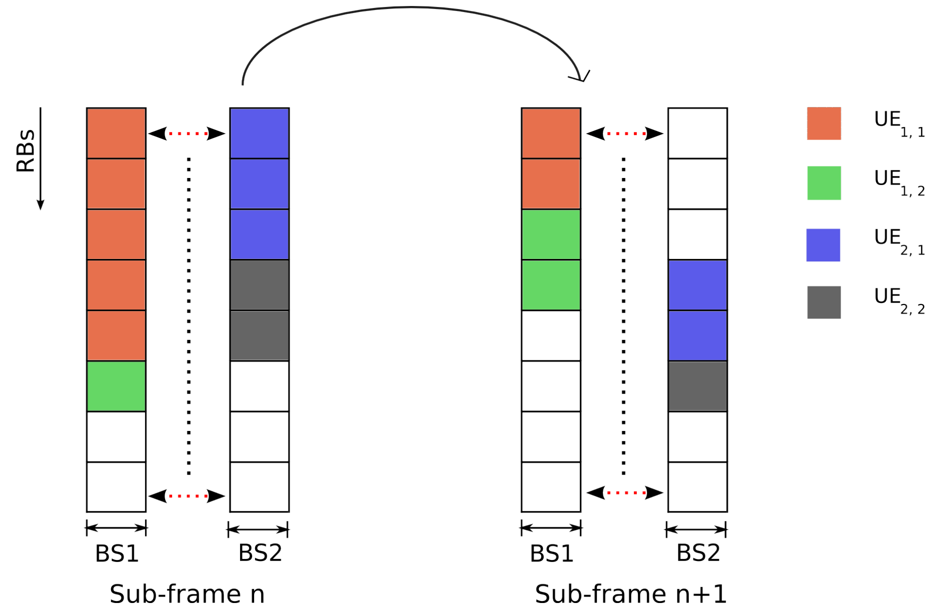


Figure 3.16: Downlink scheduling

However, in case of the downlink, the influence of SINR does **not** depend upon which terminal is active in the interfering BS's cell. In fact, it is just a question of, if there is any UT scheduled in the RB that causes interference or not (with one exception being that the interfering BS uses transmission power control which depends on which terminals are being served).

Moreover, because of the on-off traffic model the call scheduling map changes every millisecond, thus resulting scheduling process is very dynamic. This is even more so since a voice call might only need to transmit once every 20 ms.

3.3.2 Uplink

The interference situation is more complex for the uplink. The influence of SINR not only depends on the occupation of RBs in the interfering BS's cell, but also on which UT is occupying these RBs. To demonstrate this phenomena consider the uplink scenario shown in Figure 3.15. UT 11 is being served by BS1 and interferred with BS2 (UT 21 occupies the corresponding RBs), and UT11 has very bad channel conditions as the interfering link is shorter than the serving link. Figure 3.17 shows the RB allocation in the respective BSs, where UT 11 is being interfered by UT 21. By reallocating the RBs of UTs, the scheduling map can be completely transformed.

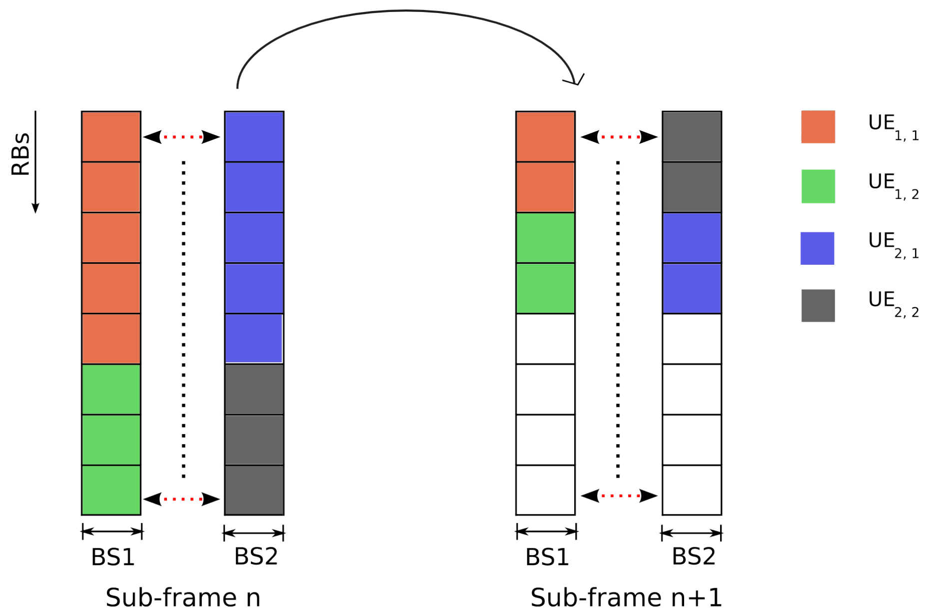


Figure 3.17: Uplink scheduling

Moreover, due to SC-FDMA, unlike the case of the downlink, only contiguous RBs allocation is permitted for the uplink as shown in Figure 3.18. This makes the scheduling map even more vulnerable to interference effects due to this extra constraint on resource scheduling.

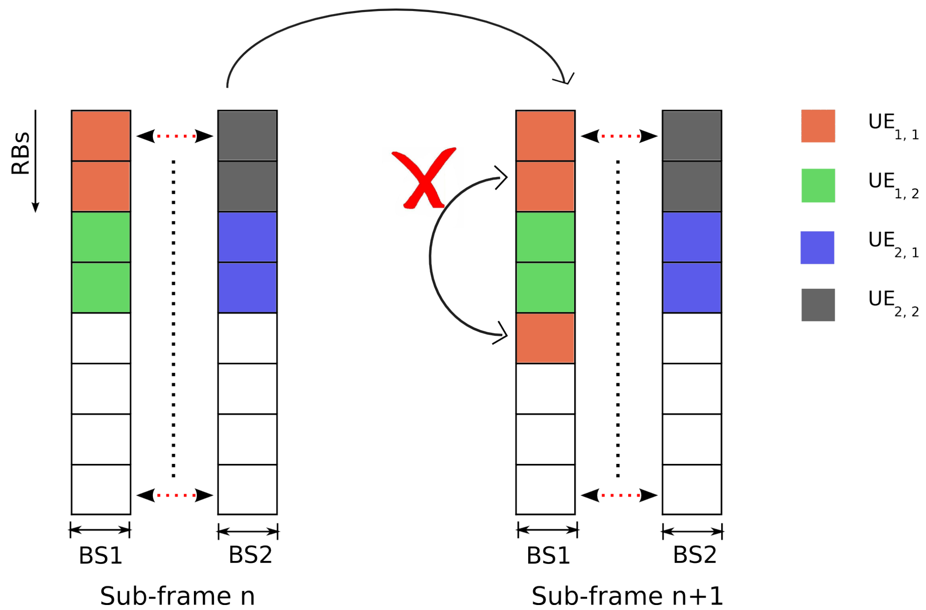


Figure 3.18: Contiguous RB allocation in uplink

3.3.3 Uplink is the voice capacity bottleneck

To sum-up, on the uplink all the constraints of the downlink apply, along with an extra constraint of contiguous RB allocation. Moreover, there is greater interference variance in the uplink channel as compared to the downlink. For these reasons we can concluded that the uplink is a bottleneck for voice capacity.

CHAPTER 4

Scheduling concepts

This chapter describes a number of fundamental concepts [25] which play a central role in the overall scheduling process. The chapter starts with the topic of link adaptation. After that different Transport Block (TB) fitting strategies are presented and their comparative performance is evaluated. Following this, smart techniques for frequency relocation and time relocation that have been employed in this thesis work are explained. Finally, specific scenarios are mentioned where dynamic scheduling is employed.

4.1 Link performance prediction

Accurate link performance prediction is the basis of good link adaptation. This performance prediction plays a vital role when dealing with algorithms which are based on the selection of an adaptive Modulation and Coding Scheme (MCS) every Transmission Time Interval (TTI).

4.1.1 Link-to-System (L2S) mapping

An elaborate model describing link performance is presented in Figure 4.1. By considering the channel characteristics individual UTs are assigned radio resources and transmission power levels. Based on measured and estimated information, a set of quality measures is obtained from the REs ($z = 1...Z$) involved in the entire process of packet transmission. This takes into account multiple factors including the number of receive and transmit antennas, beam forming, and spatial multiplexing. The number of these parameters ($\phi_{1...Z}$) is quite large, thus it is desirable to compress these parameters into a smaller number of quality measures ($\theta_{1...N}$). In this thesis SINR values are the quality measures computed using the Channel Quality Indicator (CQI) of each RB of a given UT. Subsequently these SINR vales are mapped to a Link Quality Metric (LQM) which produces scalar values. This mapping is referred to as Effective SINR Mapping (ESM) and is computed by the formula:

$$SINR_{eff} = I^{-1}\left(\frac{1}{N}\sum_{n=1}^{N} I(SINR_n)\right) \tag{4.1}$$

where
$SINRn$: SINR on the n^{th} sub-channel
N: Number of sub-channels allocated for the transmission
$I(.)$: Invertible function of information measure

The final step in L2S is mapping the LQM to the link performance measure, which is normally the Block Error Rate (BLER) or Packet Error Rate (PER). There are several ESMs that are used to predict the instantaneous link performance. In the following discussion we have utilized one of these, the Mutual Information Effective SINR Mapping (MIESM).

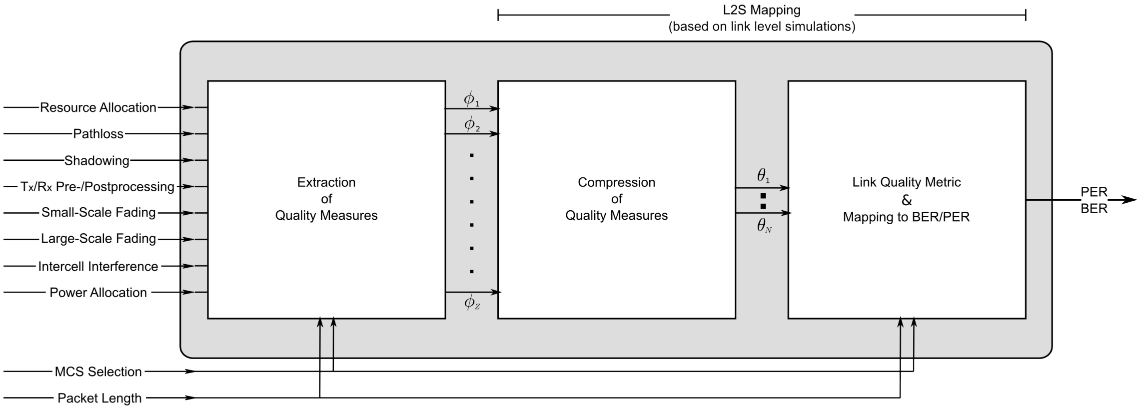


Figure 4.1: A generic link performance model [10]

4.1.2 Mutual Information Effective SINR Mapping (MIESM)

In [10] Brueninghaus, et al. have evaluated several performance models showing that the best performance is achieved by the mutual information (MI) based mapping process. Mutual Information Effective SINR Mapping (MIESM) is split up into a modulation model and a coding model. Figure 4.2 shows an representation of the computational procedure for the MIESM method.

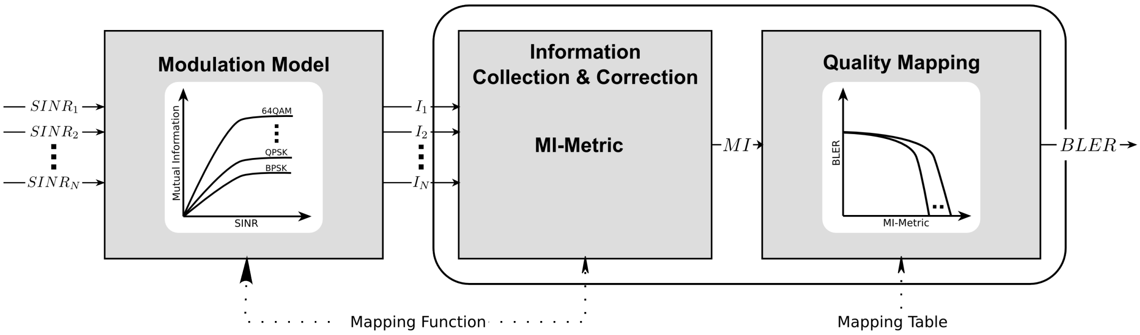


Figure 4.2: MIESM method

The modulation model determines the average mutual information per coded bit for each received symbol of a certain modulation scheme according to the coded block SINR values. The coding model is assigned the task of mapping the MI values to the decoding performance which is evaluated according to the BLER. A number of approaches can be adopted to evaluate MI. The two main classes in which these approaches can be classified are Mean Mutual Information per Bit (MMIB) and Received Bit Information Rate (RBIR). More details on these can be found in [31] and [14].

4.1.2.1 Set of potential TBs

In section 4.1.1 and 4.1.2, it was explained that SINR values per RB are mapped by the MI metric, resulting in effective SINR values for TBs which are further mapped to BLER. This gives rise to the possibility of multiple potential TBs depending on the effective SINR values. Two factors that decide upon the possible TB sizes of a call are:

- The particular RB space under consideration and

- Starting index of RB in the RB space.

In this way a candidate set of potential TBs is determined by link adaptation, as shown in Figure 4.3. Now the choice of selecting a particular TB from the potential set is made by the adopted TB fitting strategy as explained in section 4.2.

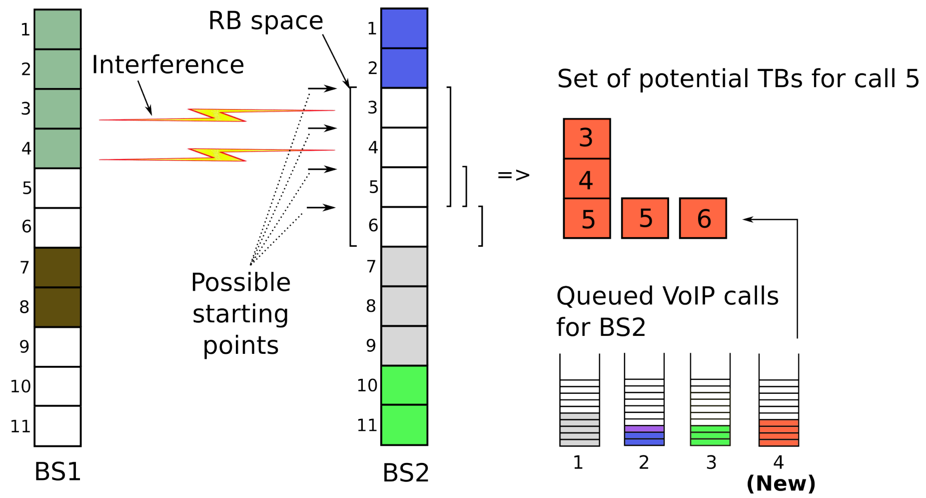


Figure 4.3: Possible potential TB set

4.2 Fitting strategies

For a VoIP call the method of Transport Block (TB) selection from the available free Resource Blocks (RBs) in a sub-frame is important as this affects the scheduling process by having a direct impact on the availability of required resources to other calls and the overall interference situation. In each sub-frame there can be RBs which are never allocated to any call, together with RBs which were assigned to a particular call but now are free due to the expiration of an earlier reservation -due to termination of the VoIP session. These resources have to be utilized efficiently in order to minimize the waste of resources. In the following subsections some algorithms are presented which can be applied to choose a TB from the available RBs spaces in a sub-frame.

4.2.1 First fit

This is the simplest fitting strategy. The TB is selected from the first RB space which is large enough to accommodate it.

4.2.2 Best fit

In best fit, that RB space is chosen which results in the minimum number of left over RBs after the TB is scheduled in it. This way the TB is scheduled in the smallest RB space which can accommodate the VoIP packet. However, a the problem with the best fit strategy is that it produces small leftover spaces which are sometimes hard to fill. This is especially true for many voice CODECs as the encoded voice payload is very small (33 octets).

4.2.3 Least fit

Least fit is just the opposite of the best fit strategy. In this approach the space that is chosen results in the maximum left over number of RBs (after the TB of a call is scheduled in that space).

4.2.4 Random fit

In random fit, a TB from the set of potential TBs is randomly chosen for the VoIP audio packets in the VoIP session.

4.2.5 Examples of the various fitting strategies

Figure 4.4 shows an example which illustrates these different fitting strategies. Initially we have three VoIP calls 1, 2, and 3 occupying RBs in the sub-frame. In the next time period a new call appears which is scheduled using each of the different fitting strategies.

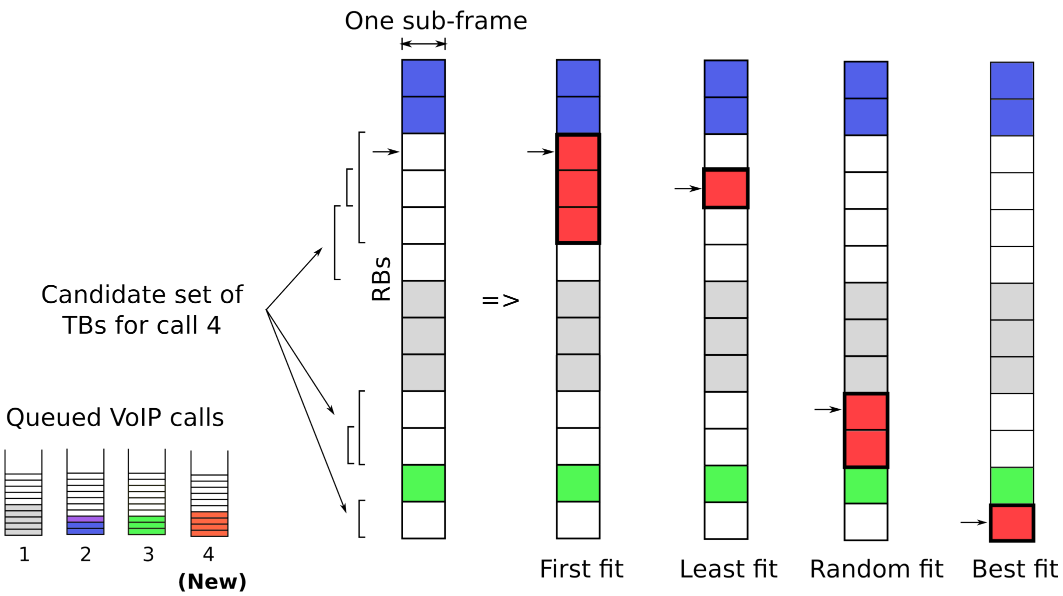


Figure 4.4: Different fitting algorithms

4.3 Frequency Relocation

Due to the changing channel conditions there is a chance that the RBs required by a persistent VoIP call for data transmission may vary from one sub-frame to the next sub-frame. As a result the TB previously allocated to a call might not be sufficient to carry the required amount of data. This requires relocation of additional RBs in the frequency domain, as shown in Figure 4.5.

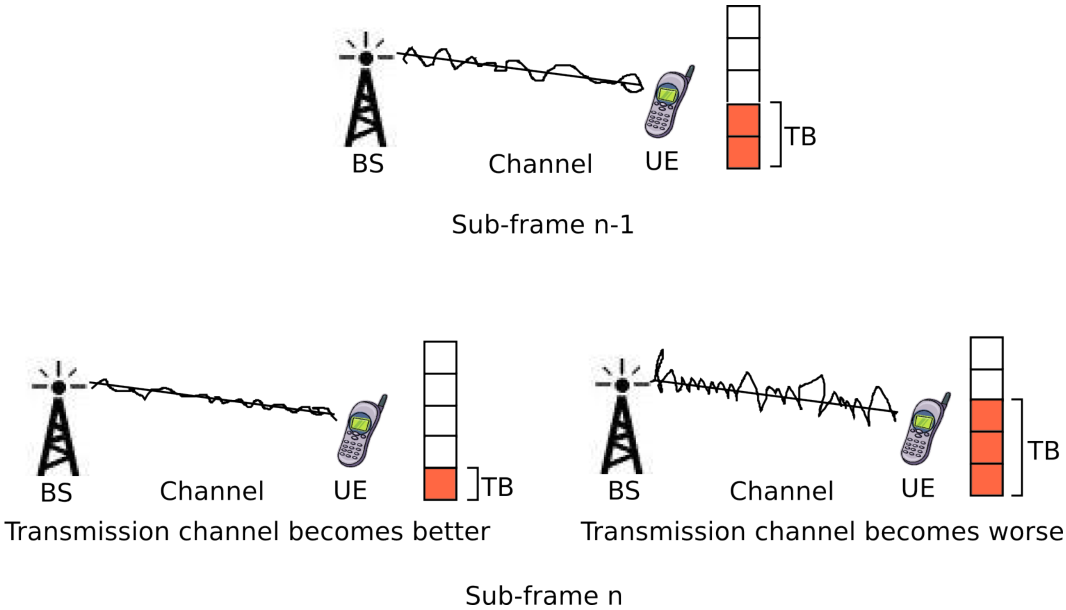


Figure 4.5: Relocation of RBs in frequency

4.3.1 Steps involved

In the frequency relocation strategy that has been adopted, the steps involved are shown in Figure 4.6.

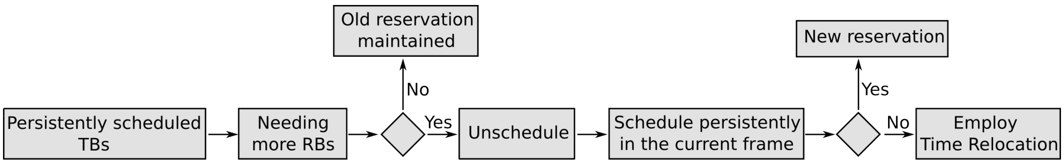


Figure 4.6: Adopted approach for frequency relocation

4.3.1.1 Check persistent calls

First, all persistent calls in the current sub-frame are checked for their TB sizes. The calls which maintain or reduce their TB size keep their previous reservation. All those calls which have an increase in their RBs requirements are identified and taken care of in later steps.

4.3.1.2 Unschedule the calls

The second step is to unschedule all the persistent calls needing more RBs from the current frame. This results in new available RBs which as they are not yet allocated to any call.

4.3.1.3 Try to schedule persistently in the current frame

We first try to schedule calls requiring more RBs in the current sub-frame while keeping in mind their new TB requirement. If successful, then these calls get a new reservation in the current sub-frame.

4.3.1.4 Employ time relocation for the remaining calls

If for any reason a call is not able to get a reservation in the current sub-frame, then this call is left for the following sub-frames. This way the channel resource reservation is delayed in time. This is called time relocation and is further discussed in Section 4.4.

In Figure 4.7, an example is shown where three calls are scheduled in a sub-frame and frequency relocation is applied to the calls requiring more RBs.

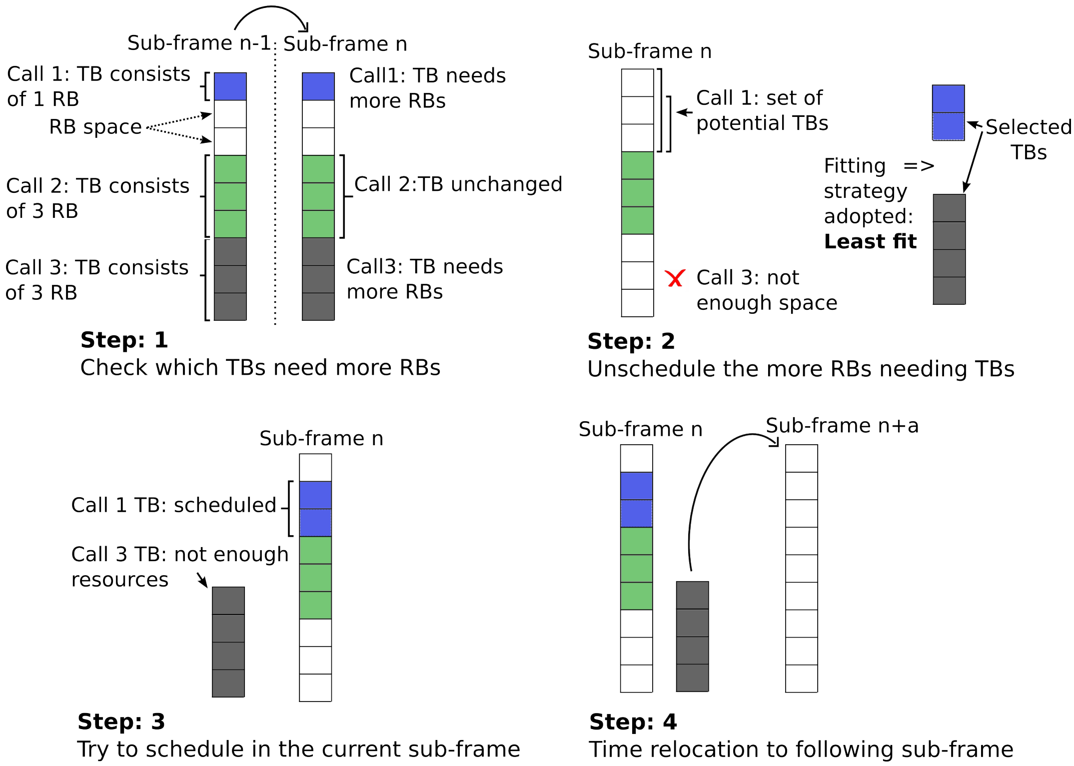


Figure 4.7: Frequency relocation steps

4.4 Time Relocation

It may happen that there are not enough available RBs which can be assigned to the calls enqueued for the sub-frame under consideration. In this situation there are two options: the calls which are not able to find resources are ignored or these calls are relocated to a future sub-frame where adequate RBs are available which can be assigned to them. The process of delaying the reservation of resources for calls to the following sub-frames is called *Time Relocation*. Resources for the time relocated calls are checked in the next 20 subframes passing during the VoIP inter arrival time period. If none of these 20 sub-frames have sufficient resources for this time relocated call (rare situation), then these packets are queued until some resources become available.

In the example shown in Figure 4.8, there are 4 calls enqueued for sub-frame 3. Calls 1, 2, and 3 were able to get the required numbers of RBs for their TBs; whereas call 4 was unable to do so. Now time relocation is applied. The following sub-frames are searched in order to schedule RBs for call 4. In sub-frame 6, RBs for this call are found. Hence call 4 is time relocated from sub-frame 3 to sub-frame 6.

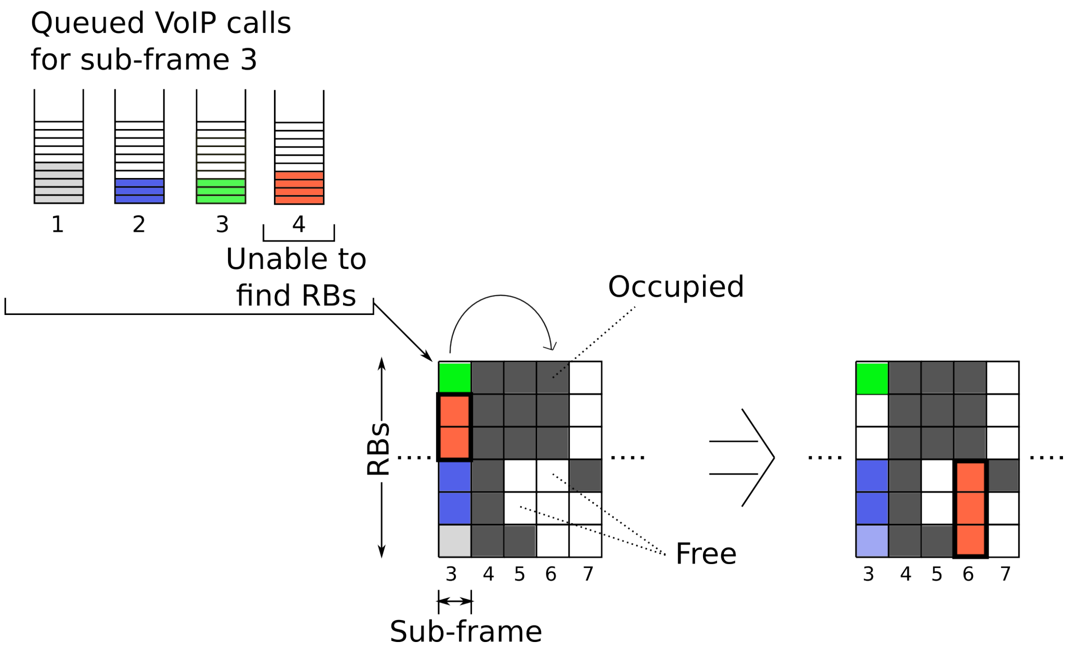


Figure 4.8: Time Relocation

CHAPTER 5

Simulation Environment

In this thesis project all the simulations results were obtained by using the Open Source Wireless Network Simulator (openWNS). Section 5.1 presents a brief description of the openWNS simulation environment. In Section 5.2 the utilized VoIP scheduler that we have utilized is explained in detail.

5.1 Open Source Wireless Network Simulator

The openWNS simulator is an event driven system level simulation platform developed at the Department of Communication Networks (ComNets), RWTH Aachen University, in Aachen, Germany. It is available under the GNU Lesser General Public License (LGPL). It is available to researchers for testing and evaluation purposes.

5.1.1 OpenWNS structure

In this subsection an overview of the openWNS simulator structure is given. As depicted in Figure 5.1, the simulator can be divided into four parts: simulation platform, evaluation, configuration, and simulation framework (within which the simulation models are executed).

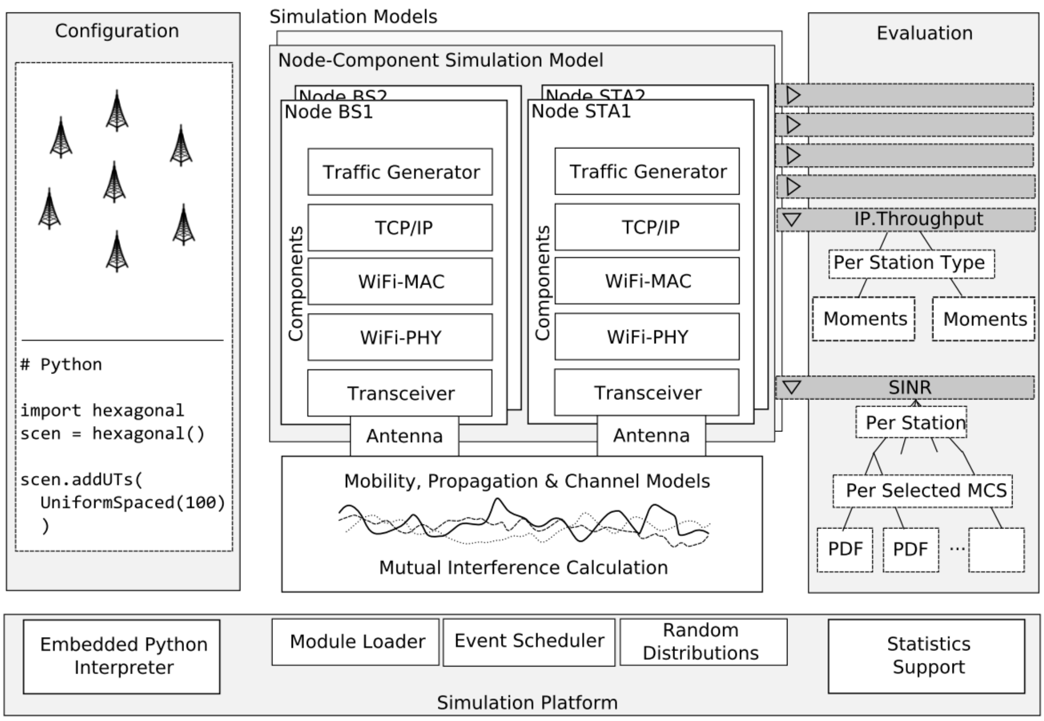


Figure 5.1: openWNS structure [9]

5.1.1.1 Simulation platform

The simulation platform forms the foundations of the simulator and supports the simulation modules. It is written in C++ and follows a modular design pattern. Figure 5.1 shows all the major components of this simulation platform. Each of these major components is briefly described below.

- **Event scheduler**: The event scheduler of openWNS can be used to directly schedule C++ functions. Both real-time and non real-time scheduling can be performed.
- **Random distribution generator**: The random number generator is based on the Mersenne Twister algorithm [21]. The available distributions are Uniform, Normal, Poisson, Ricean, Pareto, Exponential, Binomial, and Erlang distributions.
- **Statistics support**: Pre-configured measurement sources are sorted keeping in mind their context and measurement type during the simulation run.
- **Module loader**: The module loader dynamically loads the modules of the simulator.
- **Python interpreter**: The configuration of the simulation scenario, described in section 5.1.1.2, is written in the Python script language. The function of the Python interpreter is to evaluate this script. The simulation model is provided with the required parameters that can be evaluated and delivered to the relevant modules.

5.1.1.2 Configuration

The object oriented programming language Python is used for configuration of the simulation scenarios. Python has an advantage over data representation languages in terms of functionality, one of these advantages is its excellent scalability. Python is able to support scaling the configuration, thus increasing the scenario size and simulation model's complexity. Python has a clear syntax and is widely supported by the open source community.

5.1.1.3 Evaluation

The measurement sources pre-defined in the configuration are evaluated during the simulation. The main functions of the evaluation sub-system are measurement sorting based upon a measurement's context and data compression by computing statistics over the measurements made during a simulation run.

5.1.1.4 Simulation framework

The openWNS simulator provides a simulation framework that facilitates development of protocol stacks and simulation models. In order to facilitate these developments openWNS offers a well defined and clear interface, a vast set of pre-defined protocol building blocks, and code reuse helps users achieve their targets. All the major parts of the framework are described below.

- **Simulation model**: The simulation model has the capacity to support simulations ranging from a simple queuing system to complex scenarios consisting of numerous Basestations (BSs) and multiple User Terminals (UTs) with suitably equipped protocol stacks. Each simulation model has two basic methods: *start()* and *shutdown()*. When the *start()* method is executed, the simulation model sets up the scenario and starts scheduling events using the event scheduler. The *shutdown()* method completes the gathering of the simulation results and properly shutsdown the model at the end of a simulation run.

- **Node Component model**: A node component model allows the flexible specification of protocol stacks. A node is made up of components as was shown in Figure 5.1. These components represent the protocol layers, equivalent to the protocol layers of the International Organization for Standardization (ISO)/Open System Interconnections (OSI) reference model. Normally each simulator module defines a specific component type which can be instantiated inside a node.
- **Simulation modules**: There are several modules included in openWNS that form the basis of different protocol layers. These modules are in accordance with the ISO/OSI reference model layers and are namely the Physical Layer (PHY), Data Link Layer (DLL), Network Layer (NL), and Transport Layer (TL). Modules for a traffic generator and an interference calculator are also present.

In the past openWNS was used to evaluate Cell Spectral Efficiency (CSE) and Cell Edge User Spectral Efficiency (CEUSE) for IMT-A evaluation [3] and moreover in [26] [24] openWNS was analytically validated. A more detailed and comprehensive description of the openWNS structure can be found in [9] and [22].

5.2 Persistent VoIP scheduler

During this Master's thesis project a scheduler within the LTE Data Link Layer (DLL) of the openWNS simulator called the Persistent VoIP scheduler was analysed in great detail. This scheduler provides functionality for manipulating scheduling maps on a per sub-frame basis.

5.2.1 Scheduling types

Before going into the scheduler details, it is important to mention that there are three types of packets involved in the scheduling process: voice Protocol Data Unit (PDU), Silence Description (SID) PDU, and HARQ transmissions. VoIP traffic comes in bursts with a packet being sent periodically - typically every 20 ms (during an active period, i.e., during a talk spurt) and during silence periods a SID packet is sent every 160 ms. The VoIP traffic model is explained in detail in Section 6.2. Furthermore, in case of failed data reception at the receiver end, data must be retransmitted over the link using HARQ.

For the above mentioned packet types, three strategies can be employed namely: dynamic, persistent, and semi-persistent scheduling.

5.2.1.1 Dynamic scheduling

In dynamic scheduling all voice PDUs, SID PDUs, and HARQ transmissions are dynamically scheduled dynamically. PDCCH resources are consumed every time scheduling is performed. Frequency selective scheduling and fast (per TTI) link adaptation are major benefits of dynamic scheduling. However, these gains are obtained at the cost of considerable PDCCH resource consumption in the downlink and high Channel Quality Indicator (CQI) overload in the uplink [28].

5.2.1.2 Persistent scheduling

Persistent scheduling is similar to circuit switched reservations for VoIP. Persistent resource allocation and MCS allocation is done for voice PDUs, SID PDUs, and HARQ transmission. Although this is the simplest scheduling strategy and offers the lowest signalling overhead, it is unable to adapt to actually resource usage, thus resulting in limited system capacity (since resource are allocated irrespective of need).

5.2.1.3 Semi-persistent scheduling

The concept of semi-persistent scheduling is very straight forward: perform persistent scheduling for the initial voice PDU transmission and dynamic scheduling for SID PDUs and HARQ transmission. The aim of semi-persistent scheduling is to schedule voice PDUs less frequently, while maintaining the same allocation over an extended period of time. For each individual voice call, persistent scheduling is valid for the entire active period of a talk spurt. In this case PDCCH resources are required only when a call enters the active state, thus saving a considerable amount of PDCCH resources which can be utilized for other calls (hence increasing system call capacity).

5.2.2 Modules

There are four key modules of the persistent VoIP scheduler, as shown in Figure 5.2.

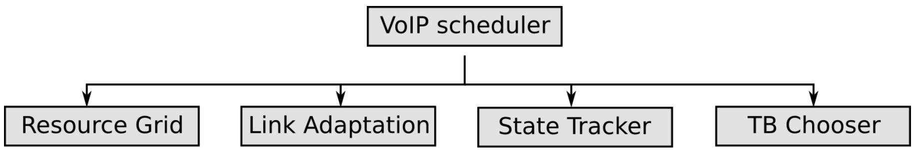


Figure 5.2: VoIP scheduler

5.2.2.1 Resource grid

The LTE resource grid structure units were explained in Section 2.1.5.4. The purpose of resource grid module is to manage all the resource units involved in the scheduling process. In a 5 MHz bandwidth scenario there are 25 Resource Blocks (RBs) which are the basic resource allocation units. Transport Blocks (TBs) are entities which carry the data for one user and their size varies from one RB to more depending upon the channel's condition. Empty RBs between TBs are called the RB space. In these spaces new reservations can be performed. Moreover, a frame consists of 20 sub-frames to exploit the 20 ms inter-arrival times typical of voice PDUs.

Tasks carried out by resource grid module include tracking of all persistent and dynamic resource reservations. Furthermore, the grid module also keeps a record of the released resources that were formerly reserved resources for inactive calls.

5.2.2.2 Link adaptation

An accurate link performance prediction is the basis of good link adaptation and this task is carried out by the Link Adaptation module. This module's basic role is to aid the resource grid module in estimating the effective SINR of a potential TB. Finally, a candidate set of potential TBs for every RB space is generated. This module works for both the uplink and downlink.

5.2.2.3 State tracker

The state tracker is a key part of the VoIP scheduler. The function of this module is to keep track of each call's state, specifically when the call transitions from the active to inactive state or the other way around. "Otherframe calls" are calls of other sub-frames present in the frame under consideration. These are a special type of calls which are time relocated to this sub-frame. The concept of time relocation was explained in detail in Section 4.4.

5.2.2.4 TB chooser

The TB chooser implements the different TB fitting strategies. A particular TB from the set of candidate potential TBs is chosen on the basis of the particular fitting strategy used for scheduling. Fitting strategies were discussed in detail in Section 4.2.

5.2.3 Role of PDCCH

As mentioned earlier, PDCCH resources are required for the scheduling of calls which causes the PDCCH to play a central role in the scheduling process. Chapter 3 gave depth insights into the structure and of the various CCE aggregation formats of PDCCH. A unique CCE aggregation format is assigned to a TB depending upon the current channel condition leading to varying number of schedulable TBs per sub-frame. In this way PDCCH resources define a limit and whenever this limit is reached, no additional TBs can be scheduled within the sub-frame, hence with the exception of persistent calls which only require PDCCH resources at the initial time of entering the active state all other allocations of resources will require use of PDCCH resources.

5.2.4 Defined priority

One of the major challenges faced in scheduling is to prioritize different types of calls. This feature is incorporated in the VoIP scheduler. The call scheduling priorities are ranked as

- Persistent call scheduling,
- HARQ transmission, and
- Dynamic call scheduling: SID and other sub-frame calls.

This particular ordering produces the best performance results as it will be shown in Chapter 6.

5.2.5 Algorithm description

The overall operation of the scheduler is elaborated in Figure 5.3. As mentioned in Section 5.2.4. the highest priority is voice PDU scheduling. At the start of every sub-frame all the active calls are sorted and the systems attempts to schedule them. In case of additional RBs are required because of more robust MCS, the strategy of frequency relocation is adopted. Similarly the time relocation is applied if RBs are not available in the current sub-frame, but are available in the following sub-frames. All of the above concepts were explained in detail in Chapter 4. The next step is the dynamic scheduling of HARQ re-transmissions. Following this is the scheduling of SID PDUs. Finally other voice PDUs are scheduled. PDCCH resources are constantly being monitored and scheduling is halted whenever the maximum resource limit is reached.

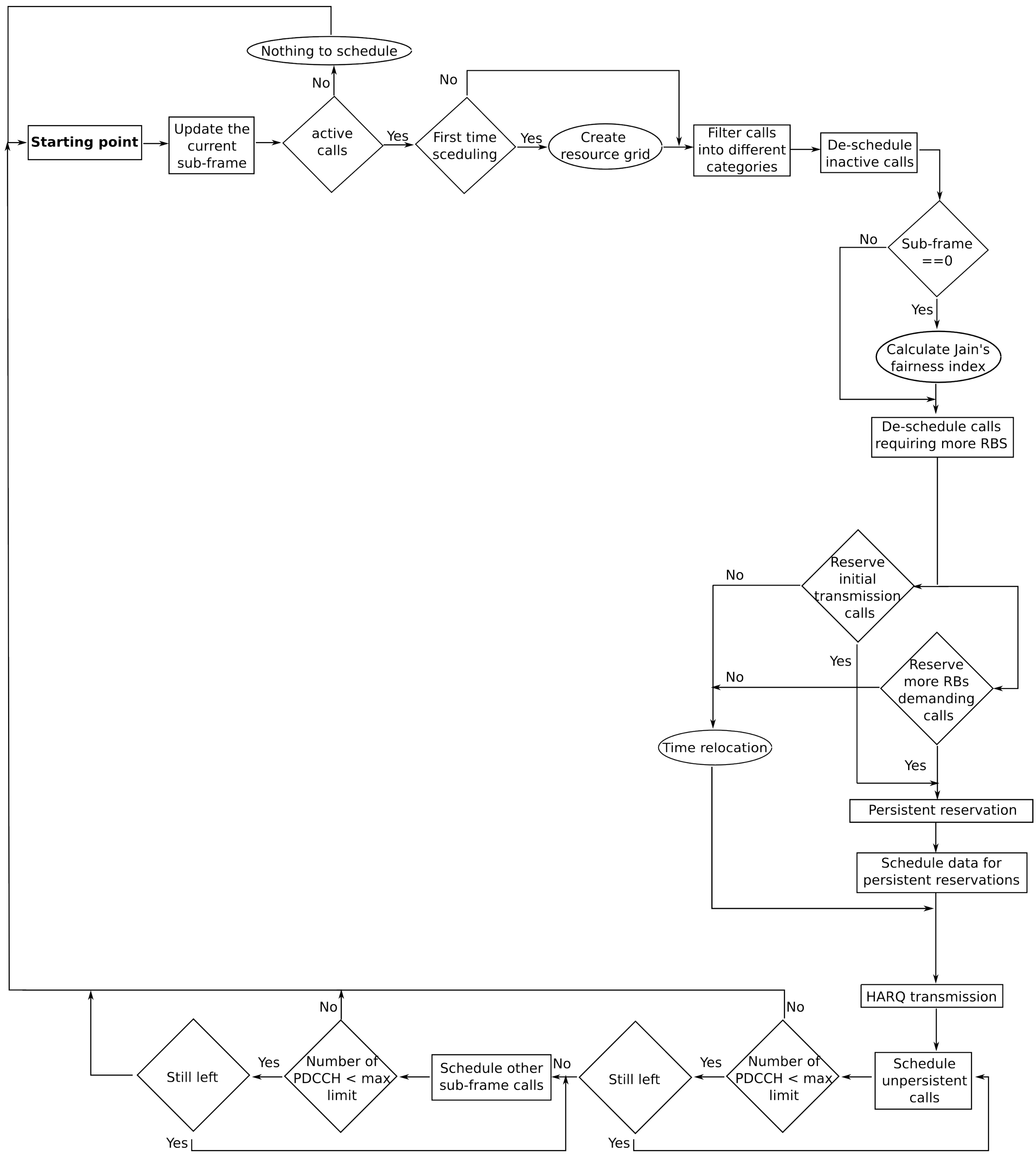


Figure 5.3: Control flow of scheduler

CHAPTER 6

Simulation scenario and results

In this chapter the strategies presented in Chapter 4 are simulated. The goal is to show the key results obtained, while utilizing realistic assumptions. This chapter presents the LTE specific parameters and the scenario configurations used in our evaluation of the different strategies.

6.1 IMT-Advanced evaluation methodology

The International Mobile Telecommunication Advanced (IMT-A) specification issued by the International Telecommunication Union Radiocommunication Sector (ITU-R) for mobile phone and internet access services specifies features far superior to earlier standards. These IMT-Advanced specifications [18] define what the ITU considers the requirements of fourth generation (4G) mobile wireless broadband technologies.

6.1.1 Evaluation guidelines

The requirements and evaluation guidelines of IMT-Advanced are essential as not only do they ensure consistent evaluation, but they also defined the **minimum** performance level that is considered acceptable for 4G. Different methodologies are applied to assess the distinct evaluation characteristics shown in Figure 6.1.

6.1.2 Test environments

Four test environments indoor, microcellular, base coverage urban, and high speed are defined by the ITU-R. These test environments along with their deployment scenarios are shown in Table 6.1.

Table 6.1: Deployment scenario for evaluation

Test environment	Indoor	Microcellular	Base coverage urban	High speed
Deployment scenario	Indoor hotspot scenario	Urban micro-cell scenario	Urban macro-cell scenario	Rural macro-cell scenario

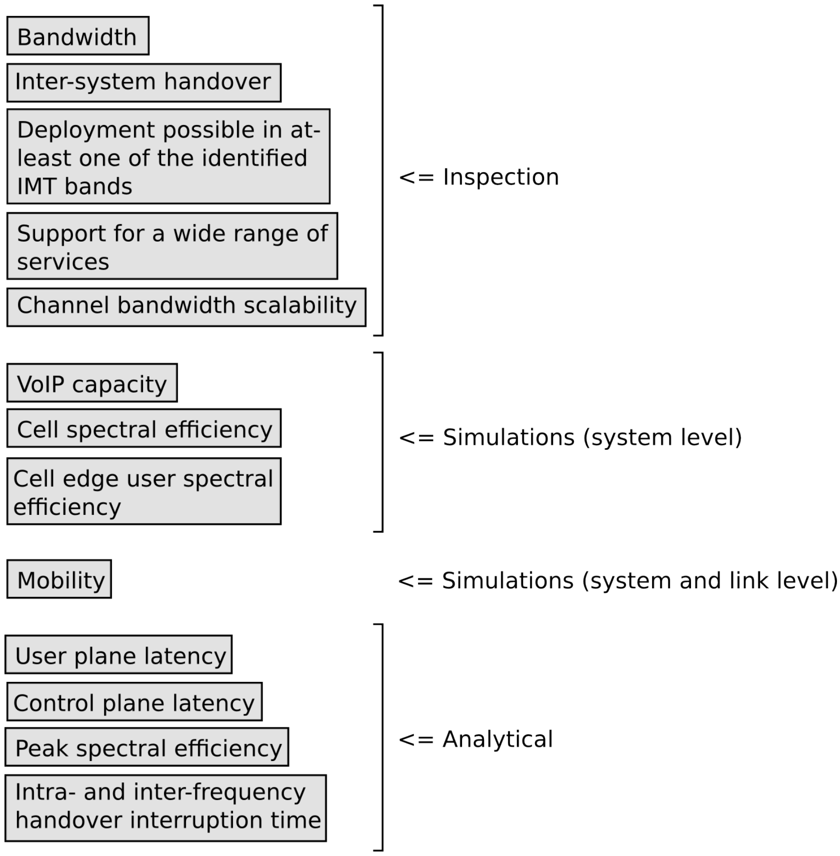


Figure 6.1: Methodology of assessing different parameters

6.1.3 Indoor Hotspot Scenario (InH): VoIP capacity

In this thesis project the emphasis is on the VoIP capacity in an Indoor hotspot scenario. This scenario primarily targets an indoor environment offering isolated cells and hotspots serving stationary and pedestrian users. The main focus of this scenario are small cells and high user throughput in homes, offices, and other buildings. The environment consists of a rectangular area having dimensions of 120 m by 50 m. There are two BS sites present with the coordinates shown in Figure 6.2. These BSs each have an omnidirectional antenna.

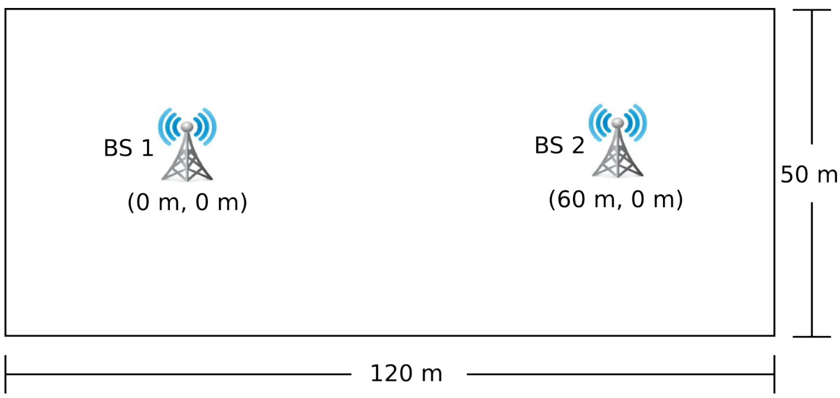


Figure 6.2: IMT-Advanced Indoor Hotspot scenario

Indoor hotspot scenario is considered because high user density is mostly observed indoor. However, investigation of VoIP capacity can be done for the outdoor scenarios mentioned in Table 6.1.

6.1.4 Parameters for system simulation

Table 6.2 shows the parameters which were applied in our system level simulation. These parameters are in accordance with our scenario of interest, i.e, the indoor hot spot scenario. Moreover, results obtained from this scenario can be extended to the other scenarios as well. Small-scale fading as described in the ITU-R evaluation guidelines [18] was not used. Channel with small-scale fading is out of the scope of this thesis and is recommended for further investigation in the future work in this topic. Control channel resources (PDCCH) are required for the scheduling of user connections. However, these resources are limited. Hence in order to see how PDCCH resource availability affects the overall performance for VoIP users, the simulation results are evaluated both with and without this constraint.

Table 6.2: General simulation parameters

Parameters	**Values**
Layout	Indoor floor
Inter-site distance	60m
Channel model	Indoor hotspot model (InH)
User distribution	Random & uniformly distributed over area
Inter-site interference modelling	Explicitly modelled
BS noise figure	5dB
UT noise figure	7dB
BS antenna gain	0dBi
UT antenna gain	0dBi
Thermal noise level	-174 dBm/Hz
Evaluated service profiles	VoIP
Simulation bandwidth	5 + 5 MHz (FDD)
Simulation time span for a single drop	20 s

6.2 VoIP traffic model

The IMT-Advanced VoIP traffic model is a two state model. This model is shown in Figure 6.3.

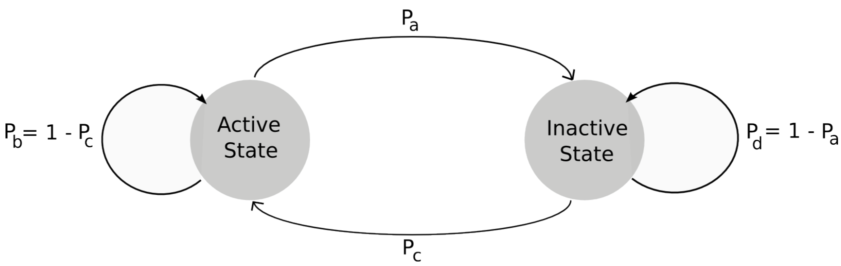


Figure 6.3: VoIP traffic model

According to this model VoIP traffic comes in bursts with a packet being sent periodically every 20 ms (during an active period, i.e., during a talk spurt), during silence periods a Silence Indicator Description (SID) packet is sent every 160 ms. The packets must be delivered from the sender to the receiver within a bounded period (typically less than 180 ms from mouth to ear). The choice of encoder/decoder (CODEC) determines the duration of an audio frame. For CODECs such as G.711 each audio frame is 20 ms in duration. The duration of the audio frame leads to a deterministic transmission interval of packets, hence there is a deterministic inter-arrival time (IAT). Today, a popular family of CODECs are Adaptive Multi rate (AMR) CODECs. These CODECs utilize a voice activity detector to only transmit encoded audio when there is useful content to send, hence the transmission is discontinuous. At the receiver side comfort noise is generated so that the listener is presented with noise roughly similar to the background noise during talk bursts. Table 6.3 summarizes the characteristics of VoIP traffic according to the IMT-A VoIP model.

Table 6.3: VoIP traffic characteristics

Parameter	**Characteristics**
Codec	Source rate = 12.2 kbits/s
PDU Inter Arrival Time (Active State)	20 ms, 344 bit MAC PDU
PDU Inter Arrival Time (Silent State)	160 ms, 144 bit MAC PDU
Voice Activity Factor (VAF)	50 % (P_a= 0.01, P_d= 0.99 P_c= 0.01, P_b= 0.99)

Evaluating the capacity of a system (for example, a cellular telephony cell) is generally done in terms of the number of users in the cell being served when more than 98% of these users are satisfied. It is assumed that a user is satisfied if more than 98% of their speech frames are delivered successfully within 50 ms. This criteria is according to the guidelines for evaluation of radio interface technologies for IMT-Advanced [18].

6.3 Simulation Results

In this section simulation results are presented. The scenarios were configured using the parameters listed in Table 6.4.

Table 6.4: Simulation configuration parameters

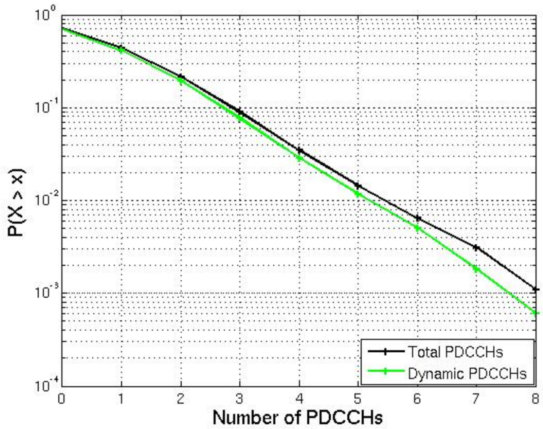
Parameters	Values
Settling time	2 s
Maximum simulation time	22 s
Maximum number of nodes	910

6.3.1 Scenarios considered

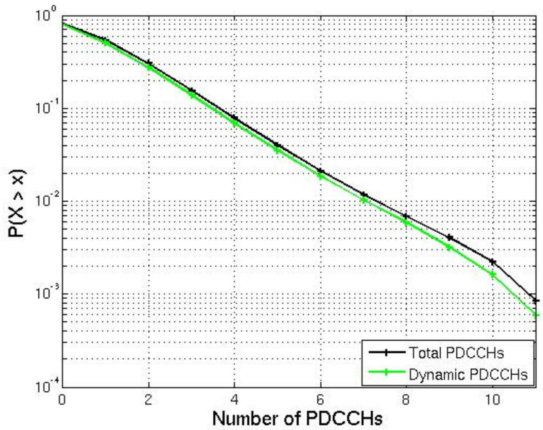
In this thesis, all simulations were carried out while considering three different PDCCH models. The first scenario has unlimited PDCCHs. This is a hypothetical conjecture as in reality only limited PDCCH resources are available in each sub-frame. The purpose of this scenario is to illustrate the results with out restriction on the control channel resources when scheduling of voice calls; this scenarios provides an upper bound on the number of simultaneous VoIP calls that the system can handle. The second scenario has the realistic PDCCH limit of 11. This limit is based on the calculations in Chapter 3, where it is shown a maximum of 11 PDCCH units are available per sub-frame. In this case, the actual aggregation CCE formats of 1, 2, 4, and 8 are allocated to the calls depending upon the downlink channel conditions. Finally, the last scenario has a PDCCH limit of 8. This is according to 3GPP simulation assumptions and only aggregation the CCE format 1 is used for all the calls–irrespective of their channel condition. The last scenario, as explained in 3GPP TS 36.814 [4] is considered so that the results of this thesis project are comparable with current research where the limit of 8 is used by and large. Furthermore, fitting strategies least fit, first fit, and random fit (outlined in section 4.2) are adopted for the simulation of each scenario. Results corresponding to each of these fitting strategies are presented later in this section.

6.3.2 Number of PDCCH utilized

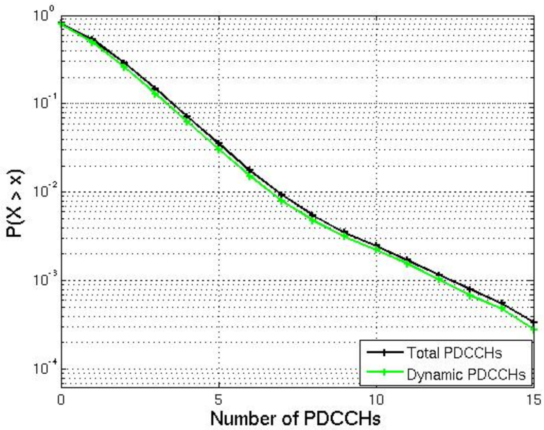
It is interesting to see how PDCCH resources are utilized by different types of TBs in the scheduling process. As mentioned in Chapter 4, PDCCH resources are mainly consumed by persistent setup TBs, persistent relocated TBs, and dynamic TBs. Simulations have shown that PDCCH resources employed for dynamic TBs make up most of the total utilized PDCCHs, thus rendering the effect of other PDCCHs utilization almost negligible. Figure 6.4 shows dynamic PDCCHs versus total PDCCHs for all the three scenarios for 870 nodes.



***(a)** PDCCH limit 8*



***(b)** PDCCH limit 11*



***(c)** No PDCCH limit*

Figure 6.4: Total PDCCHs versus Dynamic PDCCHs

From above discussion it is clear that PDCCHs plays a key part in the scheduling of dynamic TBs and their availability or non-availability has a strong impact on the overall performance of dynamic scheduling. Dynamic TBs are further composed of three types of components:

$$\text{Dynamic TBs} = \text{HARQ TBs} + \text{other frame TBs} + \text{SID TBs}$$

Figure 6.5 shows the individual contribution of each of these components.

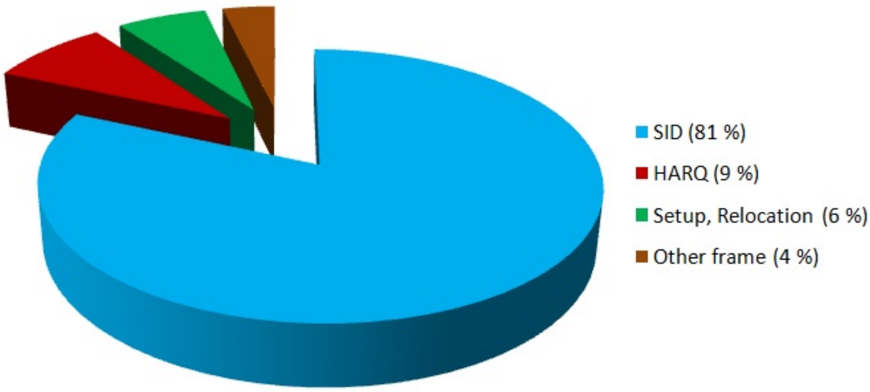


Figure 6.5: PDCCH components

6.3.3 Required aggregation format

Aggregation format (number of CCEs) allocated to a UT depends on the downlink Wideband Signal to Interference Noise Ratio (WB-SINR) [11] [16]. Assuming the block error rate (BLER) target to be 0.01, then the SINR values above which a aggregation format can be adopted are shown in Figure 6.6. A single CCE is allocated to a call if the SINR value is approximately above 4.0 dB, 2 CCEs if SINR values is between 4dB and 0.5dB, 4 CCEs if the SINR value is between 0.5 and -2.2 dB, and finally 8 CCEs if the SINR value is below -2.2 dB.

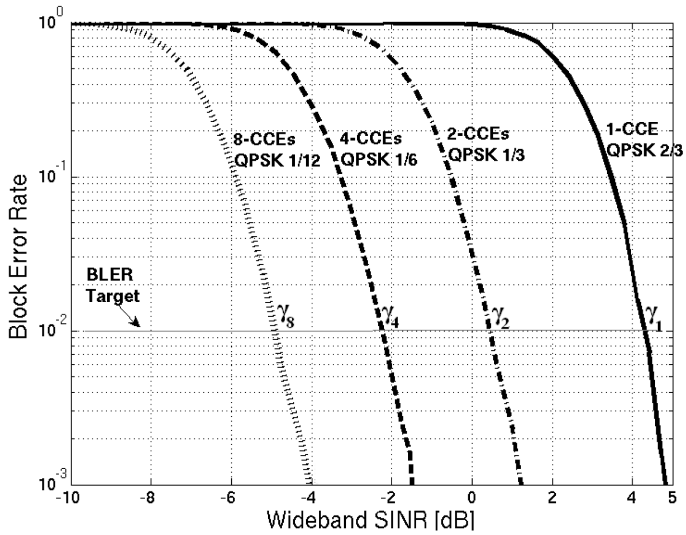


Figure 6.6: BLER vs SINR [11]

Simulation results show that most of the TBs need aggregation format 1 and atleast some TBs require aggregation format 8 for the scheduling. Furthermore, the aggregation format allocation factor is vital for the realistic PDCCH limit scenario, as in the PDCCH limit 8 scenario only the aggregation format 1 is allocated; while for the unlimited PDCCH limit scenario this factor has no effect (due to unlimited amount of available resources). Figure 6.7 shows results obtained for 870 nodes.

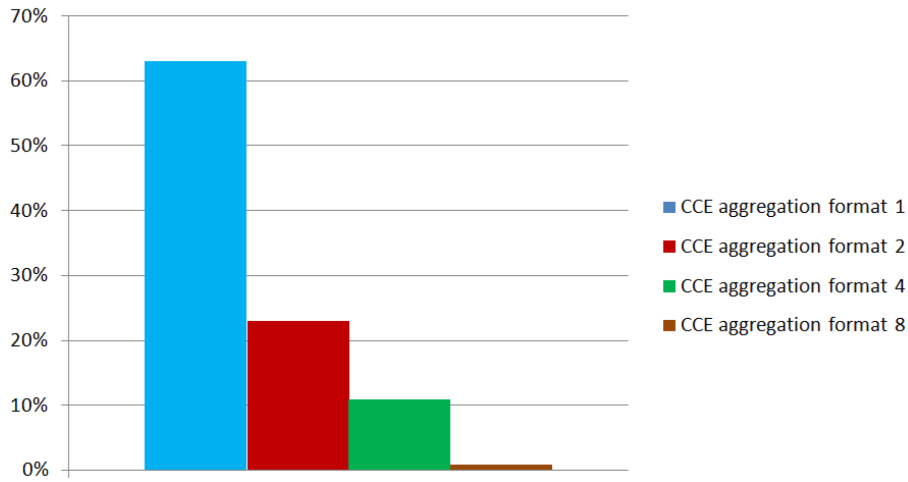
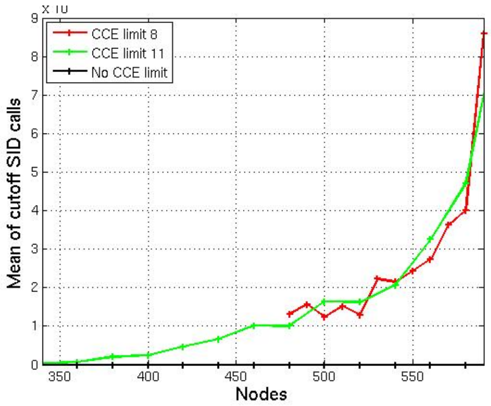


Figure 6.7: CCE aggregation format distribution

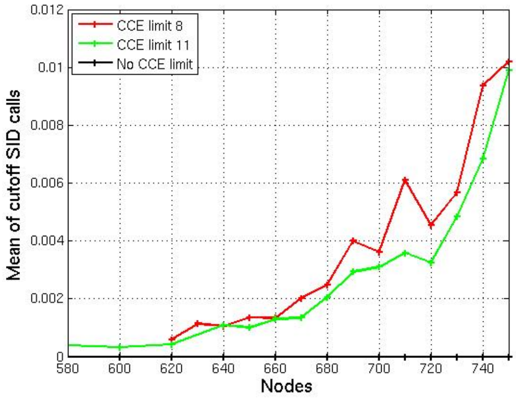
6.3.4 Impact of PDCCH limit

In case of the unlimited PDCCH model, the control channel resources do not limit the number of schedulable TBs in a sub-frame. Figures 6.8 and 6.9 illustrates this where all TBs are scheduled. The situation will change when PDCCH constraints are taken into account, resulting in left over unscheduled TBs. In Section 4.2.4 the scheduling priority was discussed. According to the scheduling priorities, persistent voice calls are scheduled first followed by HARQ, SID, and otherframe calls scheduling, respectively. Due to this priority setup of calls and HARQ transmission rarely encounter a shortage of PDCCH resources. The main effect of limited PDCCH resources is on the scheduling of SID and otherframe calls. Figures 6.8 and 6.9 illustrate this very fact for SID and otherframe calls.

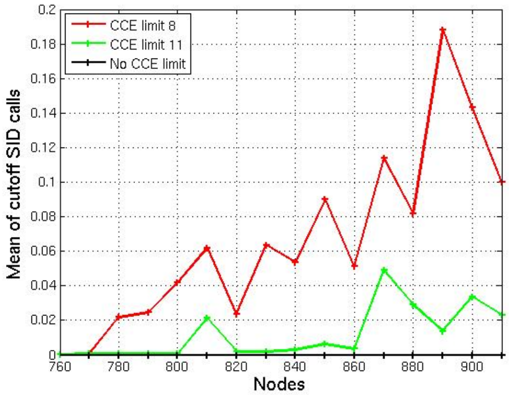
From Figures 6.8 and 6.9 it is clear that until a certain number of nodes are active in the cell there are no cutoff calls. After this threshold is passed, then cuttoff calls occur for both scenarios involving PDCCH limits of 8 and 11. Moreover, the limit after which cutoff calls appear depends upon the fitting strategy that is adopted.



(a) *PDCCH limit 8*



(b) *PDCCH limit 11*

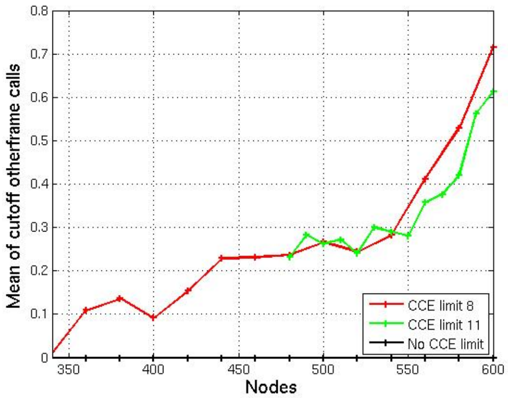


(c) *No PDCCH limit*

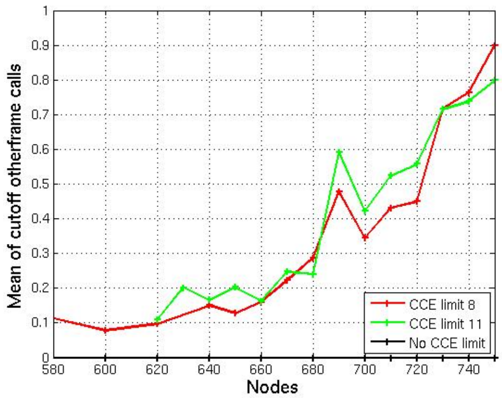
Figure 6.8: Cutoff SID calls

6.3.5 User Satisfaction

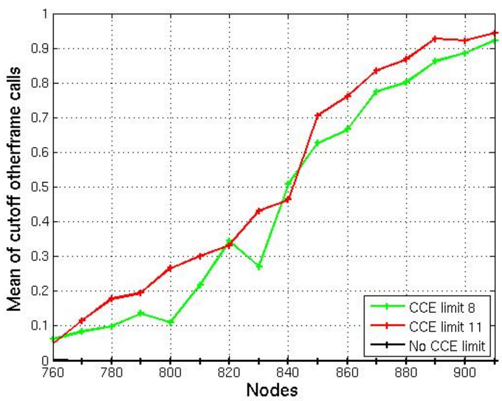
Figure 6.10 shows the user satisfaction curves for all three models with different fitting strategies. Fifty simulation runs were performed for each number of nodes. Error variance in results is very low with a large confidence interval. It can be seen that least fit strategy gives the best performance, followed by random and first fit strategies, respectively. This result is as expects, since the least fit is the most sagacious fitting strategy; while first fit results in the most interference. In random fit, the interference is equally distributed across all RBs. In least fit, RBs with low interference are favoured resulting in short TB sizes. Moreover, for a particular fitting strategy different user satisfaction curves are obtained for the three scenarios. We consider the first fit strategy first. In this case, the user satisfaction is about 100% until there are more than roughly 540 nodes. When this threshold is crossed the user satisfaction deteriorates for different PDCCH limits at different rates. For unlimited, unrealistic, and realistic PDCCH limits, the user satisfaction criteria (98 %) is maintained until about 610 nodes, 602 nodes, and 590 nodes (respectively). Further increasing the number of nodes leads to a user satisfaction below the desired 98%. Similarly in the case of random fit strategy. Finally, for the least fit strategy, the impact of PDCCH limits is most evident, with the 98% threshold maintained up till 835, 860, and 875 users for PDCCH limit 8, 11, and unlimited respectively. All three fitting strategies along with the three PDCCH limit scenarios are shown in Figure 6.10.



(a) *PDCCH limit 8*

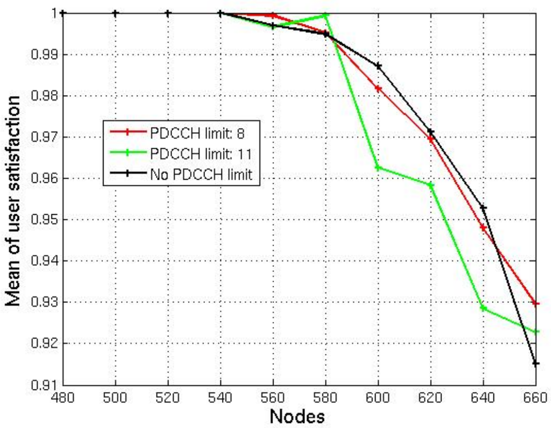


(b) *PDCCH limit 11*

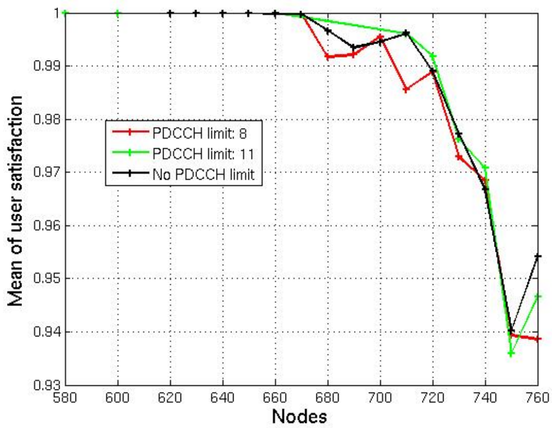


(c) *No PDCCH limit*

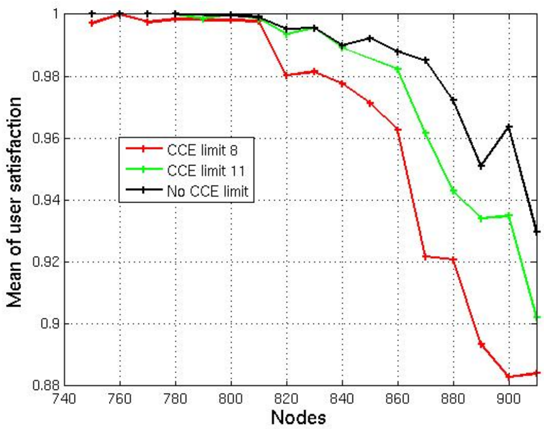
Figure 6.9: Cutoff otherframe calls



(a) *First fit*



(b) *Random fit*



(c) *Least fit*

Figure 6.10: User satisfaction

CHAPTER 7

Conclusions and future work

This chapter presents some conclusions followed by suggestions of future work.

7.1 Summary and Conclusions

In this thesis, the IMT-Advanced Indoor Hotspot scenario was considered in order to evaluate LTE VoIP capacity. After describing the basic LTE concepts, the control channel resource limitation for the scheduling of voice calls was considered. Based on this the maximum number of schedulable users in a sub-frame was calculated. Moreover, semi-persistent and PDCCH models were discussed and subsequently used to evaluate the system's performance. The openWNS simulator was used to simulate different scenarios. According to results presented in Chapter 6, the following conclusions are drawn:

- Control channel constraints restrict the number of schedulable calls in a sub-frame, thus effecting the overall system capacity;
- Dynamic scheduling consumes most of the available PDCCH resources;
- Most of the TBs require PDCCH aggregation format 1 for scheduling purpose; and
- VoIP capacity is reduced when the control channel's limitations are taken into account.

7.2 Future work

Based on the research and implementation done in this thesis project, some aspects that should be interesting to investigate in the future are:

- Consider the impact of small scale fading and
- Identify and take special care of users experiencing the worst channel conditions.

Another interesting aspect to investigate would be the three dimensional deployment of this indoor hotspot scenario. In this case the inter BS interferences and user scheduling will give rise to unique and harder challenges as compared to the two dimensional indoor hotspot scenario. Further, the results presented in this thesis are based on a SIMO system which does not consider the opportunities of MIMO systems. Therefore, an implementation of a MIMO model and an evaluation of VoIP capacity in such a system would provide additional information concerning the performance of the resource scheduling strategies discussed in this thesis under realistic control channel constraints.

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