Load balancing between LTE and WiFi

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To my family and friends

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Abstract

This work addresses Load Balance between LTE and Wi-Fi. A simulator was developed in order to evaluate the performance of the algorithms and the performance of the network when Wi-Fi APs are introduced. This evaluation was done by varying parameters such as user density, user distribution, and penetration of services, priority table, load threshold, bandwidth and number of BSs and APs. The introduction of Wi-Fi APs is responsible for the decrease of the load (up to 5% decrease of LTE's load), which may have a huge impact on the QoS in higher load situations, in all scenarios and also for the increase in throughput of users Throughout all simulations, average delay always presents values below the maximum accepted value, taking a maximum value of 20ms.As for handovers, a linear dependency with the density of users is observed and observed a higher number of handovers when more APs are introduced. Its failure rates were always below 1%. Drop rate presented a maximum value of around 7%. This value is higher than the maximum accepted one and it is explained by the fact that the scenarios contemplated a cluster where a lot of users are out of coverage and/or in the cell edges. Although results presented only contemplate scenarios with low load, the obtained values for the performance parameters are lower than the maximum acceptable values, thus leading to believe that the model could be applied to scenarios with higher load and still having those values below the acceptable ones.

Keywords

LTE, Wi-Fi, Load Balance, Algorithm, Quality of Service, Energy Efficiency

Resumo

Este trabalho tem como foco o balanceamento de carga entre LTE e Wi-Fi. Foi desenvolvido um simulador para avaliar o desempenho dos algoritmos bem como o desempenho da rede, quando introduzidos os pontos de acesso de Wi-Fi. Essa avaliação foi feita variando vários parâmetros, como densidade de utilizadores, a sua distribuição, penetração dos serviços, tabela de prioridade, o limiar de carga, largura de banda e número de estações bases e pontos de acesso. A introdução dos pontos de acessos foi responsável pela diminuição da carga (até 5%), o que poderá ter impacto na qualidade de serviço em situações de maior carga, e também pelo aumento nos ritmos binários dos utilizadores. O atraso médio apresentou sempre valores abaixo do valor máximo aceitável e apresentou um valor máximo de 20ms. Existe dependência linear com a densidade de utilizadores para o handover, verificando-se um aumento do número destes aquando da introdução dos pontos de acesso de Wi-Fi. As taxas de falha deste parâmetro foram sempre inferiores a 1%. A taxa de queda apresentou um valor máximo de 7%, mais elevado que o máximo aceitável, no entanto é explicado pela cobertura considerada e pelo posicionamento dos utilizadores nas bordas das células. Embora os resultados apresentados sejam para cenários com uma carga baixa, os valores obtidos para os parâmetros de performance são mais baixos do que os máximos aceitáveis, levando a acreditar que o modelo apresentado poderá ser aplicado a cenários de maior carga e que os valores de performance estejam abaixo dos máximos aceitáveis.

Palavras-chave

LTE, Wi-Fi, Balanceamento de Carga, Algoritmo, Qualidade de Serviço, Eficiência Energética

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List of Acronyms

3GPP	3rd Generation Partnership Project
AAA	Authentication Authorisation Accounting
ACK	Acknowledgment
AMBR	Aggregated Maximum Bit Rate
ANACOM	Autoridade Nacional de Comunicações
ANDSF	Access Network Discovery and Selection Function
AP	Access Point
ARP	Allocation and Retention Priority
BE	Best Effort
BK	Background
BSS	Basic Service Set
CAC	Call Admission Control
CL	Controlled Load
CP	Cyclic Prefix
CRC	Cyclic Redundancy Check
CRRM	Common Radio Resource Management
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
CTS	Clear to Send
DCF	Distributed Coordination Function
DFS	Dynamic Frequency Selection
DIFS	DCF Inter Frame Space
DL	Downlink
DS	Distribution System
DSMIPv6	Dual Stack Mobile IPv6
DSP	Digital Signal Processing
DSSS	Direct Sequence Spread Spectrum
EDGE	Enhanced Data Rates for GSM Evolution
EE	Excellent Effort
EIRP	Effective Isotropic Radiated Power
eNB	Evolved Node B
EPC	Evolved Packet Core Network
ePDG	Evolved Packet Data Gateway
EPS	Evolved Packet System
ETSI	European Telecommunications Standards Institute

E-UTRAN	Evolved UMTS Terrestrial Radio Access Network
FDD	Frequency Division Duplex
FHSS	Frequency Hopping Spread Spectrum
FQDN	Fully Qualified Domain Name
FTP	File Transfer Protocol
GBR	Guaranteed Bit Rate
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HHO	Horizontal Handover
HO	Handover
HSDPA	High Speed Downlink Packet Access
HSPA	High Speed Packet Access
HSPA+	High Speed Packet Access Evolution
HSS	Home Subscriber Server
HSUPA	High Speed Uplink Packet Access
HTTP	Hypertext Transfer Protocol
IEEE	Institute of Electrical and Electronics Engineers
IP	Internet Protocol
IPOM	IP Flow Mobility
IPv6	Internet Protovol Version 6
ISD	Inter Site Distance
ISI	Inter Symbol Interference
ISRP	Inter System Routing Policies
ITS	Intelligent Transportation Systems
LB	Load Balance
LLC	Logical Link Control
LoS	Line of sight
LTE	Long Term Evolution
M2M	Machine to machine
MAC	Media Access Control
MAPCON	Multiple Access PDN Connectivity
MBR	Maximum Bit Rate
MIH	Media Independent Handover
MME	Mobility Management Entity
NC	Network Control
NLoS	Non line of sight
OFDM	Orthogonal Frequency Division Multiplexing
OFDMA	Orthogonal Frequency Division Multiple Access
PBCH	Physical Broadcast Channel
PCF	Point Coordination Function

PCRF	Policy Control and Charging Rules Function
PDCCH	Physical Downlink Control Channel
PDN	Public Data Network
PDSCH	Physical Downlink Shared Control Channel
P-GW	Packet Data Network Gateway
PHY	Physical Layer
PIFS	PCF Inter Frame Space
PRACH	Physical Random Access Channel
P-SS	Primary Synchronisation Signal
PUCCH	Physical Uplink Control Channel
PUSCC	Physical Uplink Shared Control Channel
QCI	QoS Class Identifier
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
R99	Release 99
RAT	Radio Access Technology
RB	Resource Block
RE	Resource Element
REF	Reference
RLC	Radio Link Channel
ROHC	Robust Header Compression
RRM	Radio Resource Management
RRU	Radio Resource Unit
RS	Reference Signal
RTP	Real Time Protocol
RTS	Request to Send
SAE	System Architecture Evolution
SC-FDMA	Single Carrier Frequency Division Multiple Access
S-GW	Serving Gateway
SIFS	Short Inter Frame Space
SMS	Short Message Service
SON	Self-Organising Network
S-SS	Secondary Synchronisation Signal
TDD	Time Division Duplex
UDP	User Data Protocol
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunications System
VHO	Vertical Handover
VI	Video

VO	Voice
VoIP	Voice over IP
Wi-Fi	Wireless-Fidelity
WLAN	Wireless Local Area Network
WWW	World Wide Web

List of Symbols

α_i	Coefficient value that models the autoregressive process
α_{pd}	Average power decay
γ	Distance exponent
τ	Average delay
τ_{TTI}	Time transmission interval
Ψ	Street orientation angle
B_N	Noise bandwidth
B_{RB}	Bandwidth of RB
d	Distance between BS and MT
d_B	Building separation
d_{break}	Breakpoint distance
D_d	Time interval between two consecutive packets inside a packed call
$D_{handover}$	Handover delay
D_{pc}	Reading time between two consecutive packet call requests in a session
D_r	Drop rate
F	Noise figure
f	Frequency
$g_i(n)$	Gaussian random variable
G_r	Receiving antenna gain
G_r	Gain of the receiving antenna
G_t	Transmitting antenna gain
G_t	Gain of the transmitting antenna.
h_B	Building height
h_{BS}	BS height
h_{MT}	MT height
k	Boltzmann’s constant
L_0	Free space attenuation
L_c	Losses in the cable between the transmitter and the antenna
L_{LTE}	Load index in LTE
L_{msd}	Multi-screen diffraction loss
L_{ori}	Attenuation caused by main street orientation
L_p	Path loss

$L_{p,total}$	Total path loss
L_{rts}	Roof-to-street diffraction and scatter loss
L_u	Losses due to the user
L_{Wi-Fi}	Load index in Wi-Fi
M	Modulation’s order
M_{FF}	Fast fading margin
m_i	Estimated average value of transmission speed
M_p	Margins
M_{SF}	Slow fading margin
N	Average noise power
N_d	Number of packets within a packet call
N_{fd}	Number of delayed frames
N_{HHOa}	Number of attempts to perform a horizontal handover
N_{HHOf}	Number of failed horizontal handovers
N_{pc}	Number of packet call requests per session
N_{pt}	Total number of packets transmit
N_{RB}	Number of resource blocks
N_{RB}	Number of RBs
$N_{RB/U}$	Number of resource blocks per user
N_{sd}	Total number of dropped sessions
$N_{served\ users/BS}$	Number of users served per base station
N_{st}	Total number of sessions
$N_{streams}$	Number of streams
$N_{symbols/subframe}$	Number of OFDM symbols per sub-frame
N_U	Number of users
NU_{srv}	Average number of users per service per second
N_{VHOa}	Number of attempts to perform a vertical handover
N_{VHOf}	Number of failed vertical handover
O^{cum_S}	Total output parameter cumulative mean
O^{cum_s}	Partial output parameter cumulative mean at simulation run *s*
P_{EIRP}	Effective isotropic radiated power
P_{HHOf}	Percentage of failed horizontal handovers
P_r	Power available at the receiving antenna
$P_{r_{sub-carrier}}$	Receiving power
P_{Rx}	Power at the input of the receiver
Ps	Packet size
P_{Tx}	Transmitter output power
$P_{TX_{sub-carrier}}$	Transmitted power

P_{VHOf}	Percentage of failed vertical handovers
$R_{b,global}$	Mean value of the Average bitrate per RAN
$R_{b,peak}$	Peak bit rate
$R_{b,RAN}$	Average value of the bitrate per RAN per second
$R_{b,service}$	Average value of the bitrate per service per second
$R_{b,user}$	Bit rate per user
$R_{b/U}$	Physical Layer Bit Rate per user
r_{cell}	Cell radius
s	Simulation run index
S	Total number of simulations done
T	Temperature of the receiver
T_{IU}	Uncertainty of acquiring the first available random access occasion
T_{margin}	Implementation margin
t_{OFF}	Mean Silent Phase
t_{ON}	Mean Active Phase
$T_{processing,RRC}$	Time to received message and produce a response
T_{search}	Time required to identify target cell
v_{av}	Average velocity
v_{max}	Maximum velocity
v_{min}	Minimum velocity
w	Street width

List of Software

Microsoft Excel	Spreadsheet application
Microsoft Visio	Diagramming and vector graphics application
Microsoft Visual Studio	Integrated development environment
Microsoft Word	Text editor tool

Chapter 1

Introduction

This chapter gives a brief overview of the work. The context, main motivations, work targets and scope are established. At the end of the chapter, the work structure is also presented.

Mobile communications systems have come a long way, while revolutionising the way people communicate. The first mobile communication systems appeared in the 80s. They were analogue and provided only voice. Although being analogue and having many limitations, these systems experienced a rapid growth and each country developed its own system, which was incompatible with everyone else's equipment and operation.

Europeans realised this was an undesirable situation and therefore, to solve the limitations presented by analogue systems, came with the Global System for Mobile Communications (GSM), also known as Second Generation system (2G), which was published in 1990 by the European Telecommunications Standards Institute (ETSI). Although the focus of GSM was a global standardisation within Europe, its success exceeded all expectations, and thus GSM became the most adopted worldwide mobile communications standard. GSM is completely digital and its main characteristics are the safety, robustness, reliability, efficient use of spectrum and support of data transmission services for low bandwidth services. Still in 2G, General Packet Radio Service (GPRS) was introduced in order to improve data rates and to allow transmission of small amounts of data, despite of the previous development of Short Message Service (SMS). After GPRS, Enhanced Data rates for GSM Evolution (EDGE) appeared, providing enhanced data rates with the efficient use of modulation. Figure 1.1 shows the 3GPP schedule for the commercial deployment of this technology and the ones that followed it, so far.

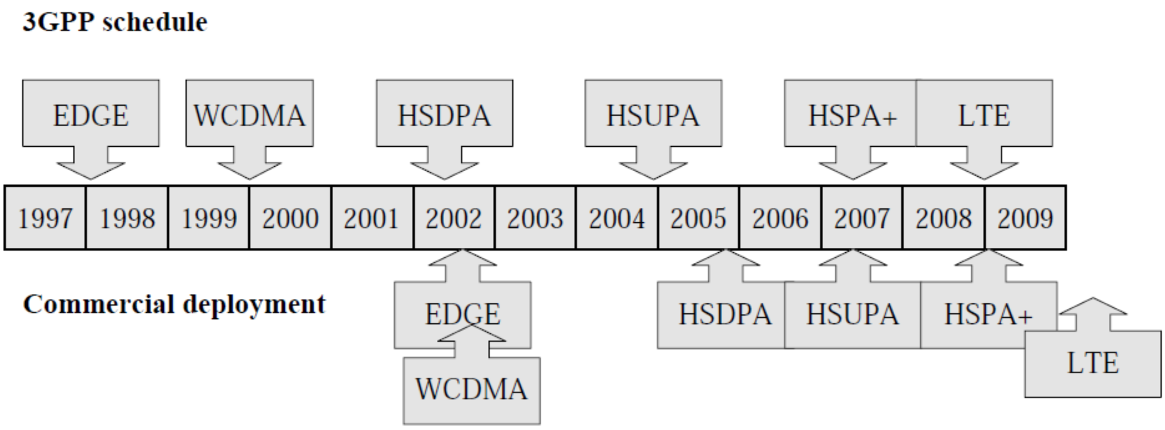


Figure 1.1 Schedule of 3GPP standard and commercial deployment (extracted from [HoTo09]).

In 1999, the Third Generation Partnership Project (3GPP) launched the Universal Mobile Telecommunications System (UMTS), also known as Release 99 (R99). UMTS was deployed and its goals were, among others, enhanced person-to-person communication, high-quality file transmission, and higher data rates. As users start to demand higher data rates, 3GPP specified, in 2005, a solution called High Speed Downlink Packet Access (HSDPA) and after it, High Speed Uplink Packet Access (HSUPA) in 2007. These solutions are known as High Speed Packet Access (HSPA). Moreover, HSPA Evolution (HSPA+) was specified in order to meet the continuous demands of users.

Long Term Evolution (LTE), also specified by 3GPP, was presented in order to meet present and future users' demands. LTE's main performance targets were higher spectral efficiency, higher peak user throughput, reduced latency, optimised packet switched, high level of mobility and security, optimised terminal power efficiency, and frequency flexibility thought the use of new digital signal processing (DSP) techniques and modulations. A further goal of this system was the redesign and the simplification of the network architecture to an IP-based system.

Going back, in 1985, 802.11 technology had its origin. Thereafter, around the time that 2G systems appeared, Institute of Electrical and Electronics Engineers (IEEE) released the standard for Wireless Local Area Networks (WLANs), IEEE 802.11, which defined all the basic principles required to deploy WLANs. Although WLANs have several benefits over wired networks, WLANs were less secure, less reliable and provided lower throughput. In order to compensate for these factors, several amendments were introduced to IEEE 802.11 specifications. Nowadays, amendments continue to be implemented in order to increase system's performance. In 1999, the Wi-Fi (Wireless Fidelity) Alliance was formed as a trade association to hold the Wi-Fi trademark.

As time went by, services started to impose new challenges to networks, as they have stringent delay and throughput requirements that must be fulfilled, i.e., one needs to have Quality of Service (QoS) guarantees provided by the network and therefore amendments continue to be defined.

However, the evolution of these systems is not enough to satisfy users' demands. As shown in Figure 1.2, mobile traffic is growing exponentially. Therefore, service providers must evolve and adjust to meet those demands.

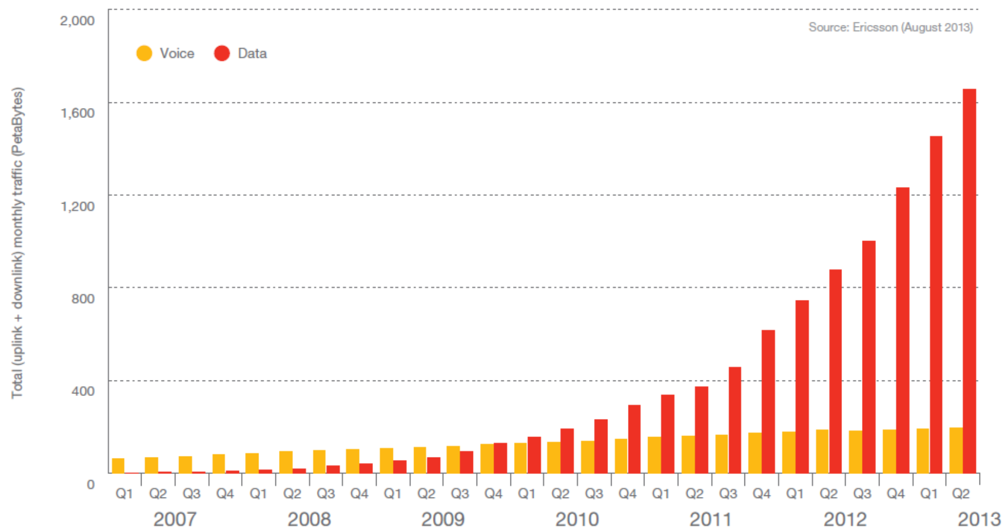


Figure 1.2 Global total data traffic in mobile networks (extracted from [Eric13b]).

Since the expensive radio spectrum for cellular networks prohibits rapid deployment, and the low bandwidth restricts system capacity, the next performance and capacity leap will come from network topology evolution by using a mix of macro and small cells.

The most efficient way to use small cells is to position them in locations where significant amounts of data are generated, and where subscribers spend most of their time, and therefore consume significant amounts of data. In many situations LTE coverage is superimposed with Wi-Fi hotspots, and since around 80% of all traffic is generated indoors, most of the phones have built-on Wi-Fi access (around 80% of all smartphones) [NSN13], and smartphones is the main device consuming data, according to CISCO forecasts (presented in Figure 1.4), Wi-Fi can be a good solution to offload traffic from macro cells. Besides this, the number of hotspots tend to grow, as it can be seen in Figure 1.3, making this solution a favourable solution for operators.

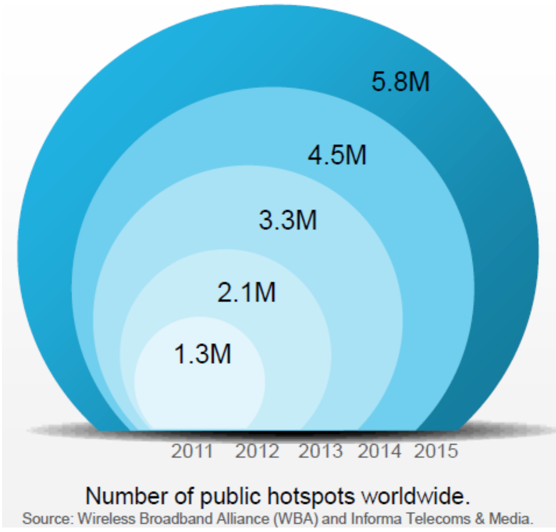


Figure 1.3 Number of public hotspots worldwide (extracted from [Qual13]).

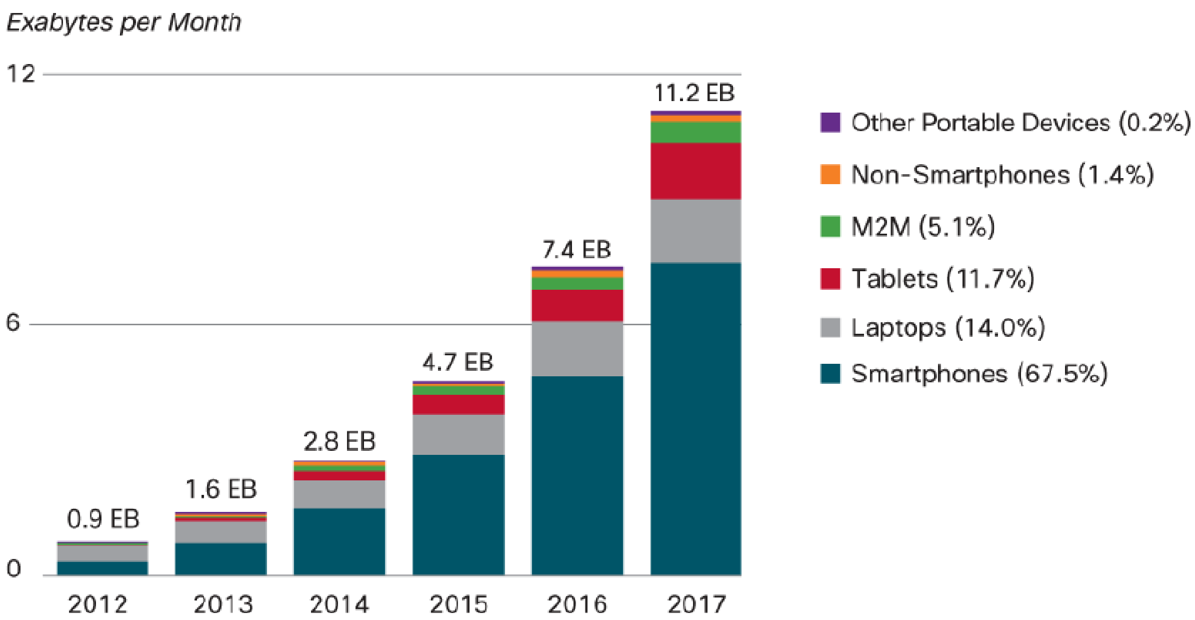


Figure 1.4 Cisco forecasts of mobile data traffic (adapted from [CISC13]).

Although currently this superimposition is not taken into consideration, certainly that operators intend to take advantage of these aspects in order to provide the best QoS possible for the different services demanded by users, and want to explore and deploy this solution.

Widespread existing deployments, availability of user devices that support technology, cost efficient, capability to address new users and devices without mobile subscription, globally available spectrum capacity, operating in license free frequency bands and standards availability for integration into mobile core networks are some of the advantages of using Wi-Fi to offload mobile data traffic from macro cells.

This thesis is composed of 5 chapters, and an appendix formed by 4 annexes. In this chapter, one presents an overview and the motivation for this thesis.

Chapter 2 presents a brief overview of LTE and Wi-Fi systems. It addresses fundamental concepts of both systems, interworking in between both, basic concepts of load balance and services and applications. The state of the art is also presented.

For Chapter 3, the load balance algorithms were implemented as well as some other parameters needed for its purpose.

In Chapter 4, the results obtained by the simulations are presented. Seven different situations are simulated, each consisting of characterising the response of the network to the variation of a single input parameter. Results are interpreted, explained and compared with theoretical ones.

Finally, Chapter 5 presents a final conclusion and future research topics.

Chapter 2

Fundamental Concepts and State of the Art

This chapter provides an overview of LTE and Wi-Fi systems, describing the fundamental concepts that are relevant for this thesis, as well as the interworking between both. Load balancing is also presented as well as a brief overview of services and applications of both systems. Finally, the state of the art concerning the scope of this thesis is also presented.

2.1 The LTE System

This section presents an overview of LTE's basic concepts. System's architecture is presented, followed by radio interface.

2.1.1 Network architecture

LTE network architecture is presented in this section, being based on [HoTo09] and [3GPP11].

LTE's network is based on Evolved Packet System (EPS), comprised of the radio network, E-UTRAN, and an IP core network in order to integrate all applications over a simplified and common architecture. LTE presents a flat architecture optimised for packet switched services, Figure 2.1.

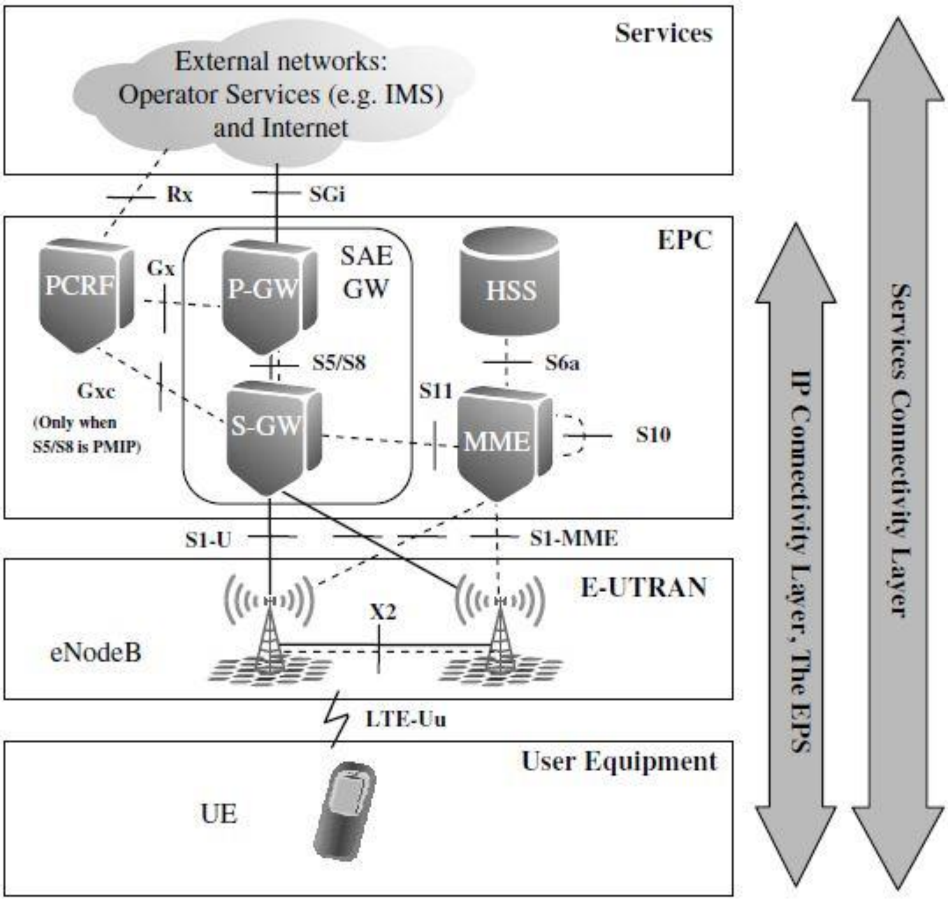


Figure 2.1 Network architecture for E-UTRAN (extracted from [HoTo09]).

As shown in Figure 2.1, the core network architecture, System Architecture Evolution (SAE), is divided into three main levels: User Equipment (UE), E-UTRAN and Evolved Packet Core Network (EPC).

LTE's radio access network (E-UTRAN) has one single element, evolved Node B (eNB). Since E-UTRAN does not have a centralised controller, eNB is responsible for IP header compression and encryption of the user data stream, radio resource control, connectivity to the EPC, transfer of paging and broadcast messages to UEs, and intra-eNB mobility control. This structure leads to reduced system complexity and cost, and to a better performance over the radio interface. Note there is an interface X2 that assures communication between eNBs. This interface is responsible for load balancing management and handover. It is also important to refer that, as illustrated in Figure 2.1, it is the S1 interface that provides connection between E-UTRAN and EPC.

EPC is composed of several entities, known as control nodes that are responsible for the overall control of the UE and bearers establishment. Those control nodes and their functions are:

- Mobility Management Entity (MME) – considered the principal control node for the radio access network, handles functions related to bearer management, security procedures, UE location management, as well as some others.
- Serving Gateway (S-GW) – user plane node that connects E-UTRAN and EPC. Responsible for intra E-UTRAN mobility, as well as mobility with 3GPP technologies such as GSM or UMTS.
- Packet Data Network Gateway (P-GW) – acts as the router between UE and external packet data networks. It is responsible for allocation of IP addresses to UE, policy enforcement, charging support and lawful interception of user traffic.
- Policy Control and Charging Rules Function (PCRF) – software element, responsible for efficient policy and charging control.
- Home Subscriber Server (HSS) – component used as database to store user's subscription data.

The focus of this thesis is load balance, where handover plays an important role. Handovers are network controlled and usually it is the eNB that triggers them, based on the measurements received from the UE. A handover can be performed inside E-UTRAN, to E-UTRAN from other Radio Access Technology (RAT), or from E-UTRAN to another RAT. Interface X2 is responsible for load balancing management and handover,, being through this interface that handover requests are sent. If X2 does not exist between source and target eNBs, procedures need to be initiated to set one up before handover can be achieved. Handover requests include information on requested SAE bearers to be handed over, handover restrictions list, and last visited cells the UE has been connected to in order to avoid the Ping-Pong effect. The core network has no control on the handover and the S1 interface is updated only when the radio handover has been completed.

2.1.2 Radio interface

This section presents an overview of radio interface aspects, being mainly based on [Agil09] and [HoTo09].

LTE operates in several arrangements of frequency and bandwidth. In Portugal, resulting from an auction of ANACOM, the Portuguese telecommunications authority, the frequency bands chosen to deploy LTE were the bands of 800 MHz, 1.8 GHz and 2.6 GHz. Table 2.1 presents the specific frequencies for each band. Although there are two types of duplex modes in LTE, Frequency Division Duplex (FDD) and Time Division Duplex (TDD), FDD was adopted across all Europe and, hence, only FDD is taken into account in this thesis.

Table 2.1 Specific frequency bands for each band (extracted from [Agil12]).

Frequency Band [MHz]	**Specific frequency band**	
	UL [MHz]	DL [MHz]
800	[832,862]	[791,821]
1800	[1710,1785]	[1805,1880]
2600	[2500,2570]	[2620,2690]

Concerning multiple access transmission schemes, Orthogonal Frequency Division Multiple Access (OFDMA) for downlink (DL) and Single Carrier Frequency Division Multiple Access (SC-FDMA) for uplink (UL) were chosen. Both schemes are well known, hence not being discussed here, Figure 2.2 showing briefly how both schemes operate. There is a cyclic prefix (CP), common to both multiple access transmission schemes used for DL and UL. It serves as a guard interval, to eliminate inter symbol interference (ISI), and as a repetition of the end of the symbol, to allow channel estimation and equalisation. CP's length must be at least equal to the length of the multipath channel in order to be effective. So, a larger CP is equivalent to a worse radio channel.

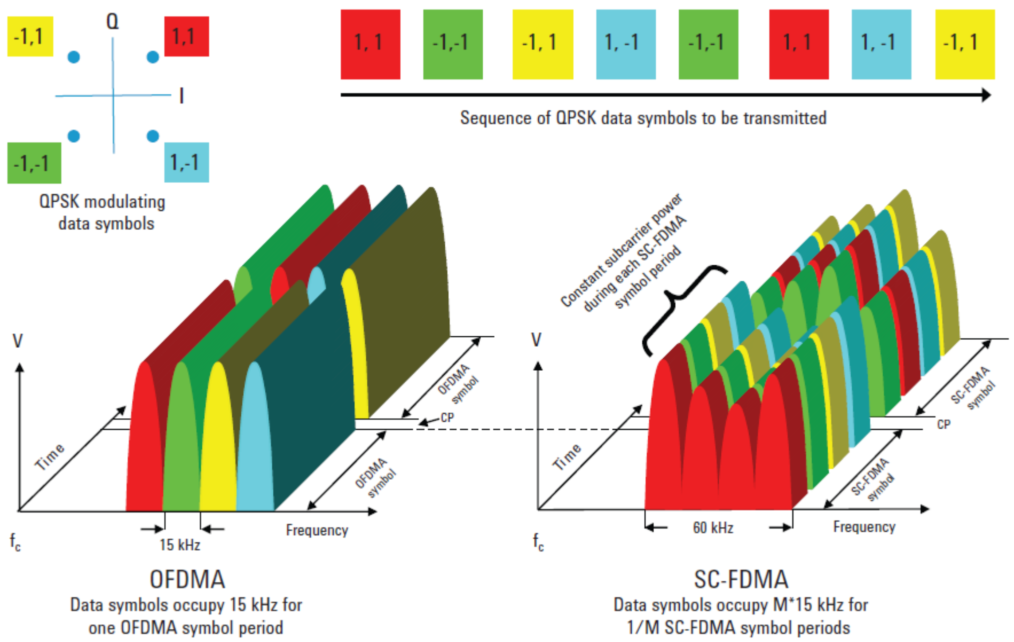


Figure 2.2 OFDMA and SC-FDMA comparison for QPSK modulation (extracted from [Agil09]).

As previously mentioned, LTE operates in several arrangements of frequencies and bandwidths, but the frame structure is always the same. One radio frame corresponds to 10 ms and has 20 slots, with

0.5 ms each. A sub-frame consists of 2 slots and its duration is 1ms.

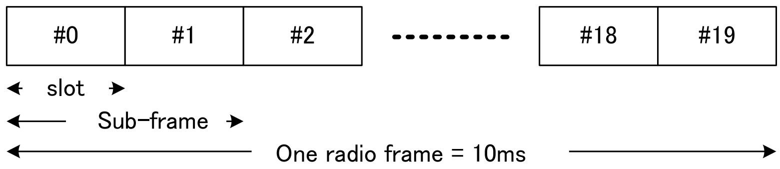


Figure 2.3 Type 1's frame structure (extracted from [3GPP11]).

LTE physical channels are based on Resource Blocks (RBs). One RB corresponds to 12 sub-carriers, 180 kHz, and also corresponds to a time slot. The number of sub-carriers depends on the channel's bandwidth, Table 2.2.

Table 2.2 RBs available for a determinate bandwidth (based on [3GPP11]).

Bandwidth [MHz]	1.4	3	5	10	15	20
Number of RBs	6	15	25	50	75	100

In both DL and UL, since the frame structure is the same, each sub-carrier carries 6 or 7 OFDM symbols, depending on whether the CP is extended or normal, respectively. RBs are composed of Resource Elements (REs), which consist of 1 OFDM symbol, having 84 or 72 REs depending on the CP length.

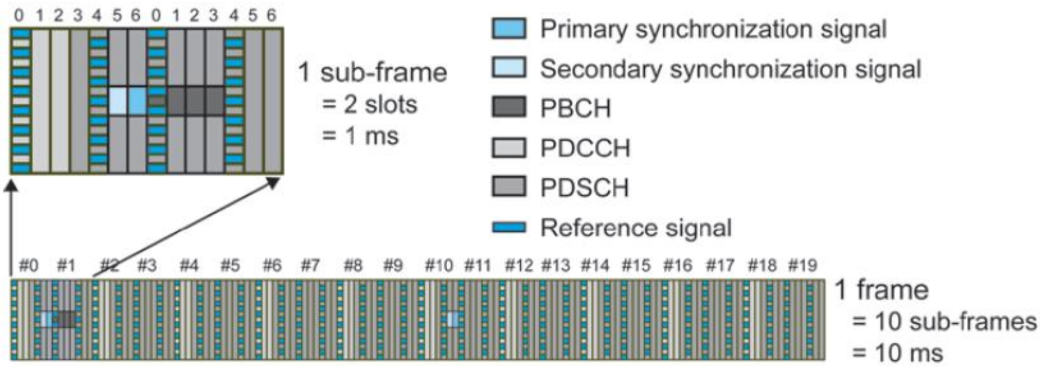


Figure 2.4 DL frame structure with normal CP (extracted from [Agil09]).

As shown in Figure 2.4, DL's frame structure is composed of physical channels and physical signals. Primary Synchronisation Signal (P-SS) and Secondary Synchronisation Signal (S-SS), which are physical signals, are used for cell search and UE network synchronisation. Besides P-SS and S-SS there is another type of physical signals, which are the Reference Signals (RS).These are used to obtain an estimation of the channel, being spread over the bandwidth. As for physical channels, Physical Broadcast Channel (PBCH) carries system information required to access the system, Physical Downlink Control Channel (PDCCH) carries information regarding resource allocation for both DL and UL for use in UE, and Physical Downlink Shared Control Channel (PDSCH) is used for data transmission and for paging information transmission.

For UL, PDCCH and PDSCH are replaced with Physical Uplink Control Channel (PUCCH) and

Physical Uplink Shared Control Channel (PUSCC) and have the same function but in the reverse way, i.e. instead of being used for DL, they are used for UL. Physical Random Access Channel (PRACH), which is a physical channel used in UL, is only used for random access transmission.

2.2 The Wi-Fi System

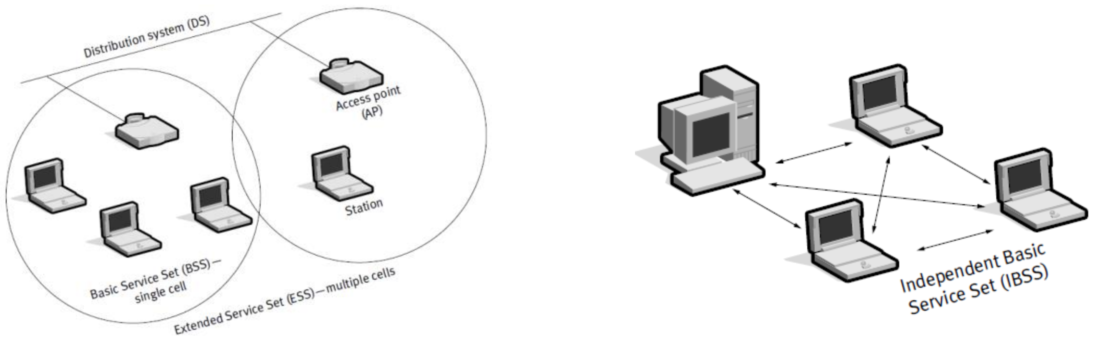
An overview of Wireless Local Area Networks (WLAN) basic concepts is provided and more specifically on IEEE 802.11, also as known as Wi-Fi.

2.2.1 Network Architecture

A brief overview of Wi-Fi network architecture is presented in this section, being based on [3COM00] and [STAL05].

WLANs were designed to be a wireless version of fixed computer networks, Local Area Networks (LANs), the main characteristics being similar to LANs' ones.

There are two main types of WLANs, ad-hoc and infrastructure networks, Figure 2.5. Although these two types exist, only the infrastructure one is considered in this thesis.



a) Infrastructure b) Ad-hoc

Figure 2.5 WLAN's architecture types (adapted from [3COM00]).

In ad hoc infrastructure, mobile units communicate directly with each other, peer-to-peer, usually being only temporary networks. In infrastructure mode, the most important and popular mode, mobile units communicate through an access point (AP) that serves as a link to other networks.

The fundamental block of the Wi-Fi architecture, in the infrastructure mode, is the Basic Service Set (BSS). A BSS contains one or more wireless stations and a central base station, the AP. The AP is connected to the distribution system (DS), which by its turn is connected to a hub, switch or router that is connected to the Internet.

Wi-Fi deployment occurs mainly indoors, which leads to a reduced coverage. A low transmitting power of the AP enhances the idea of reduced coverage. Although Wi-Fi offers a poor coverage, capacity is not an issue for this type of network, since it has a very large one comparing with other types of wireless networks. Mobility can be another problem in this type of wireless networks, since network coverage must be homogeneous in order to provide users mobility, and Wi-Fi cannot guarantee such coverage.

IEEE 802 standards' scope is the Data Link Layer and Physical Layer (PHY) of the OSI Reference Model, where in the IEEE 802 Reference Model the Data Link Layer is composed of the Logical Link Control (LLC) and Medium Access Control (MAC) and where PHY is composed of Physical Layer Convergence Procedure (PLCP) and Physical Medium Dependent (PMD). Figure 2.6 presents a comparison between both models.

Encoding/decoding of signals, preamble generation/removal for synchronisation and bit transmission/reception are some of the functions that PHY is responsible for. PLCP sublayer is responsible for defining a method to map 802.11 MAC layer protocol data units into a framing format suited for sending and receiving data and management information. PLCP uses PMD to define the characteristics and method of transmitting and receiving of data through a wireless channel.

Above PHY, in the OSI Reference Model, is the Data Link Layer. While the LLC sublayer provides an interface to higher layers and performs flow and error control, the MAC sublayer functions consists of the assemble of data into a frame with address and error detection fields, disassemble of frame, and still is responsible for address recognition and error detection, and controls access to the LAN transmission medium.

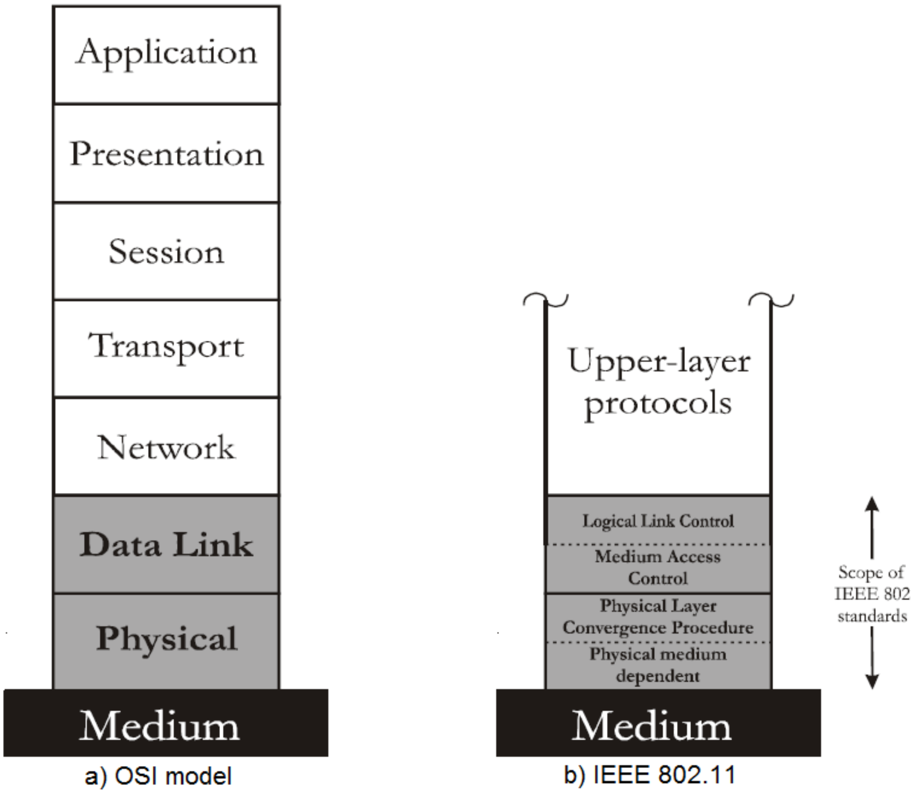


Figure 2.6 Comparison between OSI model and IEEE 802.11 layers (extracted from [Seba08]).

2.2.2 Radio Interface

This section addresses Wi-Fi's radio interface, being based on [KuRo05] and [STAL05].

Besides the original specification, there are currently five main specifications, and one of them has not yet been deployed. Table 2.3 presents the frequency, modulation and maximum throughput of those specifications thus providing a brief comparison among them. Table 2.4 presents a basic description of some amendments, also defined by IEEE, that are relevant for this thesis.

Table 2.3 802.11 network standards (adapted from [INTEL13]).

Standard	Freq. Band [GHz]	Max. Throughput [Mbps]	Modulation
802.11b	2.4	11	DSSS
802.11a	5	54	OFDM
802.11g	2.4	54	OFDM,DSSS
802.11n	2.4 / 5	150	OFDM
802.11ac	5	866.7	OFDM

For the 2.4 GHz band, there are 14 channels but only 13 are available to use, where only 3 are non-overlapping. Channels are situated between 2.401 GHz and 2.495 GHz, and are centred in 2.405+5k where k is the channel's number (k=1, 2, 3…, 14).

For the 5 GHz band, there are 24 channels available for various situations. Channels are situated between 5.180 GHz and 5.320 GHz (used for Indoor, Dynamic Frequency Selection (DFS) and Transmit Power Control (TPC)), between 5.500 GHz and 5.700 GHz (used for DFS and TPC) and between 5.745 GHz and 5.825 GHz (used for Short Range Devices (SRD)). All of these channels are separated by 20 MHz.

Table 2.4 802.11 standard and amendments description (adapted from [STAL05]).

802.11 standard	Amendment description
A	PHY, 2.4/5 GHz at 54 Mbps
B	PHY, 2.4 GHz at 11 Mbps
D	Extend operation of 802.11 WLANs to new regulatory domains
E	New MAC layer to enable QoS support and enhance security
H	Spectrum management (for Europe)
G	PHY, 2.4 GHz at 54 Mbps
N	Enhancements for higher throughput over 802.11a and 802.11g
Ac	New PHY, 866.7 Mbps, accomplished by extending air interface concepts of 802.11n

IEEE 802.11 frame structure is shown in Figure 2.7, being composed of several fields. Although there

are 4 address fields in the frame, the fourth one is only used in ad-hoc networks, and since only infrastructure networks are considered in this thesis, only the first three address fields are important. Address 1 is the MAC address of the wireless station that receives the frame. Address 2 is the MAC address of the station that transmits the frame. Address 3 contains the MAC address of the router that connects the BSS to other subnets. The Sequence control field allows the receiver to distinguish newly transmitted and retransmission frames. Payload consists of an IP datagram or an ARP packet. This field can be as long as 2312 bytes, but normally it is no longer than 1500 bytes. This frame structure includes a cyclic redundancy check (CRC) so that the receiver can detect bit errors in the received frame. Duration field includes the duration value of the channel reservation by the transmitting station. This duration value needs to include the time to transmit the data frame and an acknowledgment (ACK).

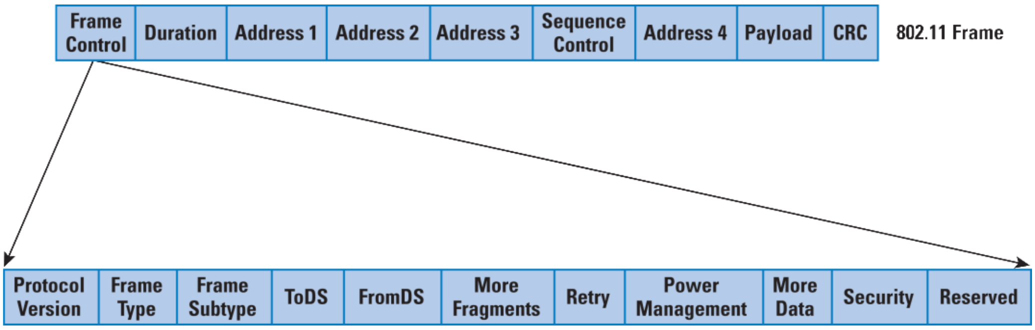


Figure 2.7 IEEE 802.11 frame structure (extracted from [CISCO08]).

As shown by Figure 2.7, the Frame control field is composed of many other subfields. Frame type and subtype fields are used to differentiate the type of the WLAN frame. Control, Data and Management are various frame types defined by IEEE 802.11. ToDS and FromDS fields, indicate if the frame is destined to the DS or if it is leaving the DS. More Fragments field indicates whether the information is split in various frames or not. Retry field indicates if the frame is a retransmission or not. Power Management informs if the station is in sleep mode or not. More Data informs if a station has more data to send, and the Security field indicates whether encryption is used or not.

As any wireless network, Wi-Fi networks are subject to unreliability due to several motives. With this unreliability, a frame exchange protocol had to be defined. This protocol establishes that when a station receives a data frame, it must return an ACK to the station that sent the frame. If this does not happen, the station that sent the frame will retransmit that very same frame. In order to enhance reliability, Request to Send (RTS) frame and Clear to Send (CTS) frame were added to that protocol.

IEEE 802.11 standard defines a required distributed access protocol, provided by MAC, where the station has the power of decision concerning transmission. For this propose, Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) is used. This mechanism is implemented through the Distributed Coordination Function (DCF), presented in Figure 2.8.

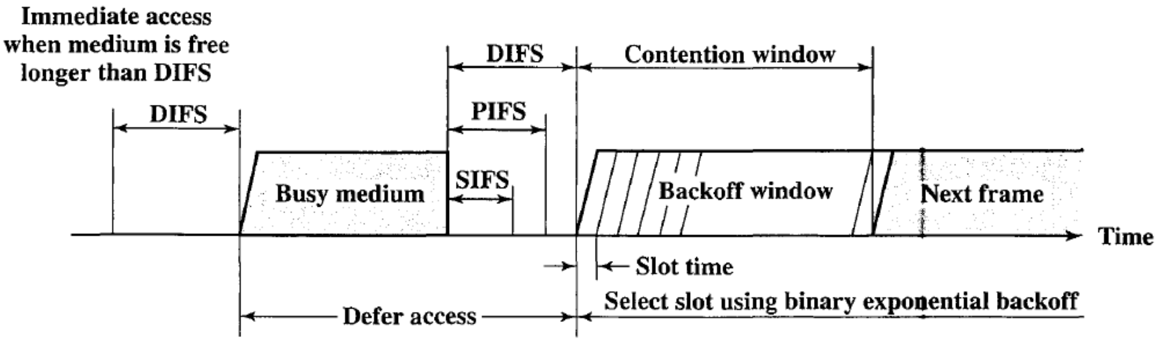


Figure 2.8 DCF function (extracted from [STAL05]).

2.3 Interworking in between LTE and Wi-Fi

A brief overview of interworking in between LTE and Wi-Fi is presented in this section, being mainly based on [HoTo09].

Although not implemented yet, Figure 2.9 presents realistic network architecture for 3GPP and non-3GPP access network.

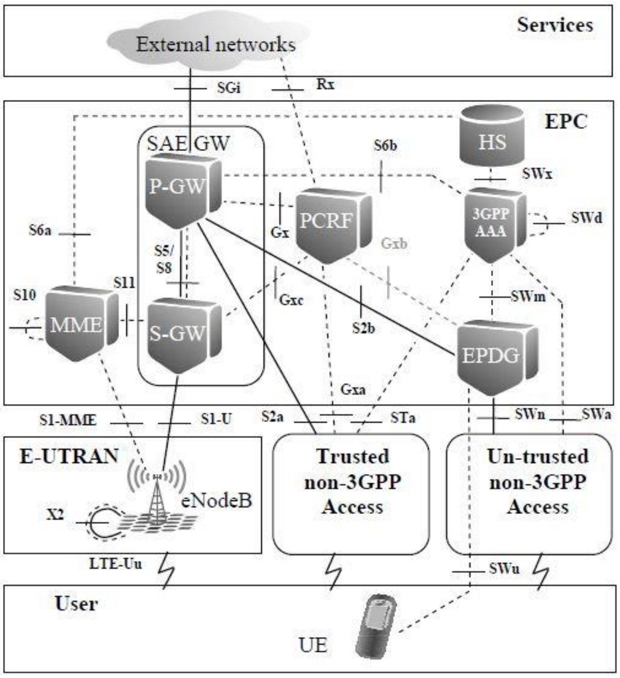


Figure 2.9 Network architecture for 3GPP and non-3GPP access networks (extracted from [HoTo09]).

Regarding LTE's EPC, presented in Section 2.1.1, this interworking system implements more control nodes. The main changes are in the P-GW, PCRF, HSS and S-GW, with new elements being introduced, such as the ePDG and the 3GPP AAA.

- Evolved Packet Data Gateway (ePDG) – responsible for interworking between EPC and untrusted non-3GPP networks, such as Wi-Fi or femto-cell access networks. It is also responsible for some security control because untrusted accesses are involved.

- 3GPP Authentication Authorisation Accounting (AAA) – control node responsible for authentication of the user in the 3GPP network, authorisation that the UE can establish communication with the system and other UE, and accounting which tracks network resource consumption by users for various purposes. In other words, this node enables control over which users are allowed to access which services, and the amount of resources they have used.
- P-GW – Besides the functions described regarding this element in LTE, in the interworking system, it is this element that supports mobility between LTE and trusted non-3GPP networks.
- PCRF – Supports Policy and Charging Control (PCC) interfaces for non-3GPP ANs.
- HSS – Besides performing similar function to those explained before, non-3GPP ANs do not interface on the AN level, and thus the selected P-GW needs to be stored in the HSS and retrieved when UE mobility involves a non-3GPP AN.
- S-GW – this node behaves as an anchor for non-3GPP ANs in case of roaming in non-3GPP.

Trusted non-3GPP and untrusted non-3GPP ANs are implemented in this architecture. Trusted non-3GPP ANs refers to networks that can be trusted to run 3GPP defined authentication, typically other mobile networks, such as CDMA2000 and WiMAX, while untrusted non-3GPP ANs are typically Wi-Fi, LTE metro or femto-cell networks.

Note that untrusted non-3GPP ANs do not perform other functions besides delivery of packets. This delivery is performed by a secure tunnel established between UE and the ePDG via a specific interface. Furthermore, the P-GW has a trust relationship with the ePDG and neither node needs to have secure association with the untrusted non-3GPP AN itself. Although UE performs authentication and authorisation with the ePDG, it may, optionally, connect to the AAA server to authenticate the UE already in the non-3GPP AN level.

EPC is designed to be access-independent, and thus it can support common service delivery and session mobility to LTE and Wi-Fi, which will lead to sophisticated traffic management, managed offload techniques and policy-based use cases. With the interworking of these systems, resources of the two networks can be viewed as a shared resource pool and their management is an essential research issue.

Note that interworking between LTE and Wi-Fi ANs requires that the UE supports both radio technologies and mobility procedures. This interoperability will allow a mobile terminal (MT) to dynamically use the multiple network interfaces available, in order to maximise user satisfaction and system performance. Therefore, users do not have to turn on or off Wi-Fi, as they do at the moment.

To take full advantage of this interworking, vertical handover between both systems is essential. The most important issue in handovers between both systems is the preservation of the IP address in order to maintain the connection while being transferred between cells. To this end, Internet Engineering Task Force (IETF) designed Mobile IP, thus allowing UE to connect to other IP radio access, while keeping the connection to the EPC through tunnelling of IP packets.

In order to be able to receive packets while the UE is being handed over to another network, Mobile IP introduces Home Agent (HA) entity to P-GW. The function of this entity is to associate the original IP address with the local address in the target network, and then forward packets from one to the other.

There are two mobility concepts in EPS, which are client-based and network-based. In the former, UE is responsible for mobility signalling and movement detection, while in the latter, signalling and detection of UE movement is the responsibility of the network. However, both rely on the existence of P-GW. An example of network-based protocol is the Proxy Mobile IPv6 (PMIPv6). PMIPv6 can provide, without client involvement, handover of the IP address between different access types. P-GW is responsible for anchoring the session, assigning the IP addresses, and switching the PMIPv6 tunnels between different access gateways if handover occurs.

For client-based, an example is the DSMIPv6 protocol, which is used between the UE and the appropriate P-GW and assigns a virtual IP address. The same IP address is assigned to the UE over LTE in the event of handover. LTE is treated as the home network, thus, not being necessary to set up a DSMIPv6 tunnel in this network. When it comes to dealing with an untrusted WLAN network, meaning that P-GW and HSS does not trust the security of the WLAN, an IP Security Protocol (IPSec) encrypted tunnel has to be established between UE and ePDG.

2.4 Load Balance

In LTE, the concept of self-organising networks (SON) is introduced. In this concept, contrary to what is practiced in many existing networks, parameters are adjusted automatically based on measurements in order to achieve a high level of network operational performance. Moreover, there is the need to further improve network efficiency. In order to achieve that extra performance, the use of load-balancing (LB) is essential to solve the unequally load distribution over cells and to provide users with the required quality of service needed. The main idea of LB is to relocate part of the users from overloaded cells to less loaded cells by adjusting the network control parameters, in such a way that overloaded cells can offload the excess traffic and this way provide the best QoS possible to all of them.

One important element in LB is the function used to measure the balance between systems. For this purpose, the load index is used, which indicates the quantity of resources of a system that are available or being used. Load indexes for LTE and Wi-Fi are presented below in (2.1) and (2.2) respectively.

$$L_{LTE} = \frac{\sum N_{RB,user}}{N_{RB}} \tag{2.1}$$

$$L_{Wi-Fi} = \frac{\sum R_{b,user}}{R_{b,peak}} \tag{2.2}$$

where:

- L_{LTE} is the load index in LTE;
- L_{Wi-Fi} is the load index in Wi-Fi;
- N_{RB} is the number of resource block, which depends on the bandwidth configuration;
- $N_{RB,user}$ is the number of resource blocks schedule for the user;
- $R_{b,user}$ is the user's data rate;
- $R_{b,peak}$ is the peak data rate for the AP.

After the load index has been defined, the basics of an LB process should be explained. Users start by choosing the preferable network according to the service requirements and network conditions. Then, when the QoS requirements cannot be satisfied, or when the load of the network reaches its threshold parameter, the process of handover is triggered. Handover can be triggered by a certain event, for instance a download of a large file, or by monitoring periodically the available networks, to check if there is a more suitable one.

Handover can be either horizontal (HHO), where an MT changes to other BS in the same system, or vertical (VHO), where an MT changes to other system. In this thesis, only VHO is addressed for the reasons that have already been mentioned previously.

In the process of handover, both networks, LTE and Wi-Fi, are evaluated. Parameters as dwell time, to avoid the ping pong effect, network load, internet connectivity, characteristics of link/path, latency, bandwidth, round trip time, among others, should be evaluated in order to select the best network for the user and for the operator. Signal strength should also be evaluated, although users may have a poor experience from a network with excellent signal strength, since it may be blocked by a firewall or even suffer from backhaul congestion.

There are several different approaches to VHOs in terms of the degree of collaboration between different entities that are the MTs, the cellular network and the WLAN network. Interaction among these entities can be done in many different ways, and thus leading to different VHO processes. Typically, there are three different approaches in VHO decision, [MPLK05]:

- *Tight coupling:* HO decision can be made either by the MT or by the network. Easy to deploy in WLAN networks owned by cellular operators.
- *Loose coupling:* HO is made by the MT but network provide some parameters like network load or coverage area.
- *No coupling:* MT measures and evaluates all the necessary parameters for HO and then makes the HO decision without any help of the network.

For the tight and loose coupling scalability and operator's management policy can be a drawback and for the no coupling approach resource management and network load balancing are problems.

Even if Wi-Fi can provide a better QoS, the solution to offload traffic should keep some data flows on preferred networks (e.g. Voice should be kept on LTE). Other types of traffic, as best effort (BE) and with low QoS, should be offloaded to Wi-Fi.

Although applying certain policies for specific types of traffic can be challenging, it is necessary in order to maximise available resources and user experience. In order to achieve this, 3GPP introduced the Access Network Discovery and Selection Function (ANDSF) framework which provides access network information and mobility policies. This information provided by the ANDSF framework is used to select the preferable network for the services being used by the users. In the early release of the ANDSF framework, all traffic was offloaded to LTE or to Wi-Fi. In order to solve this, Multi Access PDN Connectivity (MAPCON), IP Flow Mobility (IPOM) and non-seamless Wi-Fi offload were introduced by ANDSF framework's Inter System Routing Policies (ISRP), allowing the operator to indicate preferred radio access technologies as a function of the type of traffic.

In order to select the appropriate traffic to offload, identifying different types of Internet traffic is essential. Port 80 (HTTP) carries more than 50% of total Internet traffic including web browsing, video streaming, etc. Although those types of traffic share the same port number, there are different QoS requirements for all, consequently, identifying traffic independent of destination port is needed to provide users with the best QoS possible. IP Flow throughput, file size, application name or identifier, role identifier and Fully Qualified Domain Name (FQDN) are some of the ways of identifying the traffic type.

When offloading traffic from LTE to Wi-Fi or vice-versa, a smooth handover needs to be assured and that is why, for IP Flow Mobility, 3GPP specified the use of Dual Stack Mobile IPv6(DSMIPv6). DSMIPv6 allows users to move within the area covered by LTE and Wi-Fi networks while maintaining reachability and on-going sessions by preserving the IP address. The use of DSMIPv6 will give the user a better experience when handover is triggered.

Coverage and capacity are very important for LB and an estimation of both parameters can be obtained. For this estimation, the number of resource blocks each user has available is equal, and it depends on the bandwidth configuration, [Pire12].

$$N_{RB/U} = \left\lfloor \frac{N_{RB}}{N_U} \right\rfloor \tag{2.3}$$

where:

- N_U is the total number of users in the system.

The physical layer bit rate for each user, $R_{b/U}$, can be estimated by:

$$R_{b/U} = \frac{12\ N_{RB/U}\ N_{symbols/subframe}\ \log_2(M)\ \ N_{streams}}{\tau_{TTI}} \tag{2.4}$$

where:

- $N_{symbols/subframe}$ is the number of OFDM symbols per sub frame (already specified in Section 2.1.2);
- M is the modulation's order;
- $N_{streams}$ depends whether MIMO is used, if MIMO it is not used, this parameter is equal to zero (0), if it is used this parameter is equal to the order of the MIMO configuration;
- τ_{TTI} is the time transmission interval (1 ms).

Thus, with (2.3) and (2.4) an estimation of the capacity of the system can be obtained:

$$N_U = \left\lfloor \frac{12\ N_{RB}\ \ N_{symbols/subframe}\ \ \log_2(M)\ \ N_{streams}}{R_{b/U}\ \tau_{TTI}} \right\rfloor \tag{2.5}$$

The coverage area radius of a cell is given by:

$$r_{cell[\text{km}]} = 10^{\frac{P_{TX_{sub-carrier}}[\text{dBm}] + G_t[\text{dBi}] - P_{r_{sub-carrier}}[\text{dBm}] + G_r[\text{dBi}] - L_p[\text{dB}] - M_p[\text{dB}]}{10\ \alpha_{pd}}} \tag{2.6}$$

where:

- $P_{TX_{sub-carrier}}$ is the transmitted power;
- G_t is the gain of the transmitting antenna;
- $P_{r_{sub-carrier}}$ is the minimum receiving power required by UE;
- G_r is the gain of the receiving antenna;
- L_p is the reference path loss;
- M_p refers to the margins;
- α_{pd} is the average power decay.

Note that for LTE, an approximation of the limit of users per sector of a BS is the maximum number of available RBs, i.e.100 when 20 MHz bandwidth is considered, since RBs is the granularity used in the distribution of radio resources. However, this number does not contemplate RBs dedicated to signalling and control, so the maximum number of users is less than 100.

Regarding Wi-Fi, this number is limited by the AP as well as by users' throughput and by the IP addresses available for use. Although there is a theoretical value, that value is very difficult to reach, and in the case that it is reached, the users' experience would be poor.

To obtain the cell range of both systems, the expressions above specified to get an estimation of this parameter can be used. Mostly, LB between LTE and Wi-Fi occurs in urban areas where LTE cells are mainly micro-cell, leading the coverage range to vary between 50 and 500 m. As for Wi-Fi APs, a well-defined area does not exist since little variations can result in dramatic alterations in the signal strength. Note that the coverage area is associated with the AP, with the standard used, throughput provided as well as some other factors. However, this value is usually between 20 and 100 m.

2.5 Services and Applications

There are a wide variety of different applications at user's disposal and all of these applications have different requirements.

The most popular service is voice. This service is very predictable since its demanding performance from the network is very well known. Additionally, there are data services, which, contrary to the voice one, have a huge group of services, also popular, as Short Message Service (SMS), email, web browsing, among others, and its traffic growth makes them much more challenging. In order to face these challenges, and especially to provide QoS guarantees, data services must be classified so applications can be differentiated. Regarding this classification, 3GPP proposed four distinct service classes: Conversational, Streaming, Interactive and Background. Those service classes and their main characteristics are presented in Table 2.5.

Table 2.5 Service classes' main characteristics (extracted from [3GPP12a]).

Service class	Conversational	Streaming	Interactive	Background
Real time	Yes	Yes	No	No
Symmetric	Yes	No	No	No
Guaranteed throughput	Yes	Yes	No	No
Delay	Minimum and fixed	Minimum and variable	Moderate and variable	Large and variable
Buffer	No	Yes	Yes	Yes
Bursty	No	No	Yes	Yes
Example	Voice	Video streaming	Web browsing	SMS, e-mail

Although this classification is directed to cellular networks and the characteristics of WLAN are somehow different, this classification can be used to analyse services differentiation. Table 2.6 presents the services classes of WLAN and their corresponding services classes of 3GPP.

Table 2.6 QoS mapping rules between cellular network and WLAN (extracted from [CWSML11]).

WLAN	Cellular network
AC_VO	Conversational Class
AC_VI	Streaming Class
AC_BE	Interactive Class
AC_BK	Background Class

This classification had to be re-specified by 3GPP in order to keep up with the performance requests.

LTE's specified QoS parameters are:

- QoS Class Identifier (QCI) is an index that identifies a set of values for priority, delay and loss ratio. Instead of signalling these three parameters separately, QCI is signalled. Operators may create additional classes within their network;
- Allocation and Retention Priority (ARP) indicates the priority of the bearer compared to others when conducting decisions of admission control;
- Maximum Bit Rate (MBR) identifies the maximum bit rate for the bearer;
- Guaranteed Bit Rate (GBR) identifies the guaranteed bit rate to the bearer;
- Aggregated Maximum Bit Rate (AMBR) indicates the total maximum bit rate for non-GBR

bearers that a UE can have.

In Table 2.7, Resource type indicates whether classes have GBR or not, Priority defines the priority for the packet scheduling, Delay Budget helps the packet scheduler to maintain the delay requirements for the bearer and the Loss Rate helps to use appropriate Radio Link Channel (RLC) settings. Most of GBR services have a higher priority than non-GBR. Thus, the available resources are first distributed for users requiring GBR services and the remaining resources distributed within non-GBR ones.

For WLAN, another type of traffic differentiation is proposed in the IEEE 802.1D standard which provide latency and throughput guarantees. IEEE 802.1D defines seven traffic types:

- Network Control (NC) – consists of traffic needed to maintain and support the network, it is both time-critical and safety-critical;
- Voice (VO) – It is time critical, characterised by a low maximum delay;
- Video (VI) – It is also time critical but the maximum delay is not as severe as with the VO;
- Controlled Load (CL) - Loss sensitive and non-time-critical. Used by important applications, such as business applications, that are subjected to admission control or controlled throughput;
- Excellent Effort (EE) – It is best effort traffic with a higher priority. Non-time-critical but loss sensitive;
- Best Effort Traffic (BE) – It is non-time critical and loss insensitive i.e., the normal LAN traffic;
- Background (BK) – Non-time-critical and loss insensitive. Allowed on the network but should not interfere with the other types of traffic.

Table 2.7 QoS parameters for QCI (extracted from [3GPP13b] and [HoTo09]).

QCI	Resource Type	Priority	Packet Delay Budget [ms]	Packet Error Loss Ratio	Example Service
1	GBR	2	100	10^{-2}	Conversational Voice
2		4	150	10^{-3}	Conversational Video
3		3	50	10^{-3}	Real Time Gaming
4		5	300	10^{-6}	Non-conversational Video (Buffered Streaming)
5	Non-GBR	1	100	10^{-6}	IMS Signalling
6		6	300	10^{-6}	Video (Buffered Streaming) TCP-based
7		7	100	10^{-3}	Voice Video (Live Streaming) Interactive Gaming
8		8	300	10^{-6}	Video (Buffered Streaming) TCP-based
9		9			

For a comparison purpose, Table 2.8 presents the mapping between UP, traffic types and AC

Table 2.8 Mapping between UP, traffic types and AC (adapted from [IEEE12a]).

Priority	User Priority (UP)	802.1D designation	AC	Example Service
Lower	1	BK	AC_BK	Tradition applications. E-mail, web browsing, etc.
	2	-	AC_BK	
	0	BE	AC_BE	
	3	EE	AC_BE	Video streaming
	4	CL	AC_VI	Interactive Services
	5	VI	AC_VI	Video conference
	6	VO	AC_VO	VoIP
Higher	7	NC	AC_VO	Signalling

2.6 State of the Art

The concept of LB between Wi-Fi and cellular networks is not new and some work has been done in this area. Although, the focus of this thesis is LTE and Wi-Fi systems, other information about interworking of cellular networks and Wi-Fi can be beneficial.

A policy framework for resource management where load balancing policies are designed to efficiently utilise the pooled resources of the network is discussed in [SoWC07]. These LB policies consist of a two-phase control strategy in which call assignment is used to provide a statistical QoS guarantee during the admission phase, and dynamic VHO during the traffic service phase is used to minimise the performance variations. Through the numerical results presented, authors demonstrate that this LB solution achieves a significant performance over two reference schemes.

A new approach of LB is presented through its algorithm and its dynamic optimisation, [NaSA07]. A LB algorithm is proposed with auto-tuned load threshold and then its simulations and results are presented. Simulations results obtained for this algorithm show that it controls the amount of VHOs between both systems, prevents unnecessary HO and increases overall call success rates for both real time and non-real time traffic.

[CWSM11] presents a LB scheme composed of QoS-based and load-related admission control and dynamic VHO. Authors claim that this scheme provides better QoS and jointly optimises resource utilisation, providing better performance in packet loss rate and load balance compared with the traditional schemes.

Although, the references above are related to UMTS and not with LTE, they provide an important overview and some approaches for the LB between Wi-Fi and cellular networks.

[Naik10] provides an overview of the LTE WLAN interworking architecture and the reference implementation, for operator owned and aggregator provided Wi-Fi hotspots. Several challenges are presented as well as several recommendations for them.

In [LLZL10], a novel distributed mobility LB algorithm is introduced through adjusting Radio Resource Management (RRM) parameters based on the source cell load and its neighbouring cell conditions. By adjusting these parameters, the load can be directed to the target neighbouring cell without causing much degradation of user's satisfaction. Three different load levels are defined so LB can be as smooth as possible, one for the energy saving procedure, other for the LB preparation and the last one for the LB trigger.

In [AzRS10], a fuzzy based VHO algorithm is proposed to provide an efficient VHO between LTE and WLAN. In this work, the considered fundamental parameters are bandwidth, SNR, battery life and network load, and a fuzzy logic system is exploited for decision making to achieve an efficient VHO.

Qualcomm offers a connectivity engine through a 3G/LTE Wi-Fi offload framework in [Qual11a]. Qualcomm's connectivity engine has three components, a mechanism to provide operator's policy for unplanned networks to the device in a dynamic fashion, algorithms in the device to detect characteristics of unplanned Wi-Fi networks and determine the best possible use of available networks, and a mechanism to allow seamless handovers between 3G/LTE and Wi-Fi. This will lead to efficient capacity management, improvements in user experience, longer battery life and support for simultaneous 3G/LTE and Wi-Fi access.

[Qual11b] presents a comparison of LTE advanced HetNets and Wi-Fi. Its objective is to compare offload solutions, LTE pico-cells and Wi-Fi. Qualcomm presents various results for throughput in different scenarios. In this comparison, LTE pico-cells offer a significantly better user experience and system capacity improvement than Wi-Fi APs, and it also has better support for mobility/handoff, QoS, security and SON. However, from the conducted analysis, Wi-Fi provides a gain in user throughput when a significant portion of UEs can be offloaded from macro cells to Wi-Fi APs (hotspot scenarios), which is the focus of the thesis.

In [ChSo11], a data rate based VHO triggering mechanism based on IEEE 802.21 Media Independent Handover (MIH) standard is presented, in order to maximise capacity of both WLAN and LTE. In this mechanism, whenever a MT discovers a WLAN, WLAN's achievable data rate is obtained through MIH and, based on that information, MT decides whether to execute VHO or not. From the obtained results, capacity of overall networks can be maximised with the use of this triggering mechanism.

[Hago12] addresses solutions for Wi-Fi offloading in LTE networks when performance needs exceeds the capability of the LTE access. Novel offloading algorithms are implemented, evaluated and finally compared. Authors propose an optimised SNR-threshold based handover solution that can be used for Wi-Fi offloading.

Authors of [SNPG12] present a LB algorithm for EPS using various radio interfaces. The main goal of this algorithm is to minimise the number of unsatisfied users in the network and, thus, it is only active if unsatisfied users are presented. This algorithm considers moving users with high PRB usage in order to minimise the number of handovers needed.

In [MeRo12], a heuristic algorithm is used in order to achieve the required LB between macro-cells and low power node cells. This algorithm makes use of a range optimisation framework, because cell range extension can be used to achieve load balance, and it evaluates users' satisfaction measured by Jain's fairness index.

In [FuKw13], authors present a novel load balancing scheme for the OFDMA LTE cellular and Wi-Fi coexisted network. A LB algorithm to balance the network load measured by the ratio of the average Wi-Fi load to the LTE cell load as well as the Jain's fairness index is presented.

[ChKS13] uses MIH to provide MT with the necessary information for HO and it proposes a MIH VHO. In the proposed model, systems and vertical handover decision agent (VHDA) exchange event messages with each other, and when VHDA decides the VHO for each flow, it sends the result to the MT. The simulation showed that the proposed algorithm can control the flow composition according to Wi-Fi conditions to save costs and resources of 3G.

[Cast13] is focused in energy efficient VHOs. By implementing a VHO algorithm in mobile devices, a large energy saving can be achieved. In the presented approach, the automatic changes between systems are service-driven: every time a service is requested, the UE leverages the characteristics of such service, the environmental conditions and its own characteristics can then decide. The author states that Wi-Fi is more energy efficient than LTE and that there is no problem from an energy point of view of WLAN offload.

Besides all of the previous stated work, 3GPP has several specifications regarding LTE and Wi-Fi networks and its interworking.

Chapter 3

Model Development

A description of the models used in this thesis is provided in this chapter. Its formulation and implementation is described. Finally, one presents the assessment of the model implemented in the simulator.

3.1 Theoretical Model

This section provides a description of the theoretical model, algorithm and its metrics, which were implemented in a simulator.

3.1.1 Load Index and Load Threshold

The objective of LB is balance load in order to offload some traffic from an overloaded cell to a cell that does not has load over a predefined threshold. Therefore, the Load Index is an important element in LB, because it indicates the quantity of resources of a network that are available or being used. It is this value that gives the load of a cell, this value being monitored to ensure the load balance of the network.

$$L_{LTE} = \frac{\sum N_{RB,user}}{N_{RB}} \tag{3.1}$$

$$L_{Wi-Fi} = \frac{\sum R_{b,user}}{R_{b,peak}} \tag{3.2}$$

For LTE, the Load Index is defined by the sum of all schedule RBs to users divided by the total number of RB. As for Wi-Fi, since there is no better way to define it, Load Index is defined by the sum of all users' throughput divided by the maximum throughput that is available for the AP. In both cases, the Load Index gives the percentage of used resources, this value being compared with the defined threshold value in order to see if it is necessary to use the LB algorithm.

Load Threshold is one of the key parameters for the LB algorithm, since this parameter defines when LB is performed. There is no defined expression for this parameter, therefore it is through simulations that one gets the optimum value for this parameter. In a real scenario, this parameter should be dynamically adjusted according to the expected load in the network and to operator's needs.

3.1.2 Service priority

A service priority list for both LTE and Wi-Fi needs to be defined, so that LB guarantees the best QoS possible for all users. The main criteria to establish the service priority was that LTE should be chosen for the high priority services, since QoS guarantees it provides are more reliable than the Wi-Fi ones.

Table 3.1 presents the services that were implemented, as well as the priority list used in the LB algorithm. Although the services are grouped into four different classes, within the same class different traffic models are used, which explained ahead as well as which services are going to use which model.

Since LTE does not support voice yet, i.e., Voice over LTE (VoLTE), only VoIP is considered and because of that Wi-Fi was chosen to be the preferable network for this service. Video call and Intelligent Transportation Systems (ITS) are other conversational services that used. Since QoS guarantees can be stricter on mobile networks, comparing to Wi-Fi, LTE is the preferable network for

Video Call. ITS is considered a conversational service since it represents the communication between machines, i.e., the exchange of data between them. Because of its characteristics, this service is prioritised for Wi-Fi.

Table 3.1 Service priority list.

Service Class	Service	LTE	Wi-Fi
Conversational	Voice		✓
	Video call	✓	
	ITS (M2M)		✓
Streaming	Video Streaming	✓	
	Music	✓	
	Surveillance (M2M)	✓	
	File Sharing		✓
Interactive	Social Networking	✓	
	Web Browsing	✓	
Background	e-Health(M2M)		✓
	Smart Meters(M2M)		✓
	E-Mail		✓

Video streaming, music and surveillance are services that are strict in the terms of QoS, so LTE is the chosen network. File sharing is not as priority as the other streaming services so its preferred network is Wi-Fi. As for the interactive services, such as Web browsing and social networking, the preferable network is LTE.

Finally, background services use Wi-Fi as their network, because of their characteristics regarding QoS. While no introduction is needed for e-mail, Smart meter and e-health need to be introduced in this category. It is considered that smart meters are electronics devices that record data and later communicate that data to the utility for monitoring and/or billing purposes. E-Health, which relates medicine/healthcare and information technology, can encompass a range of services including electronic health records, ePrescribing, Telemedicine, and consumer health informatics, among others.

3.1.3 Models to be used in the simulator

The global model combines the Random Walk Mobility Model, which was developed to simulate MTs heretic behaviour with a speed Triangular Distribution Density Model in order to be capable of generating different speeds for each MT, which is important for link load variation and CRRM handover generation due to propagation changes. Both Random Walk Mobility Model and Triangular Distribution Mobility Model are explained in more detail in Mobility Models A.

Concerning the propagation models, one uses the COST 231 Walfish-Ikegami propagation model to estimate the signal level in LTE. If the RAN under simulation is Wi-Fi, then the simulator switches to the Double Breakpoint model, which is dedicated to estimate the signal level of WLAN in the 2 and 5 GHz bands. Similarly to mobility models, propagation models are also explained in detail in B.

The volume and scheduling of each Wi-Fi packet are defined by traffic source models, described in C. For a given user, with a packet to transmit, the number of time-slots to transmit the packet is calculated according to the instantaneous bandwidth, assigned to the user at that moment. Therefore, the number of time slots needed to transmit a packet is essentially dictated by the network. Regarding LTE, it is explained in more detail in the next section.

Figure 3.1 presents the zoom over the general algorithm functionalities and also the scope of different entities, like RRM and CRRM in the simulator.

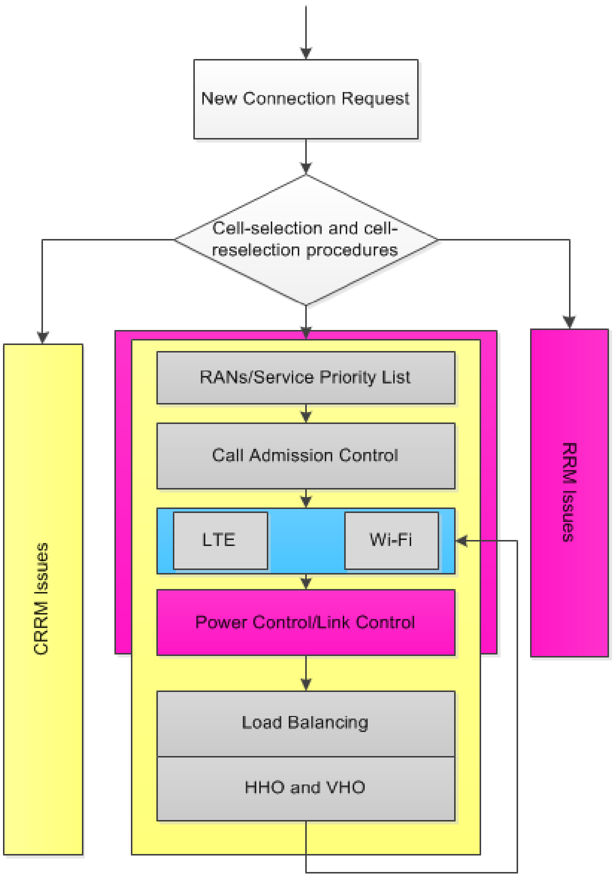


Figure 3.1 Simulator general functionalities.

The RRM entity, which is represented in pink, besides the RAT basic functions, covers the basic RAT algorithms, such as call admission control, power control, and link control. The CRRM entity is represented in yellow and besides its own functionalities, the CRRM entity may also cover or manage the RRM ones, since CRRM has the capability of changing the guide parameters of the RRM algorithms and functionalities, which excludes fast (time dependent) RRM functions, like fast power control. In any case, the CRRM entity covers other algorithms, like congestion control, HHO and VHO and load balancing.

In this thesis, load balance is divided in two algorithms: CAC and LB algorithms.

The CAC algorithm, Figure 3.2, is a very important mechanism to allocate the initial BS to a given MT/service. The CAC is triggered when a given MT has a potential starting call/session. When this event is detected, the MT launches a request that is processed by the CAC algorithm. The algorithm's predefine priority list is used to obtain the most priority RAT to a given service. If OFDM is the RAT (Wi-Fi) selected, then the MT verifies the communication viability (UL and DL propagation), after this process the BS or AP bandwidth availability is verified, if so, the MT is attached, otherwise, it is rejected by delay. If the RAT selected is OFDMA (LTE), a similar process is executed, which consists of the verification of the BS load limit and the resource blocks availability, and if both are enabled then the attachment process is executed. If any of the previous radio resources are unavailable, then the same treatment is performed as in the OFDM case.

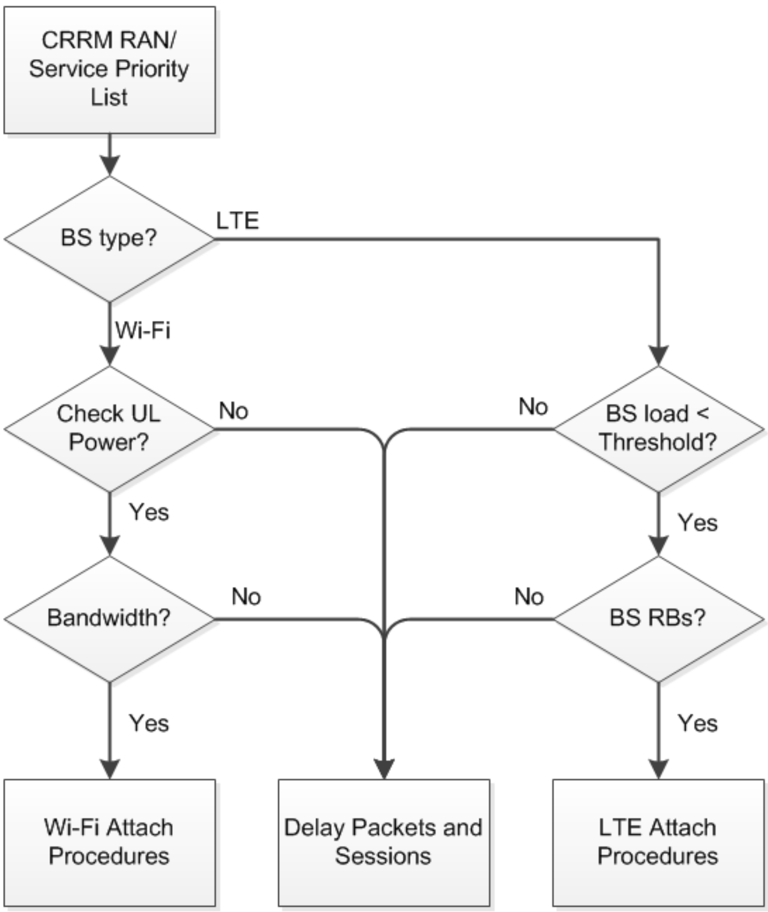


Figure 3.2 CAC algorithm.

The Load Balance (LB) algorithm (presented in Figure 3.3) is a process that may be triggered if the load index of a given base station is above a given threshold. If this condition is verified, then the LB algorithm starts. This process pushes MTs (forcing handovers) away from highly loaded BSs until they present a predefined load level. The LB algorithm may cause a ping pong effect on MTs, since they can be moved endless from one BS to another; therefore, a counter is implemented to limit and avoid this effect. If the load index is above a given threshold, the overloaded cell and its neighbouring cells are locked so that they do not accept more users, and then the algorithm determines which users are capable of perform a HO and the amount needed in order to get the load below the threshold. After the list of users able to perform HO is done, one MT is selected and it performs the HO algorithm, which is presented below. While the load is above the threshold, being possible to perform HO, the algorithm select one MT so that it can perform the HO. When the load is below the threshold or there is no possibility to perform more HO, the algorithm unlocks the overloaded cell and its neighbouring cells.

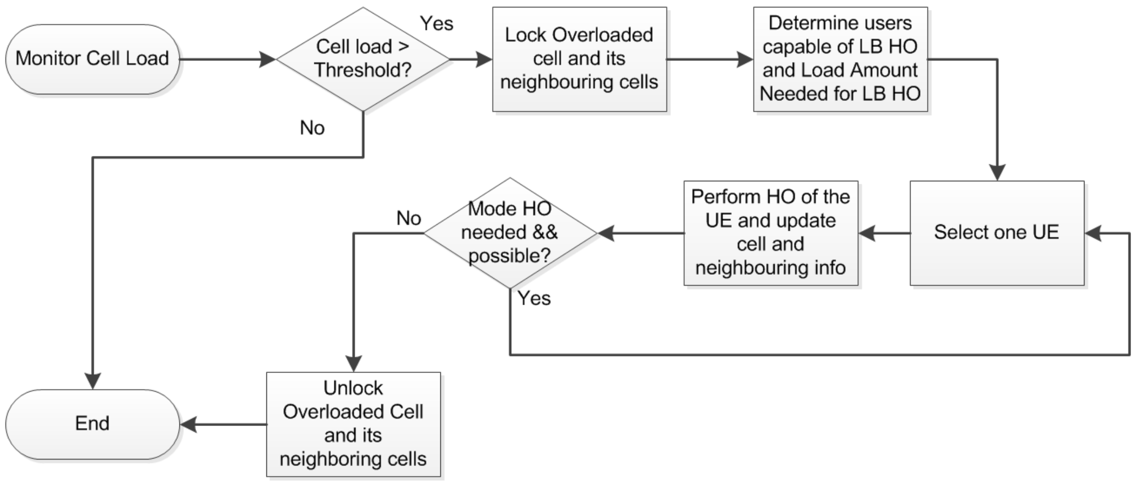


Figure 3.3 LB algorithm.

As explained before, HO plays an important role in LB. The HHO and VHO process are managed by the same algorithm, presented in Figure 3.4. This algorithm starts by generating a candidate list based on the MT/service and on the communication capabilities between BSs and the current MT. If the BS in the list is from other RAT type, then a VHO is performed, but if the RAT type is the same, then a HHO is triggered. After this decision, the old BS releases the radio resources regarding the MT/service, with the selected/new BS being requested. If in this process the BS (for any reason) rejects the request, then the next BS from the sorted list is selected, repeating the allocation process, until the list is empty or after a successful handover, otherwise, the algorithm returns a drop indication, and the MT/service drops the current service.

In this process of HO, one has implemented a delay concerning this process. It is called the cell reselection delay, or handover delay, $D_{handover}$, and is the total time between the end of transmission of the handover command on the source cell, and the start of UE transmissions to the target cell. This cell reselection delay can be calculated by:

$$D_{handover[\mathrm{ms}]} = T_{search[\mathrm{ms}]} + T_{IU[\mathrm{ms}]} + T_{margin[\mathrm{ms}]} + T_{processing,RRC[\mathrm{ms}]} \qquad (3.3)$$

where:

- T_{search} is the time required to identify the cell if it is unknown. The cell is unknown only in the case that the handover is not based on UE measurement, and otherwise is zero (0);
- T_{IU} represents the uncertainty of acquiring the first available random access occasion, which can be up to 30 ms;
- T_{margin} is the implementation margin, which is 20 ms;
- $T_{processing,RRC}$ is the time in which the UE must be able to precess the received message and produce a response and in case of connection reconfiguration this value is set to 15 ms.

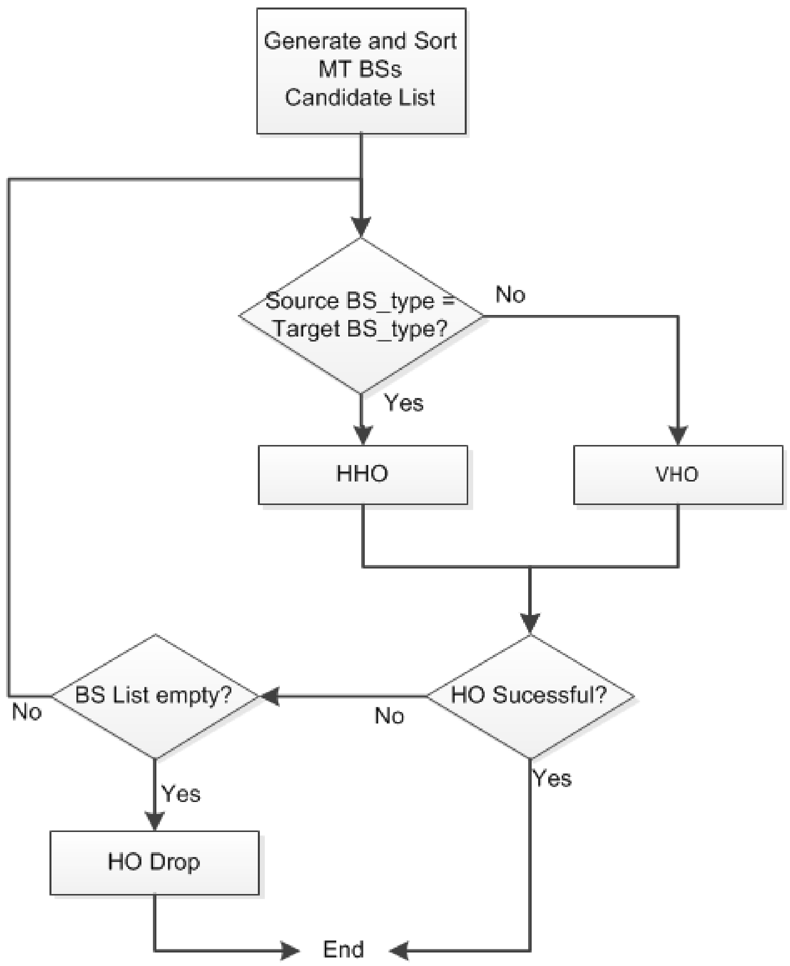


Figure 3.4 HO algorithm

In case of handover, this delay must not be over the maximum allowable delay. This maximum allowable delay represents the time for which the UE is no longer receiving and transmitting to the source cell, and has not yet started transmission to the target cell, which means that the UE is no longer attached to any BS. It takes the value of 65 ms plus the time required to identify the target cell.

3.1.4 Implementation of LTE, Wi-Fi and M2M Services

Regarding the software perspective, an upgrade of the simulator was done, in which an LTE module was added, the Wi-Fi module suffered an update in order to support new technologies and the new services were also added (ITS, e-Health, Smart meters and Surveillance).

One of the main challenges of the implementation of the LTE module was the base simulation tool temporal resolution. This resolution is of 10 ms, while LTE works with RB with slots of 0.5ms. The solution to this problem is quite complex and it required special attention and some simplifications. Hence, the 10ms resolution from the original simulator was preserved, due to the complexity of changing this factor, and the allocation of RBs is made taking into account a temporal window with the same size as the simulators' base resolution.

Any network is limited in capacity and somehow in the number of users handled simultaneously at a given instant. Therefore, it is important to regulate which users can access the network and under which conditions. In order to do so, the algorithms presented in the last section were implemented. Note that as the bitrates assigned to the users are variable in time, the traffic control algorithm is applied not only for admission but also for congestion, thus, controlling the load of the system at every instant.

When load reaches values near the threshold, QoS is affected, as a consequence of an excessive loss/delay of packets. In order to handle this problem, a maximum of 30 lost frames is assumed as the limit of quality degradation causing the MT to be dropped at the 31st lost frame.

Concerning allocation of RBs, LTE presents a strong challenge in order to allocate users in both frequency and time domains. For simplicity, this thesis only considers a one dimension problem achieved by replicating the time domain on the single frequency domain. Note that with this approach it is assumed that users experiments the same radio channel conditions for the entire radio frame and it does not reflect the frequency selective nature of the radio channel. This leads to a worse system performance compared with the ideal one. Other simplification done was the fact that the user uses the same modulation throughout the entire time that is using the service. Therefore, the results obtained have an error associated with these simplifications.

With these simplifications, an example of how RBs are scheduled should be given. For instance, for the 20 MHz bandwidth there are 100 RBs available each instant. In the simulator's time resolution, there is a total of 2000 RBs available. Of those RBs, up to a maximum of 10% are reserved for signalling purposes and the others 90% is used to schedule the users.

$$N_{RB} = \left\lceil \frac{R_{b[\text{bits/s}]}\ \tau_{TTI[\text{ms}]}}{2\ \times 12\ \ N_{symbols/subframe}\ \ \log_2(M)\ \ N_{streams}} \right\rceil \tag{3.4}$$

where:

- τ_{STR} represents the simulation temporal resolution;

Note that N_{RB} is the number or RBs needed per 10ms.

Taking as an example the video streaming service with an average bit rate of 5120 kbps and modulation of 16-QAM with coding rate of 1, the allocation of 8 RBs each 10 ms as initial allocation is necessary. This value may change throughout the simulation.

Traffic models, described in C, define both volume and scheduling of each packet. Given a user with packets to transmit, and after the initial allocation of RBs, one calculates how many RBs are necessary to transmit in the full duration of the 10ms, based on the traffic model.

The COST 231 – Walfish-Ikegami model is used for LTE and even though the model is validated for an upper limit of 2 GHz, it is acceptable to use this model that is subject to some lack of accuracy beyond this limit, since the focus of this thesis is Load Balance and not propagation issues.

Wi-Fi changes were mostly parameter changes in order to update the already existent version to the new 802.11ac. In order to do that, frequency, bandwidth, stream data rate, allowable MIMO streams, parameters were changed, and some code had to be written in order to support those changes.

M2M services were specified in the simulator. However, these services use the same traffic models that were already referred. The implementation of these services basically consisted of defining which parameters to use in those traffic models as well as QoS parameters for these services.

3.2 Simulator

3.2.1 General Description

The simulator is a system level, time based simulator, with a resolution of 10ms, which has been developed over the Microsoft Visual Studio 2005 platform. In its original version, developed by Serrador [Serr12], the tool supported UMTS and Wi-Fi only. For the current work, an upgrade to the original version was performed, in order to attach an LTE dedicated module and an update in the Wi-Fi module was done so that the simulator could support the recent versions of this technology.

Although the simulator was updated, its functions were not changed and they are divided into three main functions/blocks, as seen in Figure 3.5, identified by different colours; this figure, presents a general overview over the simulator, showing that this simulator is versatile and can be updated and/or changed to make various possible simulations. Figure 3.6 shows the simulator used in this thesis.

The green colour refers to the input data:

- Scenario inputs: represent the simulation area, services source models configuration data, services rates and duration, propagation models information, location of BSs, buildings and streets information, etc.;
- MT/Users input: number of users, service penetration, etc.;
- Multi-RRM Algorithms Policy Inputs: defines the parameters related to CF weights, maximum

QoS parameters, for each type of RAN. Note that, based on these parameters, different policies can be simulated;

- RAN#1 (BS 1) up to RAN#n (BS n): input parameters for different RANs, like pilot power level, MT maximum power, antennas patter, total power, frequency, etc..

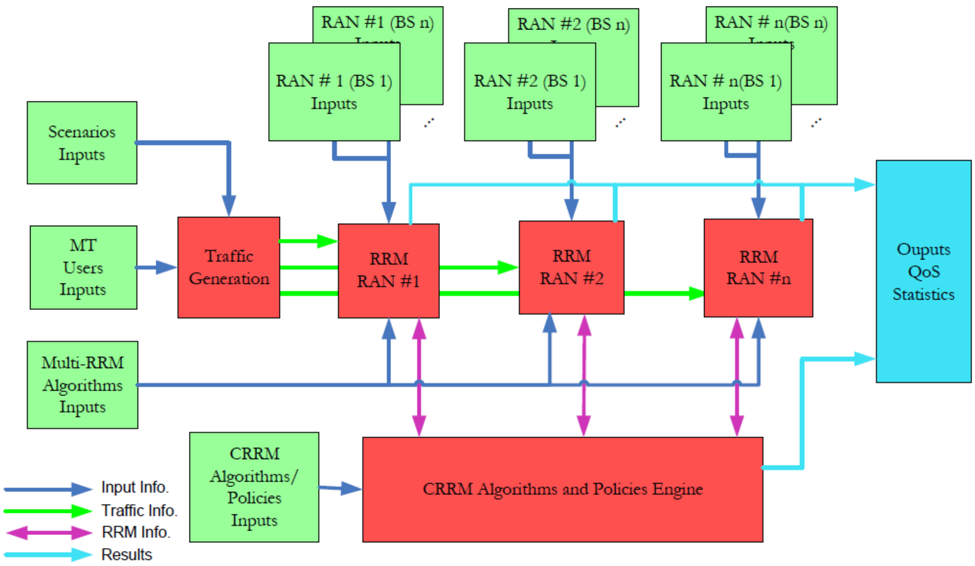


Figure 3.5 Simulator general block diagram.

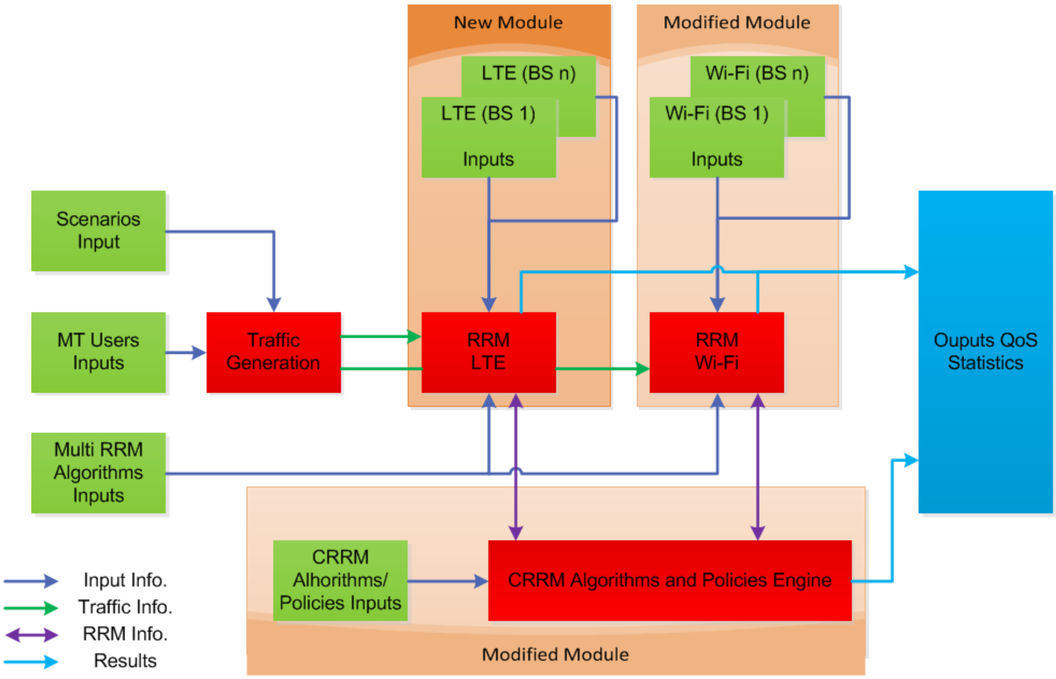


Figure 3.6 Simulator block diagram.

The red set of blocks is where most of the simulation computational load is performed.

- Traffic Generation: in this block where all traffic vectors of all MTs and services are built, usually with a time frame of one hour;
- RRM RAN#1 up to RRM#n: these blocks are very “intense”, since they perform the fundamental functionalities of a given RAN, by running/managing and monitoring the radio

links conditions and services attach (generated by the Traffic Generation block), thus, requesting a high computational effort;

- CRRM Algorithms and Policies Engines: this is also an “intense” block, since it is here where major decisions are taken; being common to all RANs, this block is many times requested, not just to perform all types of handover, but for initial BS selection, and to run the CF, which is related to all BSs and MTs active in the scenario.

Finally, the blue block is where the selected output parameters are displaced, most of them being QoS and system statistics.

3.2.2 Input and Output

The input parameters can be conceptually divided into two groups:

- Scenarios and MT users.
- Multi-RRM Algorithms.

The group “Scenarios and MT users” is composed of following input parameters:

- Number of BS and of MTs.
- Number of fast MTs and of slow MTs.
- Minimum speed for fast MTs and for slow MTs.
- Maximum speed for fast MTs and for slow MTs.
- Service mix and the corresponding individual source model parameters.

The Multi-RRM Algorithms group defines the parameters required to define the BSs signal propagation conditions, as well as some RRM initial parameters like the service priority list:

- BS type, location, height and total power, total available channels, orthogonal factor (CDMA environments), receiver noise figure, cable loss, frequency and pilot Tx power.
- Number of channels used for signalling.
- Receiver noise density.
- Thermal noise density.
- Building loss (penetration additional factor).
- Body loss.
- MT maximum Tx power and height.
- Building height.
- Street width.
- Urban type.
- Street orientation.
- Building separation distance.
- Services priority table (services priorities mapped into RANs).

The CRRM simulator output block produces a set of results that are stored in a file. The following output parameters have been defined and implemented, in order to reflect the network’s overall performance.

- The Average Delay, τ, is defined for Web browsing, Streaming, E-mail and FTP, as a measure of the delay affecting the transmission of packets:

$$\tau = \frac{N_{fd}}{N_{pt}} 10ms \text{ [ms]} \tag{3.5}$$

where:

  - N_{fd} is the number of delayed frames;
  - N_{pt} is the total number of packets transmit.
- The Number of Vertical Handover Attempts, N_{VHOa}, is defined as the number of attempts to perform a handover between BSs from different systems.
- The Number of Horizontal Handover Attempts, N_{HHOa}, is defined as the number of attempts to perform a handover between BSs from the same system.
- The Probability of VHO failure, P_{VHOf}, is defined as the percentage of failed vertical handovers:

$$P_{VHOf} = \frac{N_{VHOf}}{N_{VHOa}} 100 \text{ [\%]} \tag{3.6}$$

where:

  - N_{VHOf} is the number of failed VHO;
- The Probability of HHO failure, P_{HHOf}, is defined as the percentage of failed horizontal handovers:

$$P_{HHOf} = \frac{N_{HHOf}}{N_{HHOa}} 100 \text{ [\%]} \tag{3.7}$$

where:

  - N_{HHOf} is the number of failed HHO;
- The Drop rate, D_r, is defined as the percentage of dropped sessions.

$$D_r = \frac{N_{sd}}{N_{st}} 100 \text{ [\%]} \tag{3.8}$$

where:

  - N_{sd} is the total number of dropped sessions;
  - N_{st} is the total number of sessions.
- Average Bitrate per Service, $R_{b,service}$, is defined as the average value of the bitrate per service per second.
- Average Bitrate per RAN, $R_{b,RAN}$, is defined as the average value of the bitrate per RAN per second.
- Average Global Bitrate, $R_{b,global}$, is defined as the mean value of the Average bitrate per RAN:

$$R_{b,global} = \frac{R_{b,LTE} + R_{b,WiFi}}{2} \tag{3.9}$$

- The Average Number of Users per Service per Second, NU_{srv}, is defined as the average value of the number of users per service per second.

- Three types of Load are defined: per BS, per system and global.
- The number of each services active per BS.

3.3 Assessment

As already was stated before, the simulator used in this thesis consists of an upgrade of a previous version, developed by Serrador [Serr12] and later upgraded by Venes [Vene09]. In this way, apart from the LTE RAN block, the models in which the resulting simulator is based have been properly validated in the respective sources. The resulting version of the simulator was also validated. The overall performance of the simulator and its output were also validated by various test simulations and compared with theoretical values.

The current simulator has an initial set up period, as expected, since it is a time-based and dynamic system. This instability period occurs at the beginning of the simulation period and after this, a convergence to stability is observed. It is, as usual in these cases, necessary to assess a starting point from which the simulation parameters are expected to be stable, after the initial instability. Note that results obtained before the starting point should be discarded.

In order to assess the simulation period, simulations of 1h10 were performed. The scenario for this simulation included the two systems covered in this thesis, LTE and Wi-Fi, and considered a user density of 10 000 users/km^2. Users are active for a given service, which are characterised by some different factors, which values depends on the system and on the channel conditions.

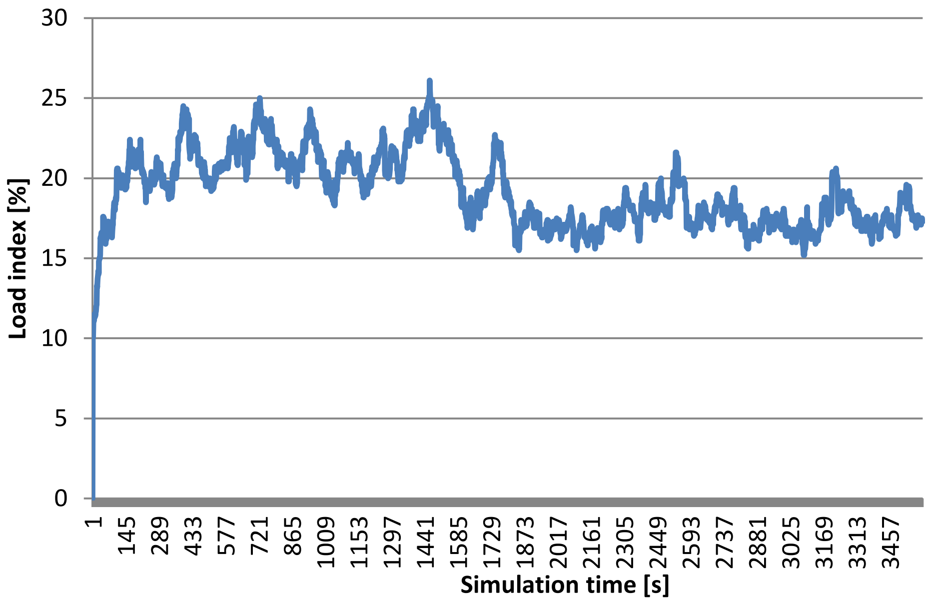


Figure 3.7 Load index for a LTE BS.

A detailed observation to the results obtained through these simulations shows that stability is achieved around the 10th minute of the simulation. This fact is also verified in the previous versions of this simulator, so it can be concluded that this version also has that same instability time. As a consequence, the instability time of the simulations are discarded.

Maximisation of the number of simulations results and the minimisation of time/CPU resources spent in simulation work configures a non-trivial optimisation problem that needs an answer; therefore it is necessary to find the number of simulations to run in order to obtain reliable values.

In order to get that number, a performance function is defined. It is assumed that this function has the inputs (Number of users, geographic distribution, service penetration, priority table, etc.) and outputs (load index, bit rates, delay, drop rate, block rate. HO attempts, etc.) already described in the previous section. The characterisation of this function is given by the characterisation of its individual functions achieved by defining a given number of scenarios, where all the parameters are fixed values, which are defined for REF, except for one that will float inside a given range of values.

The convergence of the performance function is measured by, Δ,

$$\Delta[\%] = \frac{|O^{cum\_s} - O^{cum\_S}|}{O^{cum\_S}} 100 \quad (3.10)$$

where:

- s is the simulation run index;
- S is the total number of simulations done;
- $O^{cum_s}$is the partial output parameter cumulative mean at simulation run s;
- $O^{cum_S}$is the total output parameter cumulative mean.

Figure 3.8 shows the convergence of the output parameters for the REF scenario. It can be observed that 6 simulations are needed to obtain a measure of convergence below 10%.

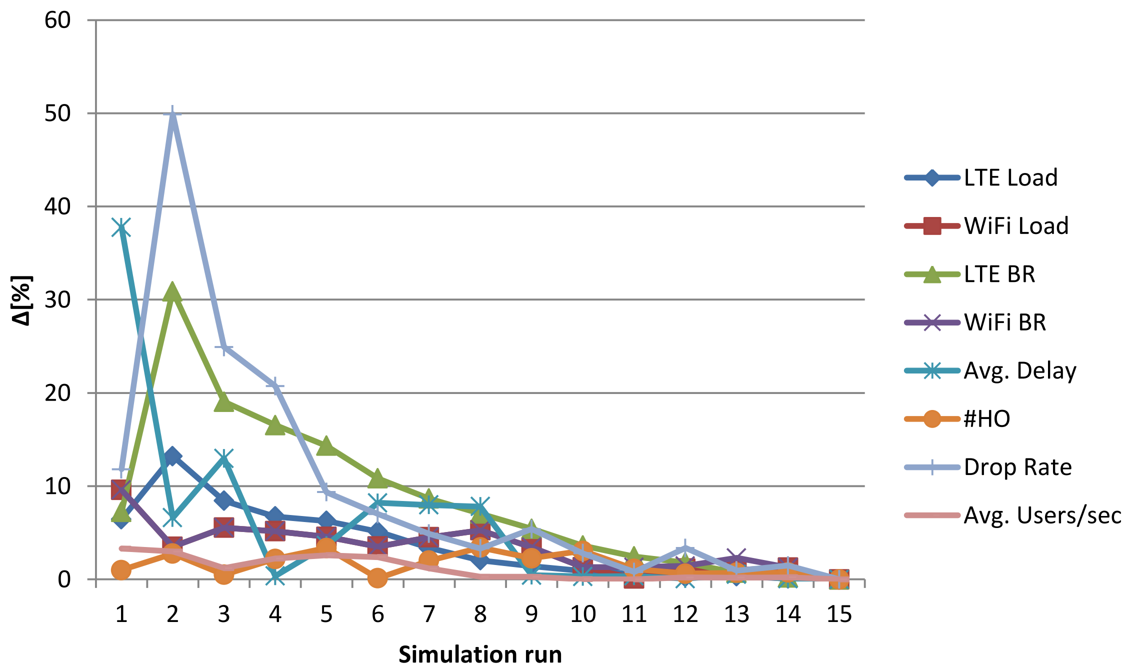


Figure 3.8 Convergence of the performance function

Considering that there are 7 variable outputs, and that the variation of its parameters can be characterised by 3 or 4 scenarios per variable, a total number of approximately 42 to 56 scenarios have to be run and for each scenario 5 simulations is done. Up to 280 simulations were performed. Assuming an average simulation time of 2h per scenario, around 10 days of simulations were needed.

Chapter 4

Result Analysis

This chapter starts by describing the scenarios used in the simulator. Then results, concerning the parameters under analysis, obtained through the simulation of those scenarios, are presented, analysed and discussed.

4.1 Scenario description

This section provides an overview on the reference scenario, upon which some variation of parameters were performed.

The Reference Scenario (REF) is an urban environment, in the city of Lisbon, located between Saldanha and Campo Pequeno, composed of 1 LTE BS and 3 Wi-Fi Aps. The main characteristics of the location are summarised in Table 4.1 and the main characteristics of the BSs of the different RANs in Table 4.2

Table 4.1 Terrain Parameters

Parameter	Value
User density [km^{-2}]	10 000
Street width [m]	20
Building height [m]	25
Building separation [m]	40
MS height [m]	1.5

Table 4.2 Simulation parameters.

Parameter	LTE	Wi-Fi
BS height [m]	22	6
Maximum Tx Power [W]	46	0.1
Frequency Band [GHz]	2600	5000
Channel Bandwidth [MHz]	20	80

The sites of the BS and APs are represented in yellow and red respectively in the Figure 4.1. Users take a uniform distribution, as it can be seen in this figure, where the blue dots are the users. For the sake of simplicity, users are only allowed to use a service throughout the whole simulation. Figure 4.2 presents the service penetration and Table 4.3 presents the main characteristics of those services. In this scenario, both LTE and Wi-Fi RAN are considered and available to the user; note however that for every scenario, there is a version with only LTE BSs, in order to compare performance parameters when both LTE and Wi-Fi are considered.

Regarding the LTE RAN, the OFDMA RAT is considered, using the 20 MHz bandwidth. The cells of this RAT have a radius of 400 m and each cell has 100 Radio Resource Units (RRUs) available.

Figure 4.1 Scenario Location and BS distribution for REF.

Wi-Fi (OFDM) is provided by means of IEEE802.11ac standard APs. These APs are configured to work at 80 MHz channel bandwidth and each one covers a cell with radius of 80 m, using beamforming and MU-MIMO in order to support up to eight spatial streaming. These APs only have five available channels due to European Union rules and regulations, being considered that half is for UL and the other half for DL. It is also assumed that only 15% of Wi-Fi AP's capacity can be used for mobile subscriber and the rest belongs to private fixed users.

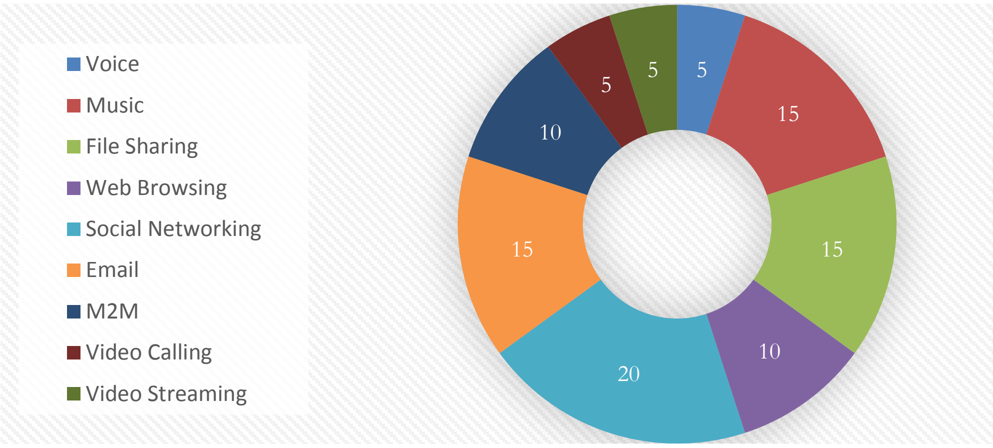


Figure 4.2 Various service penetration.

The service set in this scenario is presented in Figure 4.2 which shows the service penetration which is, is the percentage of active subscribers using that service.

Table 4.3 Service characteristics

Service		Service Class	Max Data Rate [kbps]			Duration	Size
			Min.	Average	Max.		
Voice		Conversational Real Time	5.3	12.2	64	60 [s]	–
Music		Streaming Real Time	16	64	160	90 [s]	–
File Sharing		Non-Real Time	384	1024	–	–	2 [MB]
Web Browsing		Non-Real Time	30.5	500	–	–	180 [kB]
Social Networking		Non-Real Time	24	384	–	–	45 [kB]
Email		Background Traffic	10	100	–	–	300 [kB]
M2M	Smart Meters	Background Traffic	–	200	–	–	2.5 [kB]
	e-Health	Non-Real Time services	–	200	–	–	5.48 [MB]
	ITS	Conversational Real-Time	–	200	–	–	60 [B]
	Surveillance	Streaming Real Time	64	200	384	–	5.5 [kB]
Mobile Video	Video Calling	Conversational Real Time	64	384	2048	60 [s]	–
	Video streaming	Streaming Real Time	500	5120	13000	600 [s]	–

The characterisation of the performance function for REF is based on 15 simulations, thus having a lower error margin, which is important since these results serve as reference for all the scenarios that are analysed in this work. Those results are presented in Table 4.4.

Table 4.4 Performance Parameters for REF Scenario

Parameter	Mean	St. Dev.
LTE Load	16.50	0.37
WiFi Load	10.71	0.55
Drop Rate [%]	3.31	0.48
Average Delay [ms]	13.05	1.47
# HO	132.67	2.16
LTE BR [Mbps]	28.34	1.26
Wi-Fi BR [Mbps]	17.36	0.86
Average Users	71.37	0.90

4.2 Performance as a function of user-related parameters

4.2.1 User density

In this case, three simulation scenarios were processed, considering densities of 8000, 12000 and 14000 users/km^2 besides the reference scenario with a density of 10000 users/km^2.

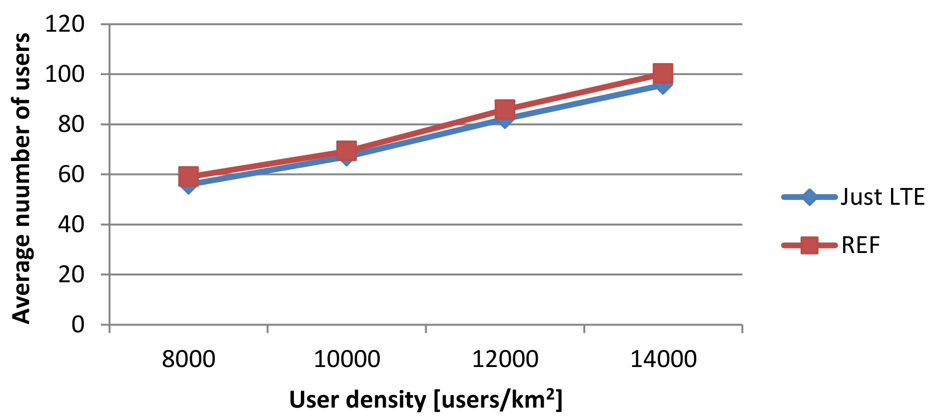


Figure 4.3 Average number of users variation with users density.

The increase of network users can be observed in Figure 4.3, where the average number of users is represented. The variation of this parameter leads to an increase of 10 active users in a given time instant. The variation between the number of users in the REF and the LTE only scenarios is minimal, however, in every scenario, the REF scenario presents a value slightly higher since it has a larger total coverage area due to its number of BS and how they are spread over the area.

This increase of active users is responsible for a growing tendency of the load index of the RANs presented in the scenarios, as can be seen in Figure 4.4, being also responsible for the growing tendency of the throughput that is seen in Figure 4.5.

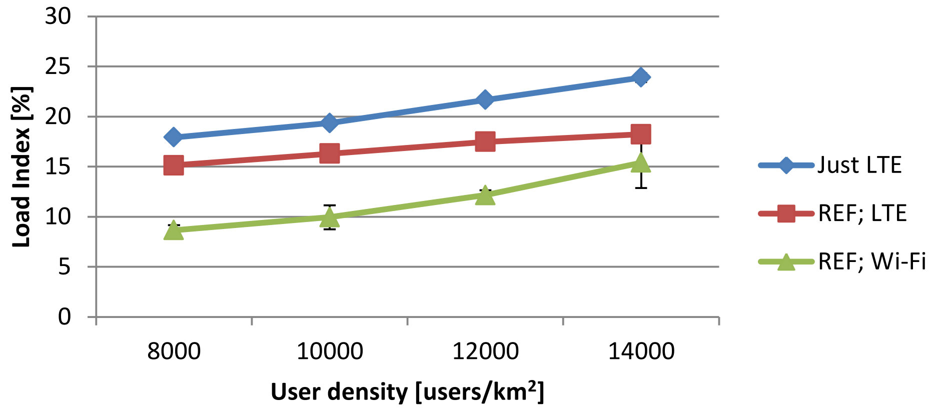


Figure 4.4 Load index variation with users density.

The placement of Wi-Fi APs makes the average load of LTE to lower up to around 5%, which may be relevant in some situations and that will improve the overall system performance.

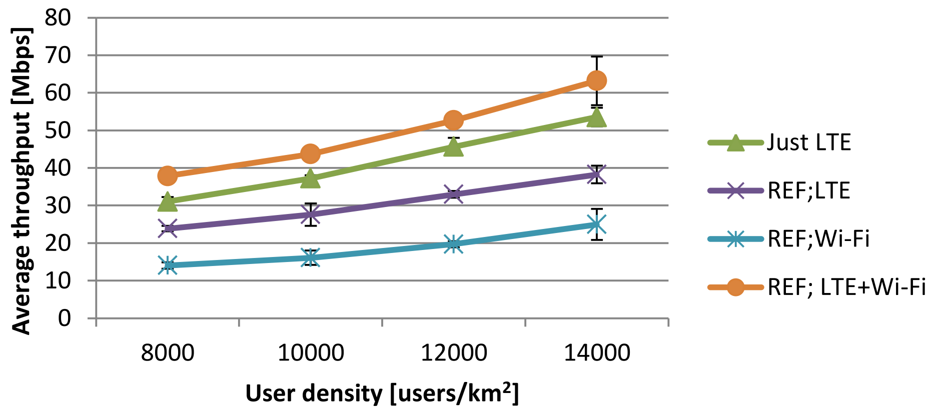


Figure 4.5 Average throughput variation with users density.

Although, both REF and Just LTE scenarios have a similar average number of users, with both LTE and Wi-Fi, which is the case for REF, a higher throughput can be offered to users. In the REF scenario, LTE presents a higher throughput than Wi-Fi, which can easily be explained by the number of users connected to each one, which is related to the priority list.

The drop rate, presented in Figure 4.6, and average delay, in Figure 4.7, are factors that allow measuring QoS. On the one hand, drop rate presents values above expected, however these values are somehow related to the coverage area, and a larger coverage area leads to lower drop rates, as seen later in this thesis. On the other hand, average delay presents a growing tendency with the growth of user density, but always with values under 50 ms, which represents an acceptable performance when analysing QoS.

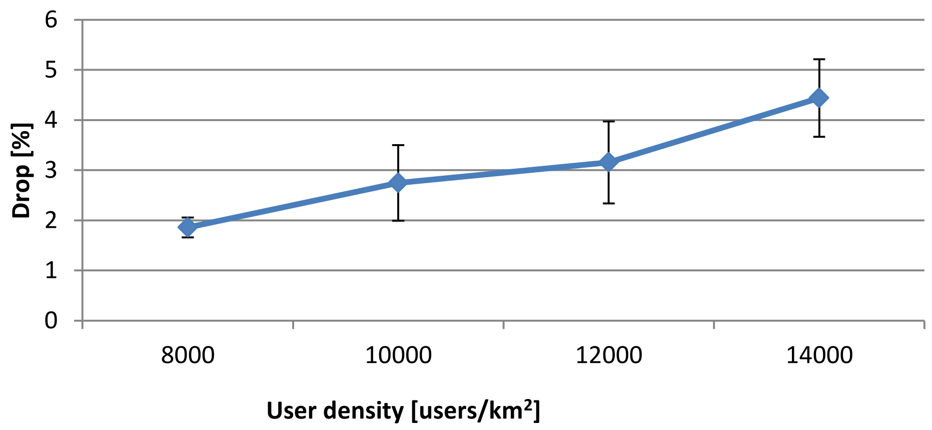


Figure 4.6 Drop rate variation with users density.

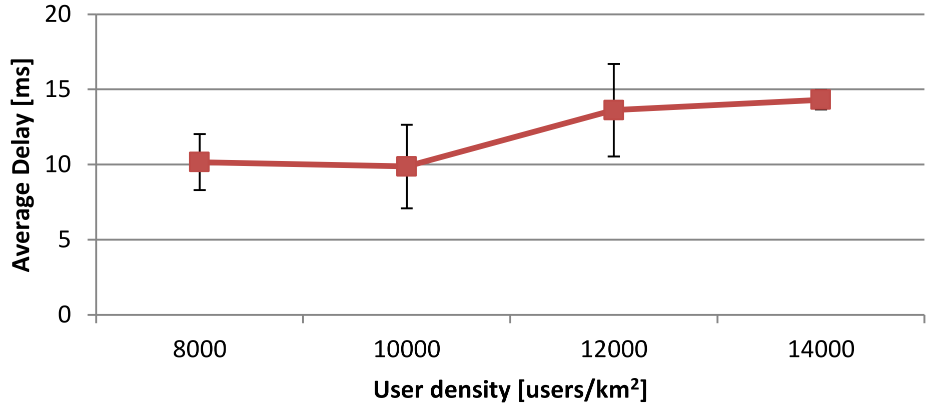


Figure 4.7.Average delay variation with users density.

Since user density corresponds to increasing the concentration of users in the area, consequently more handovers are performed. This can be observed in Figure 4.8, except for the REF scenario that presents a value similar to the scenario with 14 000 users/km^2. This disparity may be explained through the whole randomness associated with the simulator, especially in the generation of users, their movement and in the traffic models. Note that for these simulations, all handovers were successfully performed.

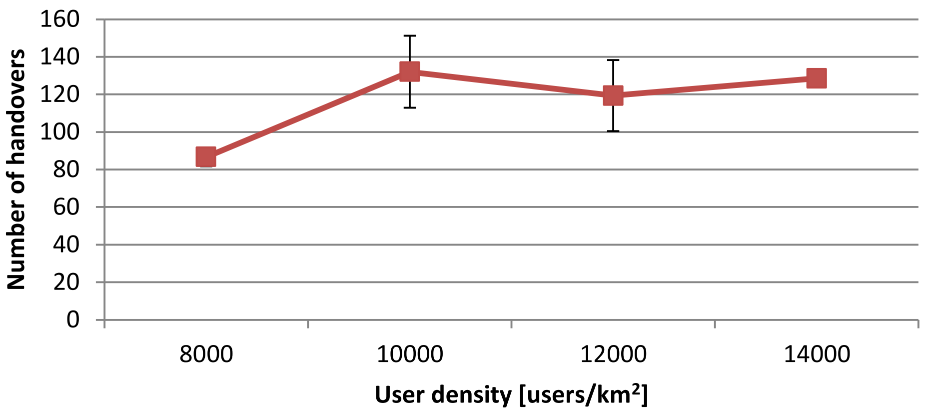


Figure 4.8 Number of handover variation with user density.

4.2.2 Geographical distribution of users

In this section, the distribution of users is analysed. In REF, users are assumed to be uniformly distributed in space, and now one additional scenario was tested in which users are distributed according to a bidimensional Normal distribution. This distribution concentrates users around a central point, with a certain standard deviation (in this case equal to 150 m). This distribution simulates a real distribution of users in locations as shopping centre, for instance.

Since this distribution concentrates users around a central point, the average number of users is higher (more users covered), which leads to a higher load index, i.e., higher resource consumption. This can be seen in Figure 4.9, where it is observed that for the normal distribution the number of active users is around 20 users higher; this increase is reflected in both the load index and the throughput.

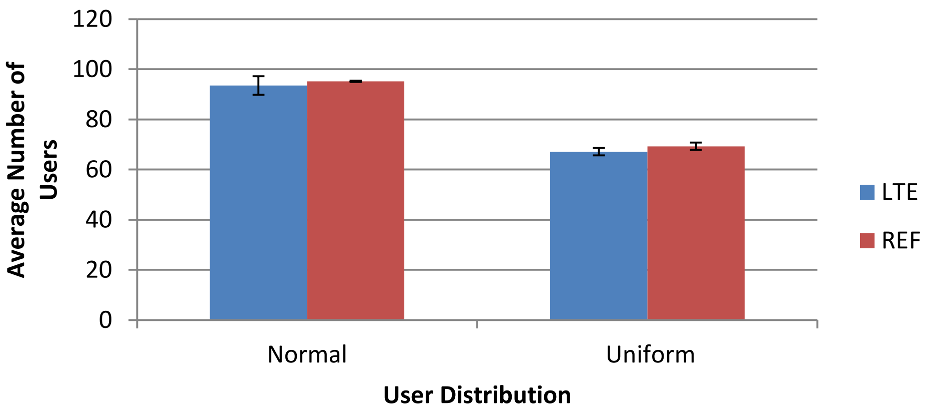


Figure 4.9 Average number of users variation with user distribution.

Regarding the load index, comparing both distributions, load is expected to grow in the scenario where the normal distribution is used, since there are more users in the coverage area. As for the scenario with only LTE BSs and normal distribution, load decreases about 5% for LTE BS when Wi-Fi APs are added.

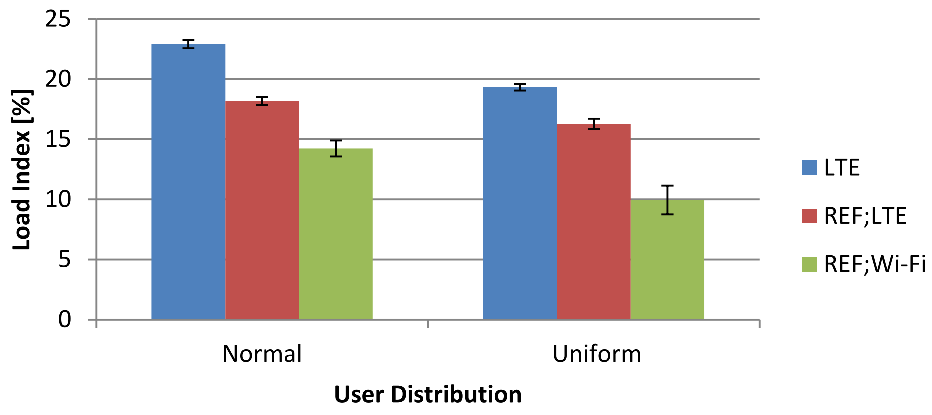


Figure 4.10 Load index variation with user distribution.

As for the throughput, it is also expected to increase when a normal distribution is used. More users are connected thus using more resources and consequently higher throughput is expected

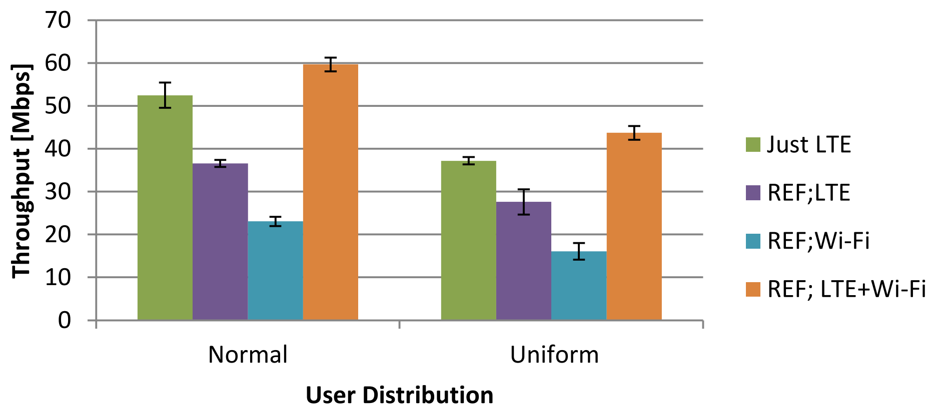


Figure 4.11 Throughput variation with user distribution.

Although the average delay suffers a raise, this raise is not significant since it is around 3 ms and thus having acceptable performance values for QoS.

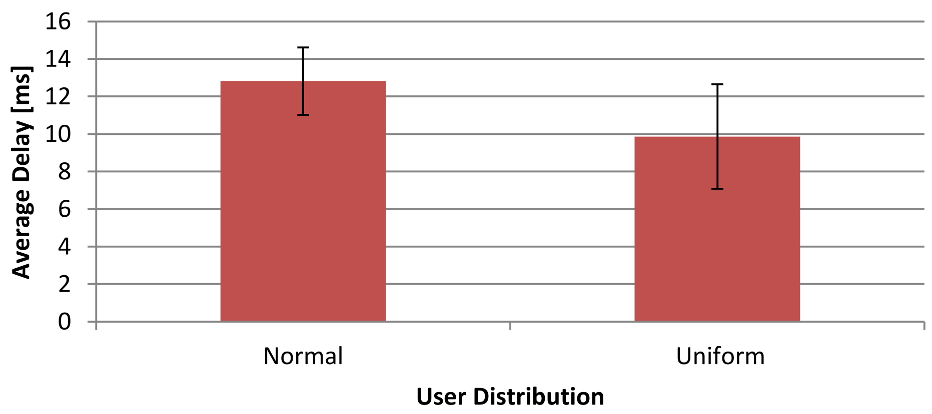


Figure 4.12 Average delay variation with user distribution.

Regarding drop rate, the situation is more complex, since for the normal distribution it raises to 7%. The Normal distribution should reduce users in cell boarders, which contribute for the decrease of drop, and the concentration of users should contribute for handover failure and drop increase. In this case, the drop rate increase, which means that the concentration of users is the dominant effect over the reduction of cell boarder users.

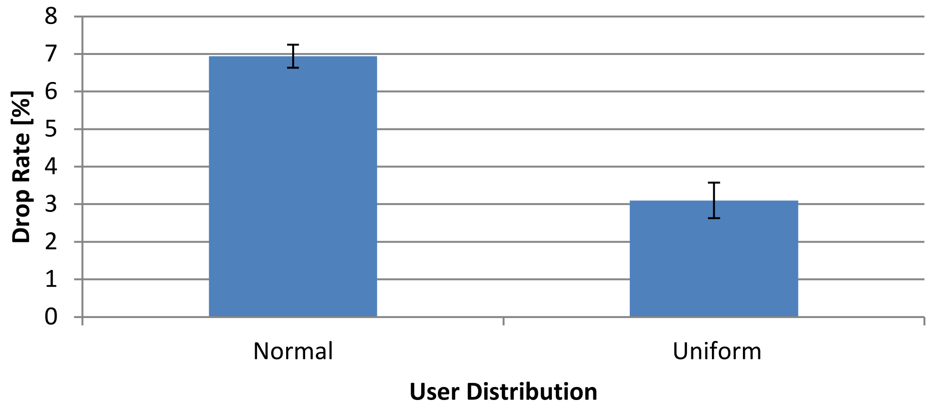


Figure 4.13 Drop Rate variation with user distribution.

As for handovers, the Normal distribution presents a lower number of handovers performed. This is explained by the fact that users are more concentrated in space, and thus the need to perform handover decreases because the QoS can be achieved without that need.

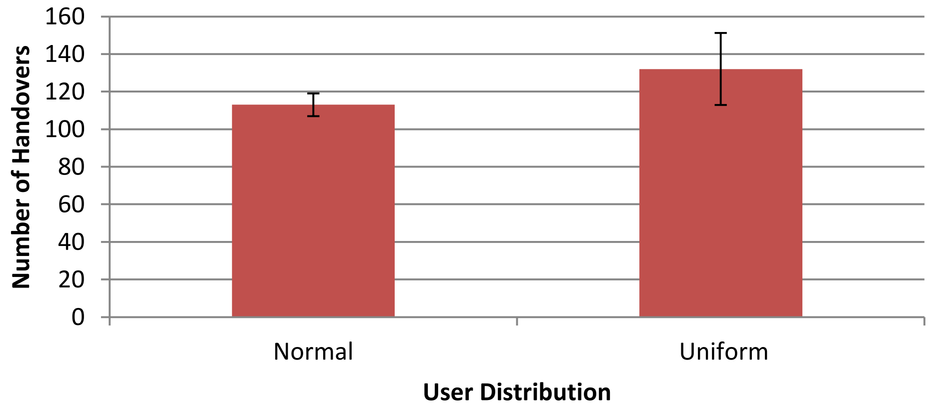


Figure 4.14 Number of handovers variation with user distribution.

Although there are some failures when performing handovers, there are a maximum of 2 failed handovers per simulation, which leads to a mean values below 1%, the acceptable performance for this parameter.

4.2.3 Service penetration

The objective of this case is to simulate reduction in penetration of services of the REF scenario in benefit of others and in order to do this study, three scenarios were designed. In the first one, the penetration of M2M services was raised till 30% while dropping penetration in the other services equally. As for the second scenario, various small changes were made on service penetrations. Voice and Video stayed with 5%, M2M and streaming services with 20%, 10% for FTP and Email services and 30% for social network and internet services. For the last one, Video, Voice and FTP services with 5%, 20% for social network and internet services, M2M and email with 10% each and finally 45% for streaming.

As observed in Figure 4.15, a raise of 10% on the average number of users is verified when the streaming scenario is analysed compared with the other scenarios, which is easily explained by the average time of use of the streaming service.

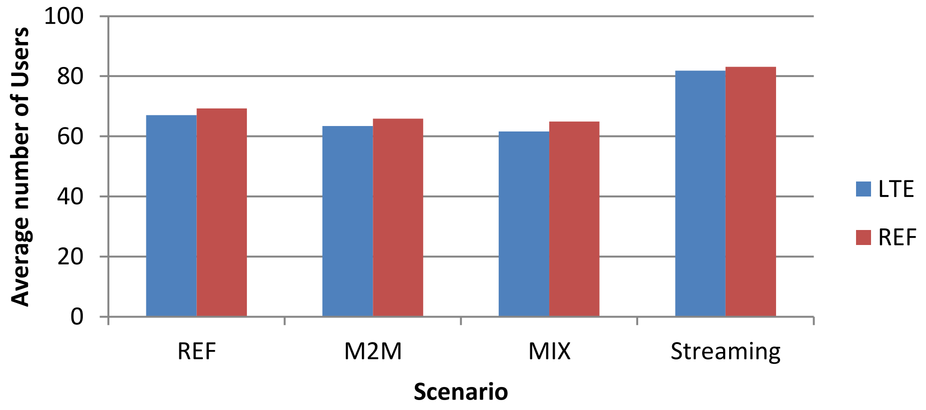


Figure 4.15 Average number of users variation with service penetration.

As presented in Figure 4.16, the first 3 scenarios present the same number of users and although they have different penetrations, the load index does not vary that much. However, the raise in the number of users in the streaming scenario is reflected in the load index, having around more 6% of resources used. As it is common, and as seen and explained previously, when comparing the LTE only scenario with the REF one the load decreases for LTE in all the presented scenarios.

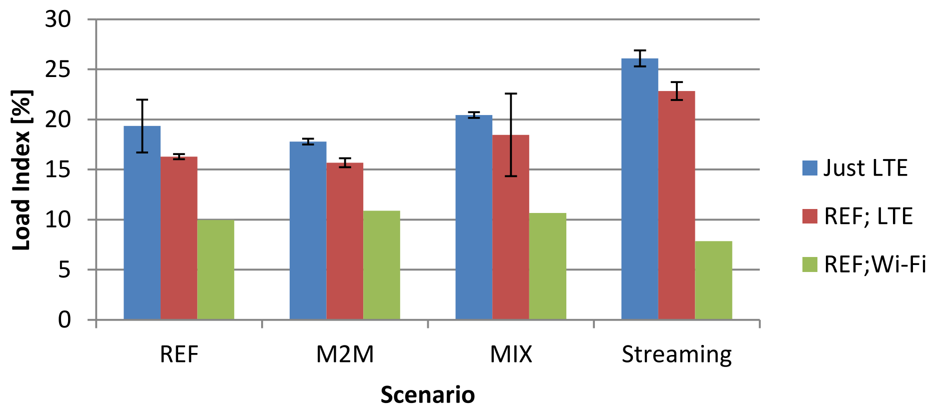


Figure 4.16 Load index variation with service penetration.

As seen in Figure 4.17, scenarios MIX and Streaming present the higher difference regarding average throughput. REF and M2M do not present any major variations, since there were not many changes in the penetration of the services. However, for REF and MIX scenarios the difference for the other scenarios is noticeable. For MIX, the largest contribution for the increase of average total throughput is the increase of the LTE throughput due to penetration of services, since the penetration is higher for services that are prioritised for LTE. The same happens to the streaming scenario further adding an important factor, the increase of users. The largest contribution comes from LTE as it is the preferred network for this type of service. Note that Wi-Fi has a lower throughput for the streaming scenario, since the services that have Wi-Fi as the preferred network got their penetration values lowered.

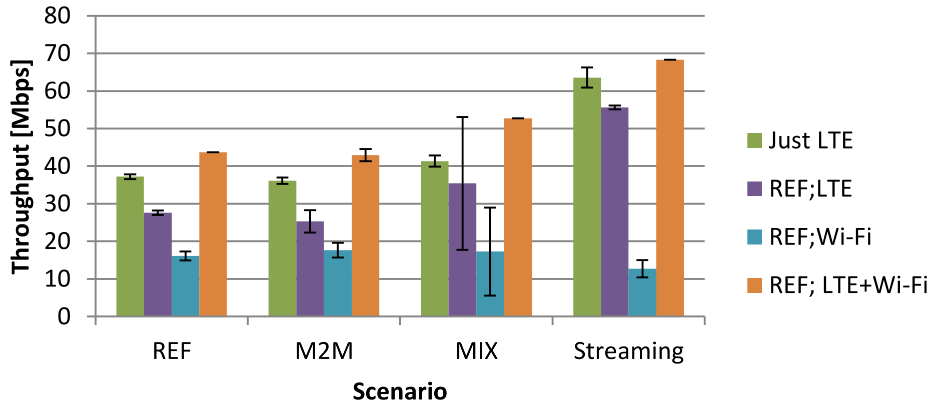


Figure 4.17 Average throughput variation with service penetration.

As for the average delay, all scenarios studied had their values below the maximum accepted value thus not being a problem for QoS.

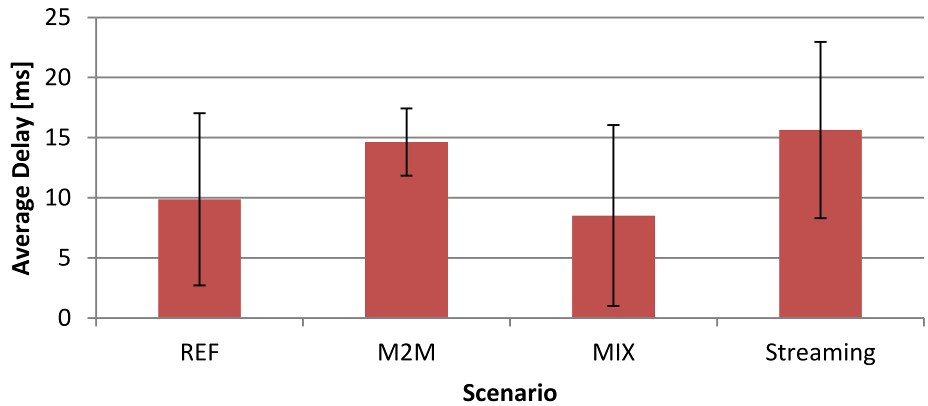


Figure 4.18 Average delay variation with service penetration.

All scenarios had the number of handovers performed reduced. The reduction of this parameter for MIX and Streaming does not have a clear explanation, apart from the randomness of the simulator and the number of simulations done for these scenarios and for the REF scenario. However for the M2M scenario, such decrease can be explained by the characteristics of these services as well by the stationary nature of those users, since most of these users are considered not to have any movement or to have a reduced movement. This factor is also reflected in the drop rate, which can be observed in Figure 4.20.

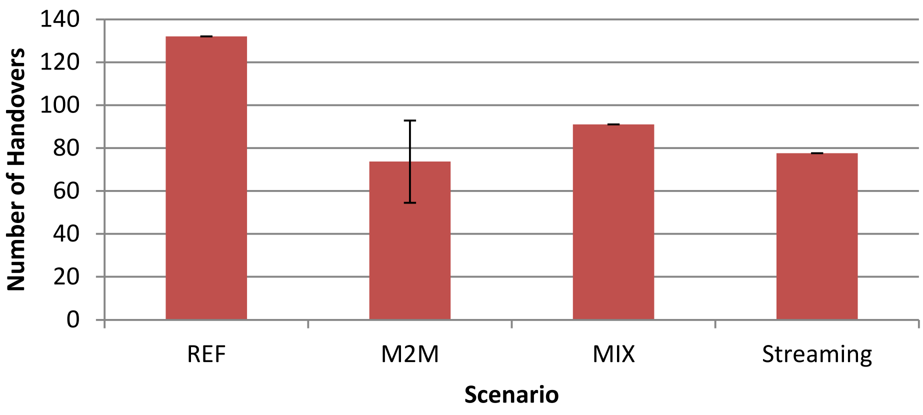


Figure 4.19 Number of handovers variation with service penetration.

Drop rate values still are above the maximum accepted value, but it should be noted that these scenarios do not contemplate a full coverage, so there are a lot of users that are not within the range of the BS, and there are users that are in the cell edge, thus contributing to the increase of this parameter.

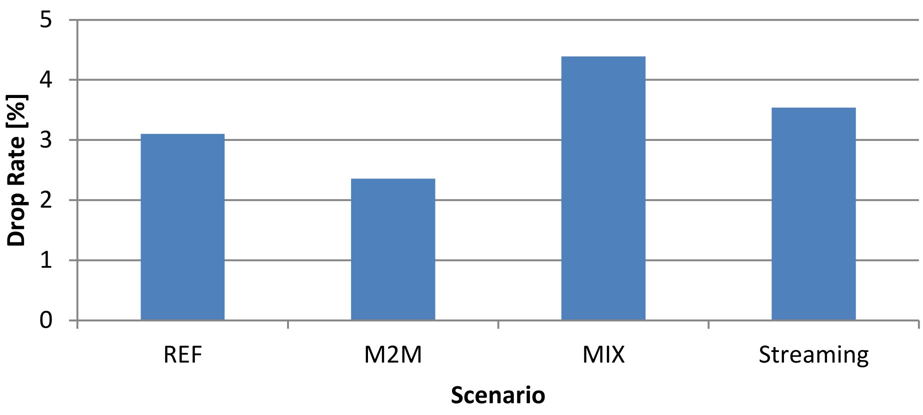


Figure 4.20 Drop rate variation with service penetration.

4.3 Performance as a function of network-related parameters

4.3.1 Priority table

In order to test this input parameter, two scenarios were studied; where in both the priority of a service was changed. In the first LTE is preferred for FTP, while in the second Wi-Fi is the chosen for streaming services. These changes lead to variations regarding the number of users for FTP scenario, which had an increase of 4 to 5 users.

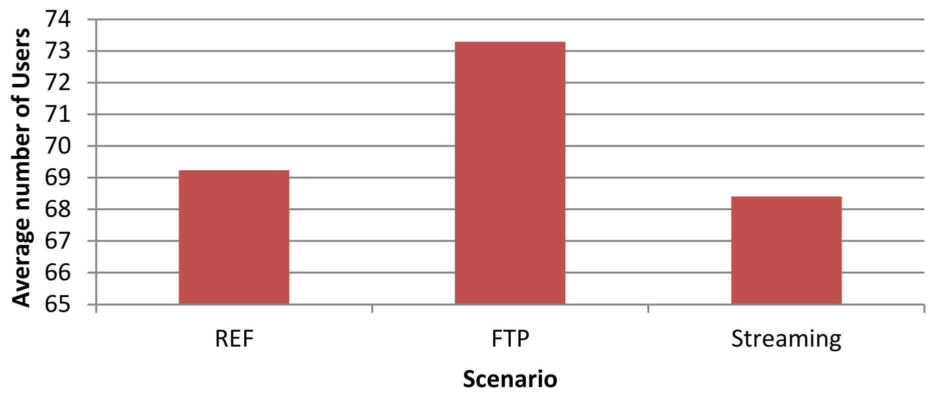


Figure 4.21 Average number of users variation with priority table.

Variations in the number of users are reflected in the load index. For the FTP scenario, comparing with REF, it is expected that the load for LTE increases and for Wi-Fi decreases, because LTE has become the preferable network for FTP instead of Wi-Fi. The same analogy goes for streaming: Wi-Fi has become the preferable network which leads to an increase in its load and a decrease in LTE load.

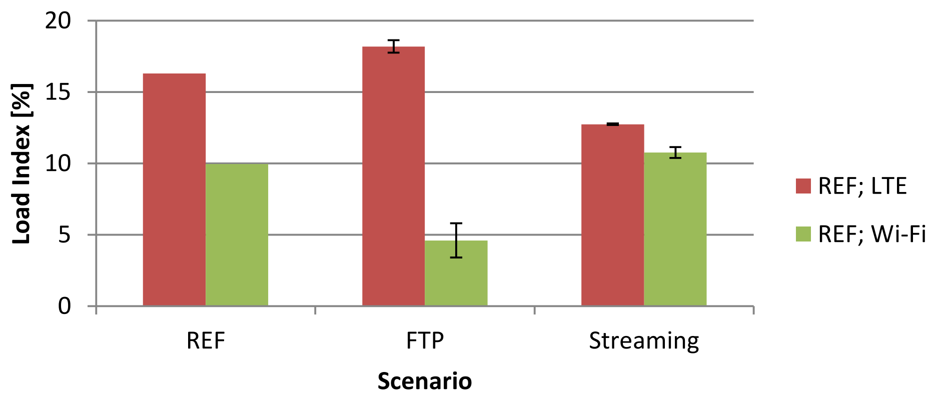


Figure 4.22 Load index variation with priority table.

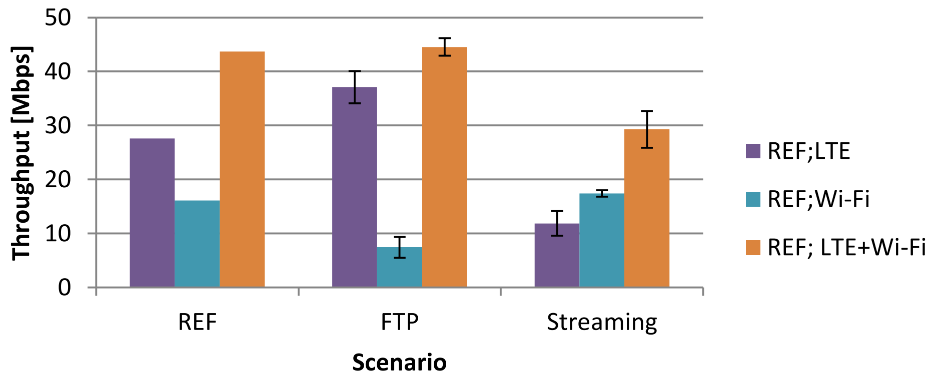


Figure 4.23 Average throughput variation with priority table.

These variations also have an impact on the throughput. LTE shows a bigger average throughput for the FTP scenario, as expected. As for the Streaming scenario, it is expected that Wi-Fi presents a higher throughput. The reason for these increases of throughput is the priority list and it shows that the priority list has a major role in the balance and QoS of the users.

As a consequence of the raise in the number of users in the FTP scenario, the average delay increases by 7 ms. In the streaming scenario, the number of users does not differ much, therefore it is expected that this parameter takes values similar to the same parameter for the REF scenario. However, for all simulations, the average delay is below the maximum accepted value.

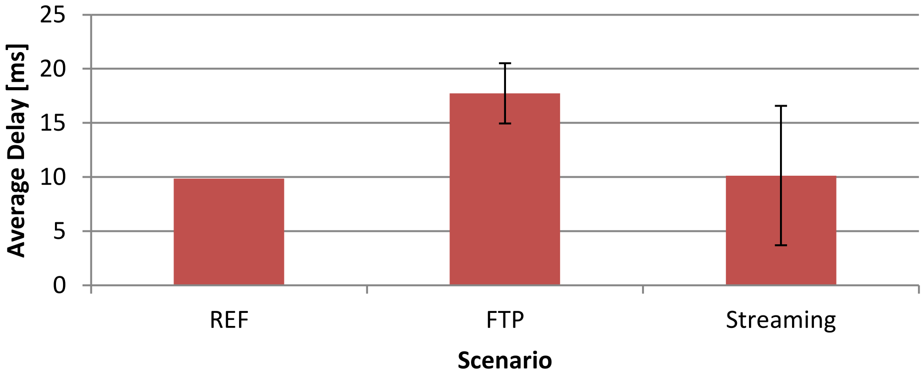


Figure 4.24 Average delay variation with priority table.

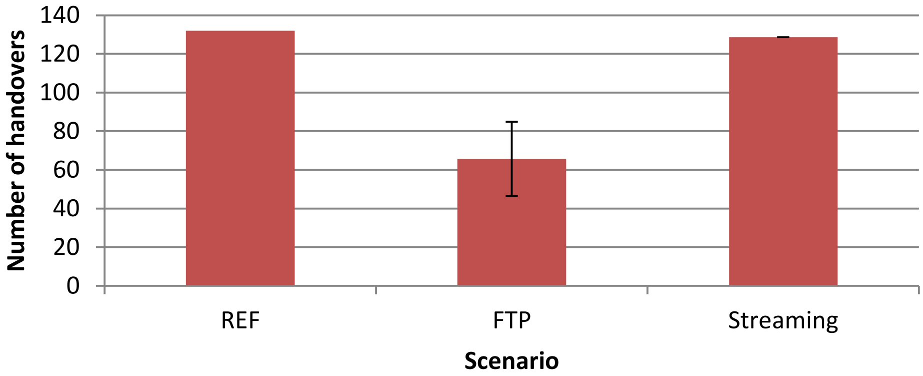


Figure 4.25 Number of handovers variation with priority table.

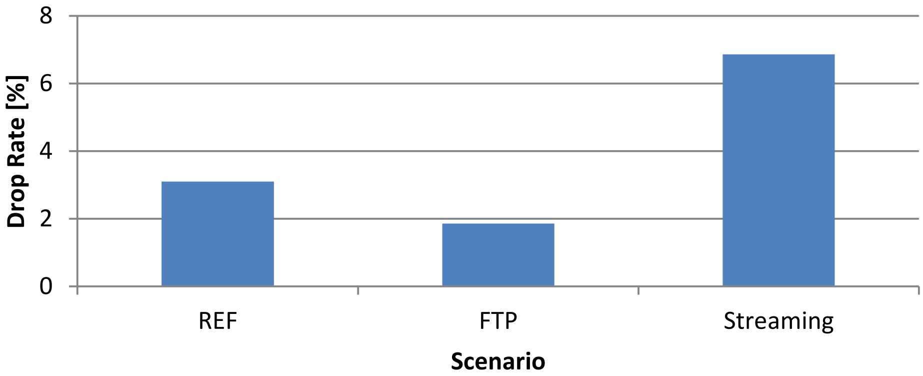


Figure 4.26 Drop rate variation with priority table.

The number of handovers and the drop rate are related to the coverage area and the average number of users. The FTP scenario suffers a decrease in the number of handovers since the coverage area of LTE is larger than Wi-Fi APs and there are more services preferring LTE, which leaves less with the necessity of performing handovers, looking for better QoS and as a consequence drop rate is lower.

4.3.2 LTE bandwidth

In order to evaluate the impact of capacity in network performance, a bandwidth variation with REF with values of 1, 5, 10 (REF) and 20 MHz is analysed.

The first thing one should notice is that with 1MHz, the Load Index reaches the load threshold in the scenario running only LTE, thus having a negative impact in terms of QoS. As a consequence the number of users for this case is far less, by 20 active users, than the scenario with both LTE and Wi-Fi.

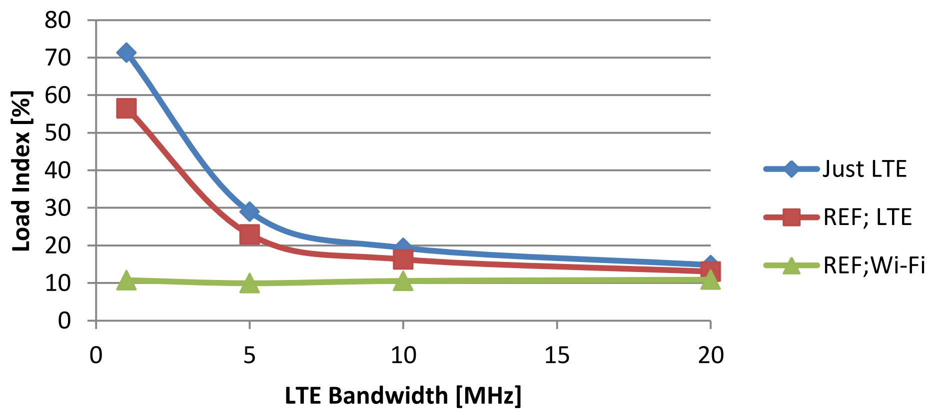


Figure 4.27.Load index variation with LTE bandwidth.

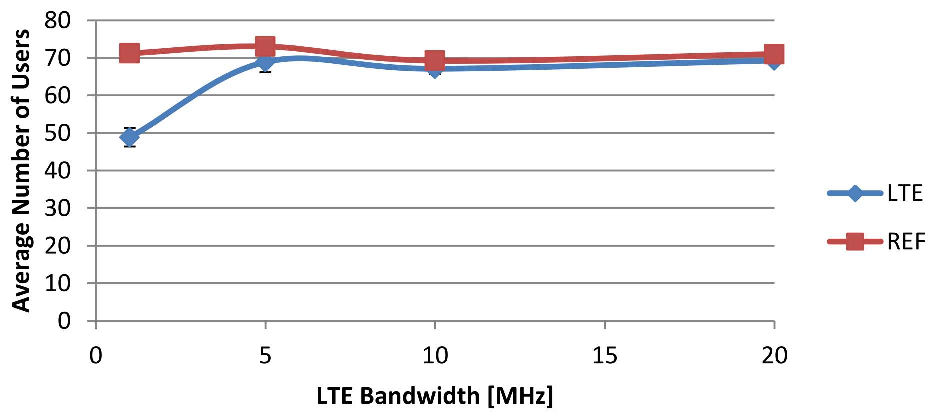


Figure 4.28. Average number of users variation with LTE bandwidth.

As previously mentioned, as a consequence of reaching the load threshold, one can see the gap between both scenarios when the bandwidth takes the value of 1 MHz. For other values of bandwidth, this gap is far less noticeable, having a maximum difference of about 5 users.

Reaching the load threshold also impacts on the throughput since a rearrangement of users is required. For other cases, one can say that throughput does not suffer any major variations. Note that, as already stated before, the REF scenario presents a higher throughput compared with the LTE only scenario.

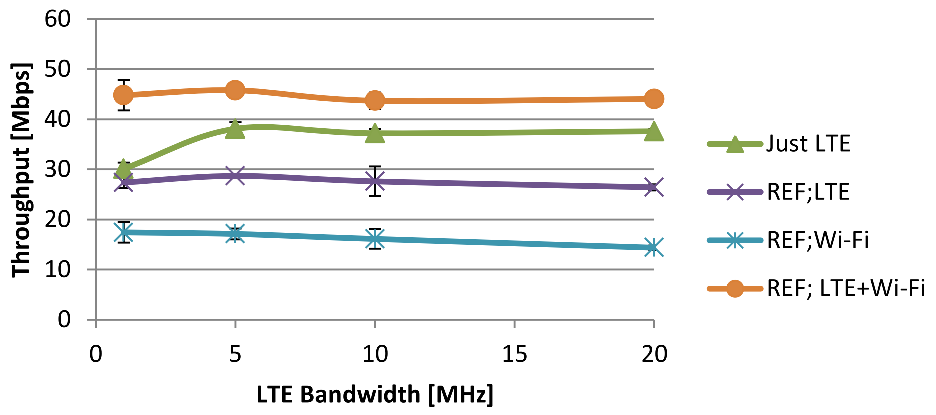


Figure 4.29 Throughput variation with LTE bandwidth.

Regarding QoS parameters, the average delay presents an improvement for values of bandwidth above 5 MHz. This difference, of around 30%, is irrelevant nevertheless, since an average delay of 50 ms is considered acceptable and this value is below that limit. It should be noted that for the case of 1 MHz with LTE being the only option, drop rate, other factor for QoS, reaches values around 30%. For the other cases, drop rate takes values near those that have been presented already, around 3%.

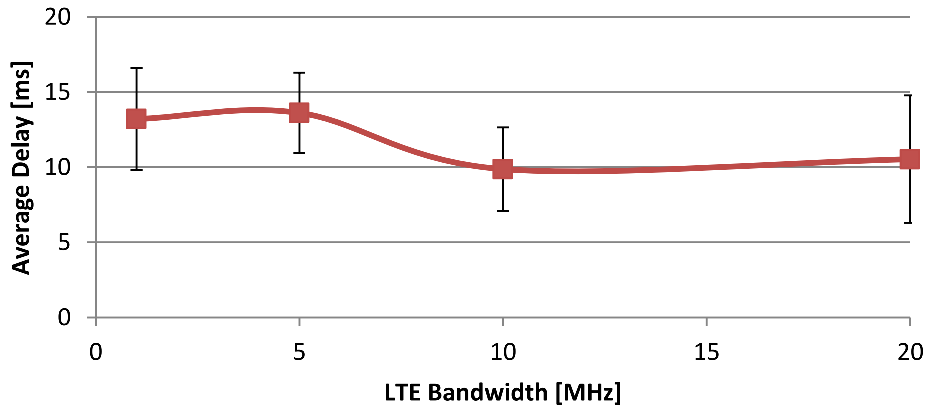


Figure 4.30 Average delay variation with LTE bandwidth.

LTE bandwidth variation does not seem to be efficient to vary the number of handovers. This happens because the REF scenario, the only one that it is relevant for this analysis, does not reach any threshold, so it does not perform any extra routines in order to balance the load. For this scenario there are handover failure rates below 1% which is not alarming since it is below the maximum accepted value.

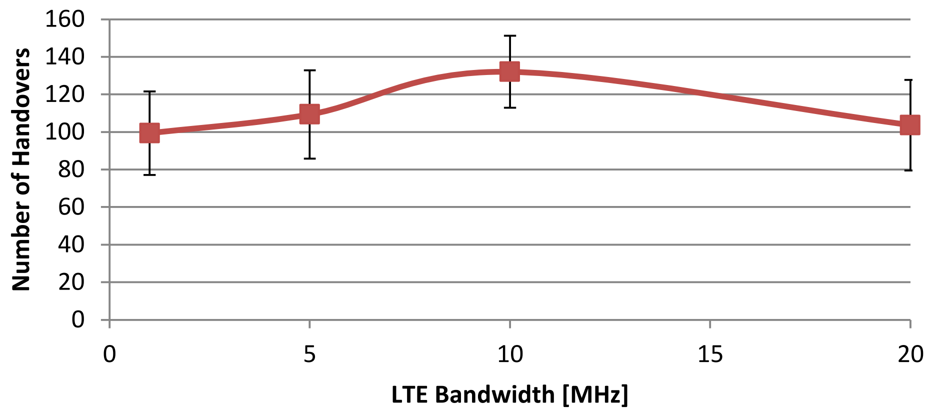


Figure 4.31 Number of handovers variation with LTE bandwidth.

4.3.3 BS load threshold

Load Threshold is an important parameter in the load balance study. Basically this parameter, upon load index of a BS reaches the threshold, forces users to move out of the BS in order to distribute the load among the neighbouring BS with the objective of providing the best QoS possible.

Since, the user generation could not generate a sufficient amount of users in order to get higher load, this parameter was lowered to a level that could be achieved by the simulator.

One presents in Figure 4.32 and Figure 4.33, the number of users and load index, respectively. As it can be observed the scenario that only has LTE BSs is the most affected with this threshold since it does not have where to offload traffic. As a consequence of the threshold for that scenario, the number of users is very low. For instance, with a threshold of 13%, 23 users are connected while with a threshold of 15% this number increases by approximately 15 users. The number of users reaches normal values with a threshold of 17% but still getting a bit less.

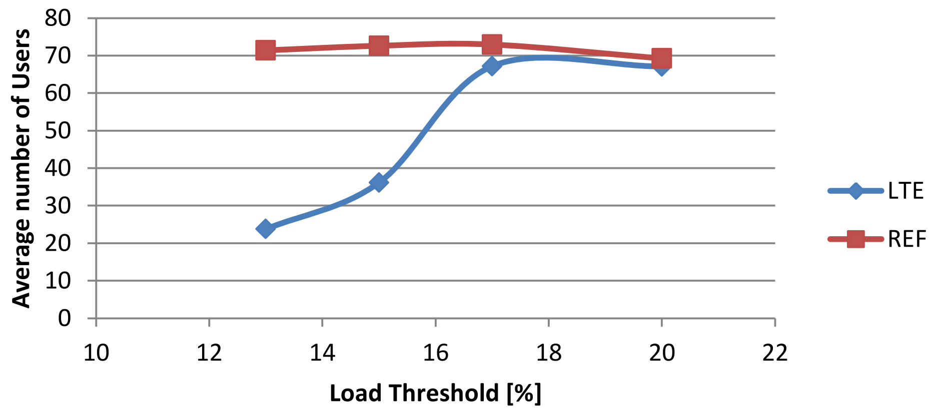


Figure 4.32 Average number of users variation with load threshold.

As the threshold is reached, system runs the load balance algorithm in order to offload traffic from the overloaded cells. This can be observed in the load index, where Wi-Fi load is higher for thresholds of

13% and 15%.

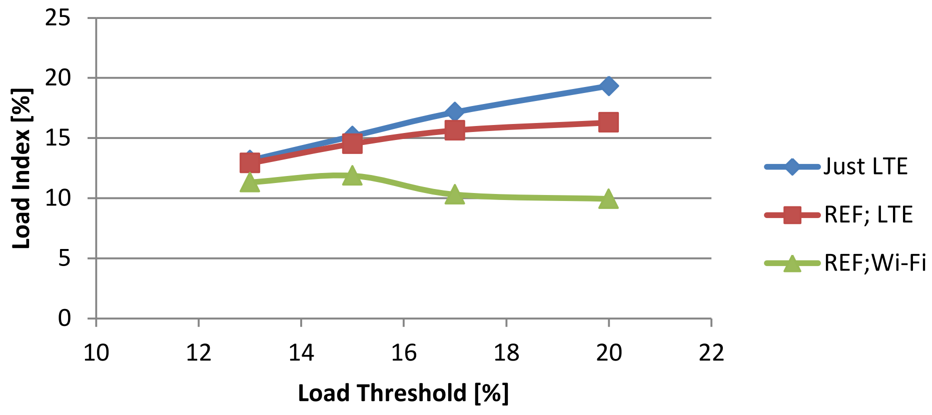


Figure 4.33 Load index variation with load threshold.

The same can be verified for the throughput where for the scenario which only has LTE BS, the throughput offered is very low but as the threshold go up, the throughput also goes up. For thresholds of 13% and 15%, in REF, LTE reaches its threshold and thus offloading traffic to Wi-Fi as can be seen.

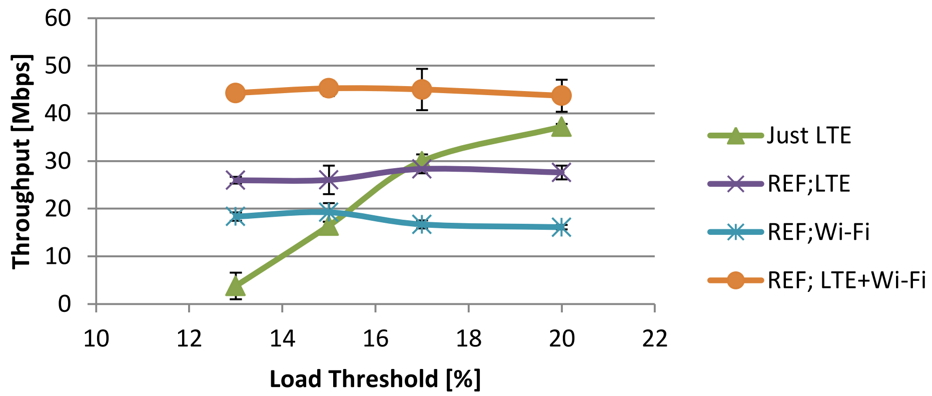


Figure 4.34 Average throughput variation with load threshold.

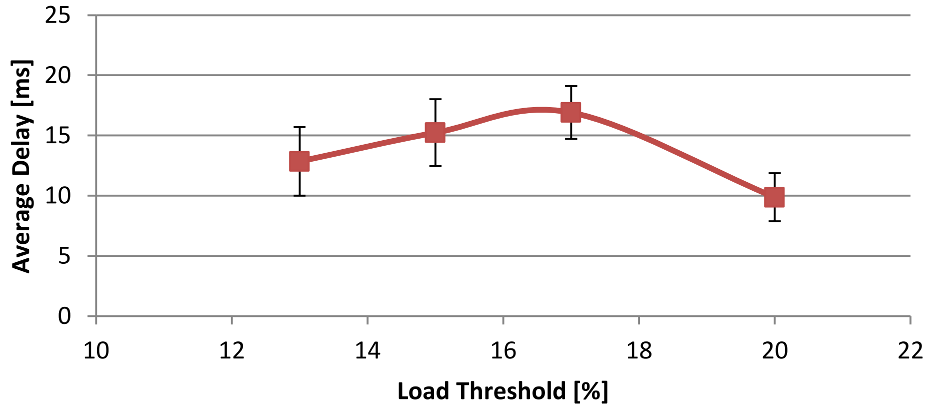


Figure 4.35 Average delay variation with load threshold.

The average delay presents a growing tendency until 17% load threshold and then it goes down. This happens because at some point in the simulations, load reach its threshold therefore having overload problems and thus increasing the delay.

It is expected that the number of handovers increases when the threshold is reached so that traffic can be offloaded to other cell that it is not overloaded. As seen in Figure 4.36, throughout the variation of the parameters, there is not much difference in between the number of HHOs but that is not the case for the VHOs in which the lowest threshold presents the higher number of VHO.

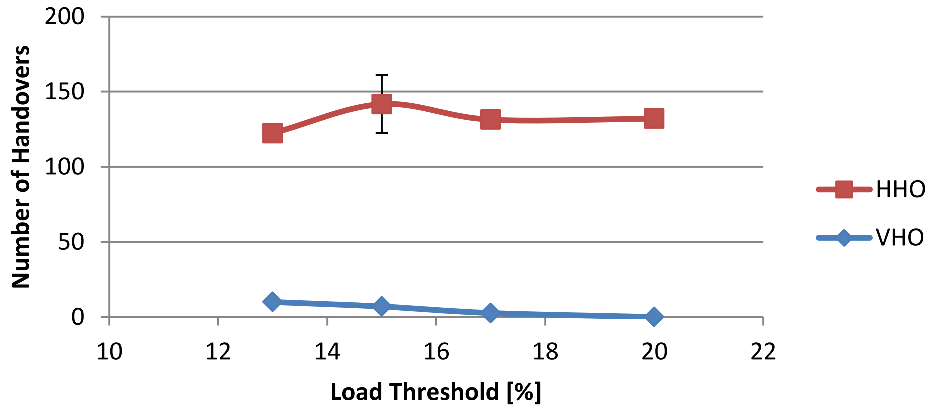


Figure 4.36 Number of handovers variation with load threshold.

4.3.4 Number of BS

For this analysis, three additional scenarios were created in which BSs were added. In the first one, +LTE, an LTE BS was added; the second, +Both, has one LTE BS and a Wi-Fi AP, and the third, AFC (Almost Full Coverage), a total of 3 LTE BS and 3 Wi-Fi AP were added.

As more BSs were added, more users are inside the coverage area of the BSs and consequently more users are connected. This can be verified in Figure 4.37 where +LTE and +Both scenarios present a raise of 5 users and the AFC scenario presents 8 more than REF.

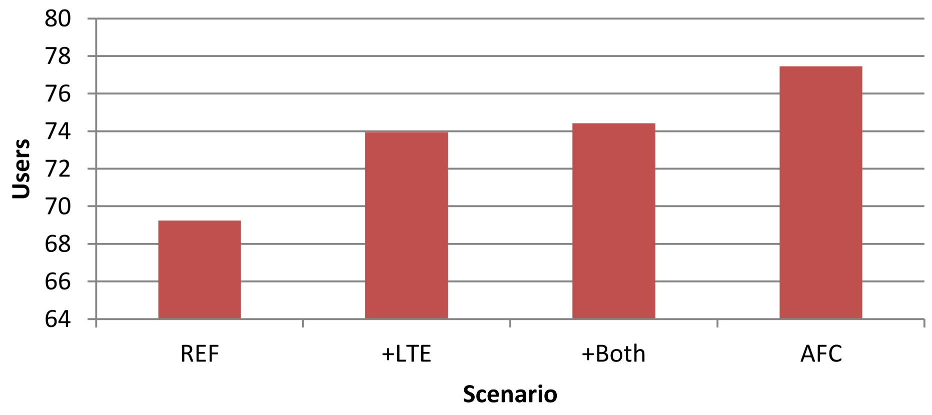


Figure 4.37 Average number of users variation with number of BS.

As already stated, the increase is reflected in both load and throughput. The +LTE scenario adds an LTE BS, so more people are covered, and the average load index is expected to increase, as shown in Figure 4.38. When an AP is added simultaneous with an LTE BS, and since the coverage area of both overlaps, one does not expect an increase but a decrease instead. This happens because there are users connected to LTE although they had preference for Wi-Fi but do not have coverage. Therefore it is expected that LTE load suffers a decrease and Wi-Fi load an increase. As for the last scenario, more users are served thus increasing the load, however, BSs and APs are added and the increase of users is not sufficient to increase the average load index per BS.

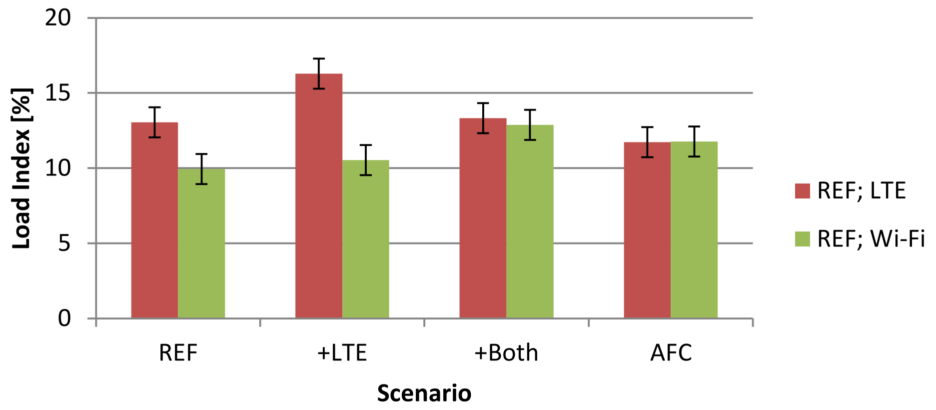


Figure 4.38 Load index variation with number of BS.

A slightly increase is observed in the throughput when BSs are added, which is a consequence of the number of users connected to the BSs.

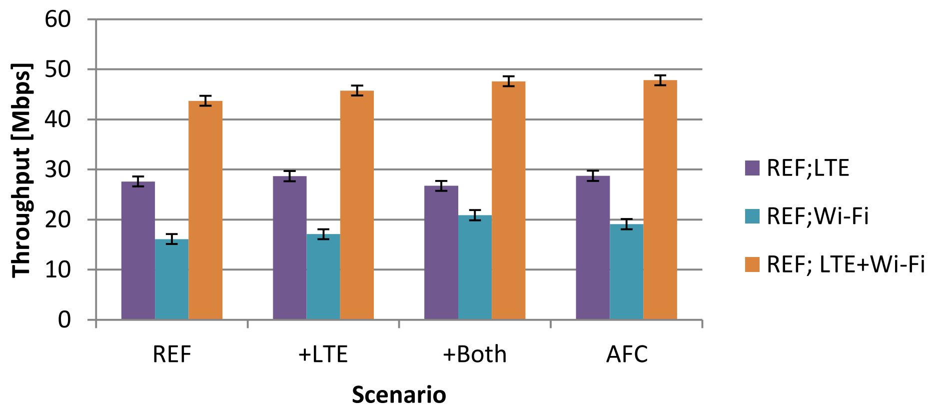


Figure 4.39 Average throughput variation with number of BS.

The number of users and number of BS are factors that contribute to the average delay. In this case, when an LTE BS is added, the average delay increases due to the fact that more users are using the network. If an AP is placed after the LTE BS, this parameter decreases since users now have a choice to use the BS or the AP and thus spreading over both and consequently decreasing average delay. For the last scenario, more BS and APs are placed, which leads to an increase of the average delay and thus showing that the number of users is dominant over the number of BSs.

Note that all scenarios present values below the maximum accepted value for average delay.

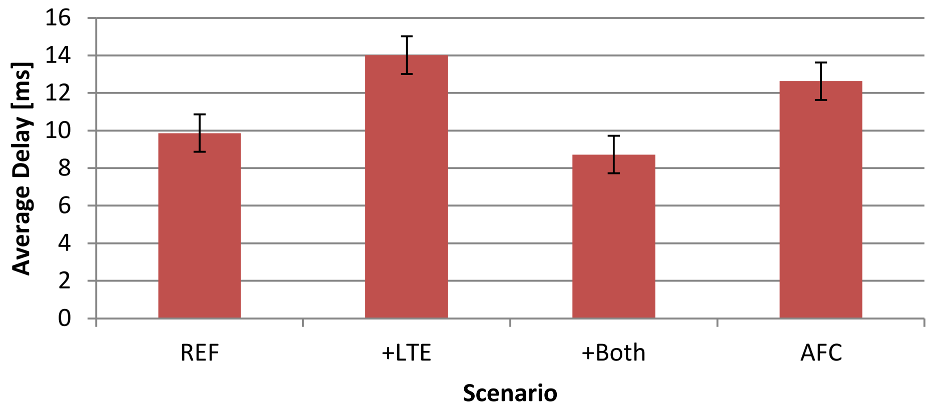


Figure 4.40 Average delay variation with number of BS.

The number of handovers performed in +LTE decreases if one compares with REF, while in +both where an AP and an LTE BS are added it increases. This is explained by the fact that more coverage area is available and users tend to stay in LTE longer than in Wi-Fi as consequence of the coverage area of each BS and thus performing less handovers. The AFC scenario presents a scenario similar to REF but with more BS and APs which leads to number of handovers in the same order of magnitude.

Although there were handovers that failed, handover failure rate never reached 1%, the maximum acceptable value.

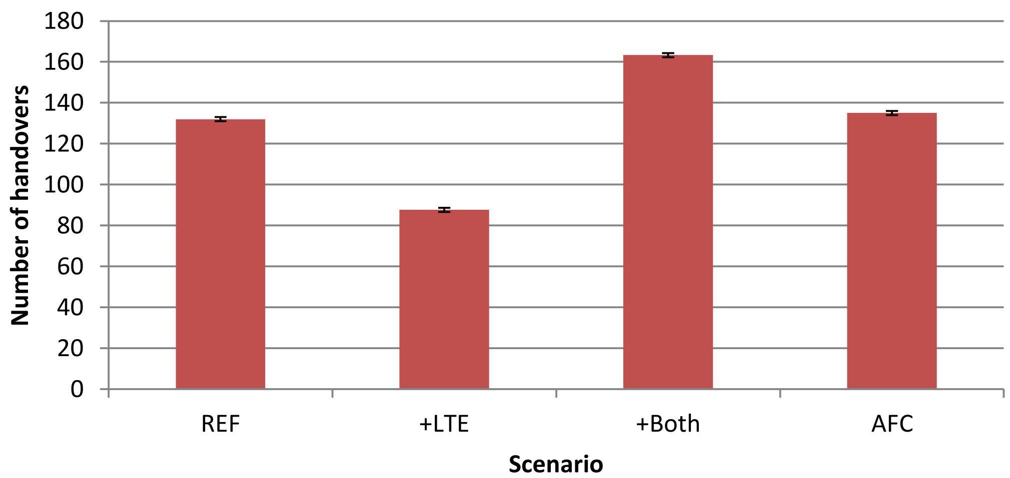


Figure 4.41 Number of handovers variation with number of BS.

A decreasing tendency is observed as more BSs are added. This decreasing tendency is related to the coverage area and the amount of users that are connected because with more coverage, the probability that they are in the cell edge decreases thus decreasing the probability that the connection is dropped.

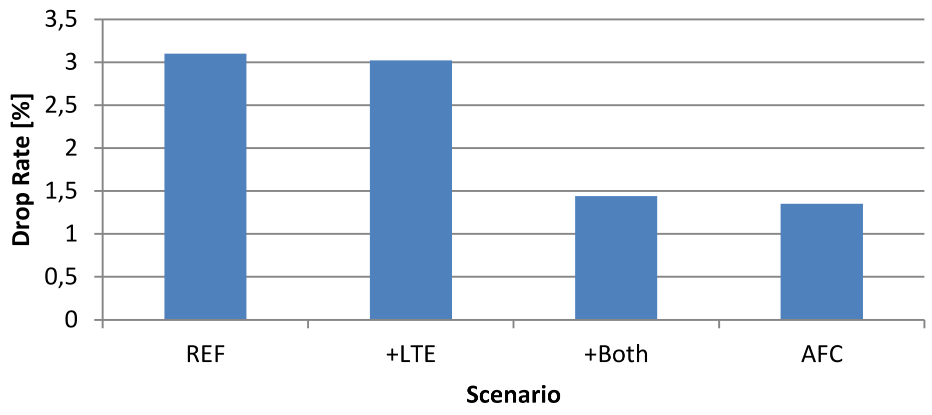


Figure 4.42 Drop rate variation with number of BS.

Chapter 5

Conclusions

This chapter finalises this work, summarising conclusions and pointing out aspects to be developed in future work.

Over the years, several wireless technologies have grown and conquered a place in the world, thus making it more heterogeneous. With different technologies overlapping in space, coexisting in a seamless integrated environment and operators owning distinct networks, it is clear that the next step is to exploit all those technologies in a coordinated manner, thus enhancing user experience and satisfaction.

Although this heterogeneity is already being explored, mobile traffic is growing and operators need to meet those demands. In order to do so, the next performance and capacity leap will come from network topology evolution by using a mix of macro and small cells, and in this case the use of LTE and Wi-Fi, which are not yet part of those heterogeneous networks.

The main goal of this thesis was to establish a load balance algorithm between LTE and Wi-Fi while taking QoS and energy efficiency into account. In order to do this a plan was made. First with the study of radio technologies, load balance and performance parameters, then scenarios development and model development and implementation, and finally results were assessed and the influence of scenarios was studied.

The first chapter presents a breif description of mobile communications systems evolution over time, as well as the growing consumer and traffic demand, followed by an explanation about the motivation and contents of this thesis.

In the second chapter, a theoretical background on LTE's and Wi-Fi's network architecture and radio interface is introduced. Interworking in-between these technologies and load balance, being the main focus of this theses, are addressed. Besides that, a description of services and applications' characteristics is done and finally, in the end of the chapter, state of the art is presented.

Chapter 3 presents the models used in the simulator, a description of its workflow and assessment. The developed model is based on factors such as load index, SNR, coverage area, throughput, priority list and of course the algorithms to perform load balance. There are three algorithms implemented: call admission control algorithm, the so called load balance algorithm, and handover algorithm. The call admission control is the algorithm that, using the priority list, chooses the most adequate network for the service. The load balance algorithm performs the load balance when the threshold is reached. This algorithm sorts the users of the overloaded BS so that they can move to other BS in order to improve QoS. The last one is complementary of the load balance algorithm since without this, users would not move of BS. This algorithm is responsible for moving the users from one BS to other.

Complementing these algorithms, other parameters were also implemented in the simulator such as, load index that is responsible to calculate the radio resource usage, throughput being an important QoS parameter, coverage area to check if users are within the coverage area of a certain BS, SNR to evaluate the power and noise and a priority list that determine which RAN is the preferable for each service.

This model was later implemented into a system level, time based simulator which has been developed over the Microsoft Visual Studio 2005.This simulator is able to reproduce heterogeneous environment composed of a combination of GSM, UMTS, Wi-Fi, WiMAX and LTE technologies, implementing low-level functionalities such as power control, load and access control. This simulator was then used to run simulations with different scenarios, based on a Reference Scenario (REF), and in most cases only a single parameter vary, in order to extract results to evaluate the performance of the load balance algorithms as well as the system performance. The performance is evaluated by the output parameters of the simulator as load index, throughput, delay, drop rate, handovers, connected users.

REF is an urban environment physically located near Saldanha, composed of 1 LTE BS and 3 Wi-Fi APs. Users are uniformly distributed over the area and throughout the whole simulation only one type of service is allowed per user. Voice, Video, Music, File Sharing, Web Browsing, Social Networking, Email, Video Streaming and M2M services are defined. Besides REF, there is one more scenario that is always studied. That scenario consists of REF but without Wi-Fi APs. This happens in order to study what is the impact in terms of load and throughput when Wi-Fi is used to offload traffic.

The simulator is quite complex and its main objective is to reproduce a real network environment and all the aspects related to it, the more accurately as possible. However, approximations need to be applied.

One major simplification of this simulator is the fact that it only accounts for the downlink channel. Other limitation is the fact of REF's network structure being a single cluster which may be responsible for unrealistic values of handover failure and drop rate, mainly caused by users in the edge of the cells. Another limitation is the fact that the simulator only focus is the radio perspective, this way ignoring transmission and core aspects. The simulator also uses approximations that introduce distortion, e.g., the granularity of LTE's scheduling or the fact that uses Walfisch-Ikegami model to estimate propagation loss in LTE.

Chapter 4 starts by providing a brief description of the scenarios, followed by the analysis of the scenarios. This analysis is divided by user parameters and network parameters.

Regarding results, one starts with the focus of this thesis, load. This parameter, throughout the whole simulation, presents up to 5% of reduction of load when Wi-Fi APs are introduced. This reduction may have a huge impact on the QoS when dealing with higher load scenarios. This is the trend during the whole set of simulations. This parameter follows the tendency of the connected users. However, if high throughput is needed for a given service this tendency might not be true.

As for throughput, it is observed that similarly to load, the introduction of Wi-Fi APs introduces a change in this parameter but instead of being a reduction, it is a raise which means that with the introduction of APs, the network can provide higher throughput. As it happens with load, this parameter follows the tendency of connected users, but the services and the penetration that users are using are important because it is those services and penetration that dictate the throughput.

The average delay is the measure of the delay affecting the services. Throughout all simulations, this parameter always presents values below the maximum accepted value. However, this parameter is affected by many parameters but it had not troubling modifications (up to a maximum of 20ms).

Another parameter to characterise performance is the number of Handovers. It is observed that a linear dependency with the density of users exists in the interval of study. It is also observed that scenarios with more Wi-Fi APs present a higher number of handovers. This parameter is also affected by the priority table, mainly when there is a raise in the number of services that prefer LTE, which can be explained by the higher coverage area thus there is no need to perform a handover since LTE can provide the QoS required. The number of VHOs is not significant since in most of the scenarios the threshold is not reached. Failure rates of this parameter are always below 1% which is the maximum accepted value.

Regarding the drop rate, the maximum value is around 7%. Although this value is much higher than the 1% considered to be the maximum accepted value, this value is easily explained by the fact that the majority of the scenarios only contemplate a cluster, which leads to poor coverage and as a consequence a lot of users are out of coverage and/or in the cell edges, where they have a great impact over this parameter. In order to justify this fact, when more BSs and APs are added to the scenarios, this value decreases to around 1.5%. This value is still above the acceptable one, but it is a value near it and it should be noticed that there are still users out of coverage, and since one has a larger coverage area, more users are in the cell edge. The value of drop rate is also affected by the priority table. In this case, concentrating more services in the Wi-Fi causes the drop rate to increase 4%.

In the analysed cases, the implemented model presents acceptable values and so it can be concluded that it is a good model to use in order to offload traffic from LTE to Wi-Fi, because load decreases when Wi-Fi APs are introduced in the scenario, since more resources were available. More important, with this introduction more throughput can be provided to users and all this without degradation of QoS, where in some cases some of the performance parameters are even better.

However, it should be noted that all of these results were obtained with low values of load (maximum of 26% load). This means that these values must be carefully analysed if it is to consider a higher load. In other words, these values, especially for handover failure, drop rate and average delay, may not be below such QoS's maximum acceptable values. However, the obtained values are much lower than the maximum acceptable values, thus leading to believe that the model could be applied to scenarios with higher load and still having those performance values below the acceptable values.

Regarding the topic of interworking between LTE and Wi-Fi, there is a lot of work to do. This thesis focused on a simple algorithm, based on a table of priority, but it can evolve towards interesting research targets. First of all, a scenario with higher loads should be studied and in order to do so, a better user generation module should be developed. A cost function should be implemented. A more robust load balance algorithm could be developed, for instance, where one could add a threshold for low loads and when it is reached, users are forced to move to other BS, and when there are no more users the BS would be in a hibernating state, so that the network can be more energy efficient. The priority table should be optimised. In the algorithm, instead of moving services, one should move users, or groups of users. A smart resources allocation and load balance algorithm making decisions based on the movement of the MTs should also be implemented.

Annex A

Mobility Models

This annex describes the mobility models present in the simulator.

A.1 Random Walk Mobility Model

The Random Walk Mobility Model [CaBD02], Figure A.1, was developed to mimic the heretic behaviour of MT's giving a memory-less mobility patters, as each step is calculated without any information of the previous one. At regular time intervals, both angular direction (uniformly distributed in [0º,360º] and speed of MTs are updated.

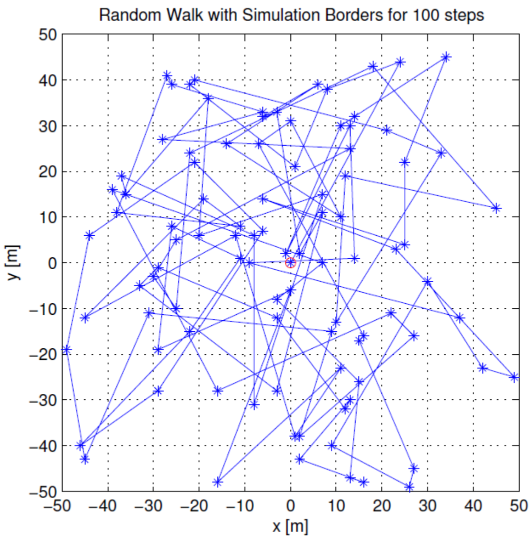


Figure A.1 Random Walk mobility pattern (extracted from [CaBD02]).

MTs bounce at the border of the simulation area, in such way that they can never roam outside this area. If the MTs speed or direction is updated frequently (short time intervals or distances), then it does not wander far off from the starting position. This effect can be useful when, e.g., one is simulating a semi-static network. If one wants to simulate more dynamic networks, larger values for the time interval or distance travelled should be used.

A.2 Triangular Distribution Mobility Model

The model presented in [ChLu95] considers a triangular distribution for speed, where the density function is given by:

$$f(v) = \begin{cases} \frac{1}{\Delta^2}[v-(V_{av}-\Delta)]\,, if\ V_{av}-\Delta \le v \le V_{av} \\ -\frac{1}{\Delta^2}[v-(V_{av}+\Delta)]\,, if\ V_{av} \le v \le V_{av}+\Delta \\ 0, otherwise \end{cases} \tag{A.1}$$

Five different mobility types, Table A.1, with average

$$V_{av} = \frac{V_{max} + V_{min}}{2} \tag{A.2}$$

And deviation

$$\Delta = \frac{V_{max} - V_{min}}{2} \tag{A.3}$$

Are considered for the speed, Figure A.2

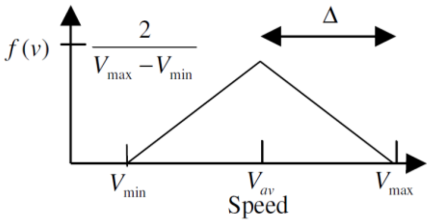


Figure A.2 Velocity probability density function (extracted from [Chle95]).

Table A.0.1 Mobility type speed characteristics (adapted from [Chle95]).

Mobility type	V_{av} $[m/s]$	Δ $[m/s]$
Static	0	0
Pedestrian	1	1
Urban	10	10
Main Roads	15	15
Highways	22.5	12.5

A.3 Model Assessment

The developed mobility model was validated. Figure A.3, presents the speed histogram resulting from simulations. The speed triangular density shape is very clear, thus, one may conclude that the speed random generator complies with the model specifications. Note that other parameters involved (direction, distance) were already validated.

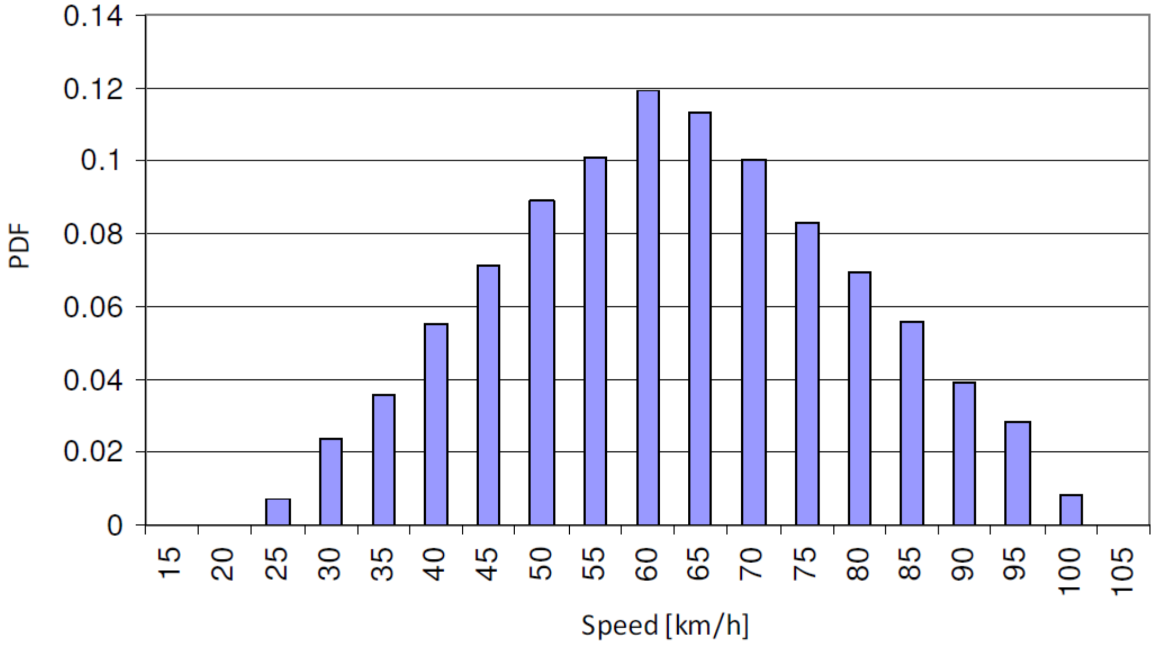


Figure A 3 Triangular PDF of the speed (from 20 up to 100 km/h) using 10 000 samples.

Annex B

Propagation Models

Propagation models adopted and used in this thesis are described in this annex.

B.1 COST-231 Walfisch Ikegami Model

For a good estimation of the received average power, one uses the well know COST 231-Walfisch-Ikegami propagation model for microcell environments [DaCo99]. This model has the following input parameters:

- h_{BS}: BS height;
- h_B: Building height;
- h_{MT}: MT height;
- w: Street width;
- f: Frequency;
- d: Distance between BS and MT;
- d_B: Building separation;
- Ψ: Street orientation angle.

The following default values are recommended:

- $d_B: [20{,}50] m$
- $w: \frac{d_B}{2}$
- $h_B: 3\, m \times [number\ of\ floors] + roof$
- $\Psi: 90°$

The path loss, when in LoS, is given by:

$$L_{p[\text{dB}]} = 42.6 + 26 \cdot \log(d_{[\text{km}]}) + 20 \cdot \log(f_{[\text{MHz}]}) \quad \text{(B.1)}$$

The path loss in the case of NLoS is given by (all path loss values are expressed in dB):

$$L_p = \begin{cases} L_0 + L_{rts} + L_{msd}, & L_{rts} + L_{msd} > 0 \\ L_0 & , L_{rts} + L_{msd} \leq 0 \end{cases} \quad \text{(B.2)}$$

where:

$$L_{0[\text{dB}]} = 32.4 + 20 \cdot \log(d_{[\text{km}]}) + 20 \cdot \log(f_{[\text{MHz}]}) \quad \text{(B.3)}$$

$$L_{rts[\text{dB}]} = -16.9 - 10 \cdot \log(w_{[\text{m}]}) + 10 \cdot \log(f_{[\text{MHz}]}) + 20 \cdot \log(\Delta h_{MT} + L_{ori[\text{dB}]}) \quad \text{(B.4)}$$

$$\Delta h_{Mobile} = h_{MT[\text{m}]} - h_{B[\text{m}]} \quad \text{(B.5)}$$

$$L_{ori[\mathrm{dB}]} = \begin{cases} -10 + 0.34 \cdot \Psi_{[°]} & ,0° \leq \Psi < 35° \\ 2.5 + 0.075 \cdot \left(\Psi_{[°]} - 35\right) & ,35° \leq \Psi < 55° \\ 4.0 + 0.144 \cdot \left(\Psi_{[°]} - 55\right) & ,55° \leq \Psi \leq 90° \end{cases} \tag{B.6}$$

$$L_{msd[\mathrm{dB}]} = L_{bsh[\mathrm{dB}]} + K_{a[\mathrm{db}]} + K_d \cdot \log\left(d_{[\mathrm{km}]}\right) + K_f \cdot \log\left(f_{[\mathrm{MHz}]}\right) - 9 \cdot \log\left(d_{B[\mathrm{m}]}\right) \tag{B.7}$$

where:

$$L_{bsh[\mathrm{dB}]} \begin{cases} -18 \cdot \log(1 + \Delta h_{Base}) & ,h_{BS[\mathrm{m}]} > h_{B[\mathrm{m}]} \\ 0 & ,h_{BS[\mathrm{m}]} \leq h_{B[\mathrm{m}]} \end{cases} \tag{B.8}$$

$$\Delta h_{Base} = h_{BS[\mathrm{m}]} - h_{B[\mathrm{m}]} \tag{B.9}$$

$$K_{a[\mathrm{db}]} = \begin{cases} 54 & ,h_{BS} > h_B \\ 54 - 0.8 \cdot \Delta h_{Base} & ,d \geq 0.5 \text{ km and } h_{BS} \leq h_B \\ 54 - 0.8 \cdot \Delta h_{Base} \cdot \dfrac{d_{[\mathrm{km}]}}{0.5} & ,d < 0.5 \text{ km and } h_{BS} \leq h_B \end{cases} \tag{B.10}$$

$$K_d = \begin{cases} 18 & ,h_{BS} > h_B \\ 18 - 15 \cdot \Delta h_{Base}/h_B & ,h_{BS} \leq h_B \end{cases} \tag{B.11}$$

$$K_f = \begin{cases} -4 + 0.7 \cdot \left(\dfrac{f_{[\mathrm{MHz}]}}{925} - 1\right) & \text{for medium size cities and suburban} \\ & \text{centres with moderate tree density} \\ -4 + 1.5 \cdot \left(\dfrac{f_{[\mathrm{MHz}]}}{925} - 1\right) & \text{for metropolitan centres} \end{cases} \tag{B.12}$$

L_0 is the free space attenuation, L_{rts} is "roof-to-street diffraction and scatter loss", L_{ori} is the attenuation caused by main street orientation with respect to the direct radio path and L_{msd} is the "multi-screen diffraction loss".

Some parameters have a validity range, presented in Table B.1.

Table B.1 Validity range of parameters.

Frequency, *f* [MHz]	[800, 2000]
Distance NLoS, *d* [km]	[0.02, 5]
Distance LoS, *d* [km]	[0.02, 0.2]
BS antenna height, h_{BS} [m]	[4, 50]
MT antenna height, h_{MT} [m]	[1, 3]

B.2 Double Breakpoint Model

The free space propagation model is not applicable to many practical situations. However, due to its simplicity, it is common to use it for estimations, in this case the distance exponent $\gamma = 2$ is changed to better match practical situations. A breakpoint model can be applied in relation to obstructed conditions, at which a distance exponent $\gamma = 2$ is used for the first metres, and a larger γ for distances above the breakpoint.

For outdoor environments with an antenna height of a few metres and a distance of a few hundread meters, a “double” breakpoint model gives a good characterisation of the path loss for urban environments in the presence of obstruction. The double breakpoint model [PrPr01] has a first breakpoint at 1m (reference distance for isotropic loss) and a second one at 100m. Note that frequencies covered by this model matches with IEEE 802.11 family, thus, this model is used for WLAN/Wi-Fi propagation estimation.

For 2.4 GHz, the path loss is as follows:

$$L_{p_{[\mathrm{dB}]}} = \begin{cases} 40 + \gamma 20 \log(d_{[\mathrm{m}]})\,, d \leq d_{break} \\ 40 + \gamma_1 20 \log\left(d_{break[\mathrm{m}]}\right) + \gamma_2 10 \log\left(\frac{d_{[\mathrm{m}]}}{d_{break[\mathrm{m}]}}\right), d > d_{break} \end{cases} \quad \text{(B.13)}$$

While for the 5 GHz band it is:

$$L_{p_{[\mathrm{dB}]}} = \begin{cases} 46.38 + \gamma 20 \log(d_{[\mathrm{m}]})\,, d \leq d_{break} \\ 46.38 + \gamma_1 20 \log\left(d_{break[\mathrm{m}]}\right) + \gamma_2 10 \log\left(\frac{d_{[\mathrm{m}]}}{d_{break[\mathrm{m}]}}\right), d > d_{break} \end{cases} \quad \text{(B.14)}$$

Where γ_1 and γ_2 are set to 2.

Annex C

Traffic Source Models

This annex presents the traffic source models used in the simulator.

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C.1 Voice Models

VoIP services present commonly a symmetric or quasi-symmetric nature and require small end-to-end transmission delays. According to [Agui03], VoIP can be characterised through a traditional ON-OFF behaviour, in which sequences of speech-bursts are intercalated with silent bursts. So, a VoIP transmission can be modelled as a Markov model with two states of "silence" and "talk": when in "silence", no packets are generated, and when in "talk", packets are generated at a constant rate. Both activity and silent periods are generated by an exponential distributed random variable with mean values t_{ON} andt_{OFF}, respectively.

The payload size of the IP packets carrying speech bursts depends on the considered speech codec and the packet rate.

As VoIP uses UDP (User Data Protocol) and RTP (Real Time Protocol) at the transport layer, the size of a full IPv6 (Internet Protocol version 6) header together with a RTP/UDP header is 60 bytes, and 40 bytes if IPv4 (Internet Protocol version 4) is used instead. As the size of a typical voice packet is 20 bytes if G729.A is used, the RTP/UDP/IP overhead figures illustrate the typical problem of the header overhead in VoIP: in this case, instead of an 8 kbps bitrate, a final bitrate of 32 kbps case IPv6 was in use would be generated (24 kbps if IPv4 is used instead). It is important to use the radio band as effectively as possible, and header overhead up to 60 bytes can severely degrade the spectral efficiency of a VoIP service over such link. Without header compression, two-thirds of the transmission would be just headers. In order to handle this purpose, protocols such as "RObust Header Compression" (ROHC) have been developed to tackle this problem [IETF01]. According to [Agui03], one can assume that header bytes can be compressed to 8 bytes. Additionally, and following ETSI recommendations [ETSI98], speech calls should be generated according to a Poisson process, with mean call duration of 120 s. The resulting VoIP modelling is summarised in Table C.1.

Table C.1 Modelling of VoIP traffic.

Activity Factor [%]	50
Mean Active Phase, t_{ON} [s]	3
Mean Silent Phase, t_{OFF} [s]	3
Payload of IP Packets [bytes]	20
IP Overhead [bytes]	8
Transmission Time Interval [ms]	20
Mean Call Duration [s]	120

C.2 Video Model

In [ChRe98], a source model for variable bitrate video traffic model based on a finite-state Markov chain is presented, being demonstrated that it accurately models a one- and two-layers video. This model assumes two types of video frames generation, the *I* and *P* frames. *I* is driven by scene changes, depends on the video source, and by itself can be considered independent of the encoder dynamics. However, in the time period characterisation by a sequence of *P* frames, one may expect that there are no significant changes in the information in successive frames. Consequently, the bit rates characterisation of successive *P* frames can be expected to be correlated, or more generally clustered around an average value. That fact allows a technique that uses correlation between successive frame bit rates to identify the *I* and *P* frames in the data. The frame rate ranges from 24 to 30 frames per second.

The *I* frame statistic is modelled by a Gaussian distribution function, which average and variance is adjusted as a function of measured data. *P* frames may be quite different, because they depend of video changes. In order to model these frames, a mechanism of $k1$ states was created, each state with its own average and variance.

The corresponding $k1$ state group to frame *P*, together with the only state that characterises frame *I*, typify the $k1+1$ Markov states that are represented by a probabilistic transitions matrix *P*. In this matrix, 90 % of the total of probabilistic transitions from one state to another are concentrated in $\{P_{ii-1}, P_{ii}, P_{ii+1}\}\forall\ I = 1, \dots k1$. Considering $i = n$ any state in the Markov chain and $j = n-1$ state. The transmission speed of a *I* frame is given by [RaMe01], [ChRe98]:

$$\begin{cases} R_i(n) = m_i(1-\alpha_i) + \alpha_i R_j(n-1) + g_i(n)\ i,j = 1,\dots,k(P - frames) \\ R_{k+1}(n) = m_k + g_{k+1}(n) \qquad\qquad\qquad (I - frames) \end{cases} \tag{C.1}$$

where:

- $m_i$is the estimated average value of transmission speed (*i* state)
- α_i is a coefficient value that models the autoregressive process, which value is adjusted empirically (based on measurements over *i* state in P frame)
- $g_i(n)$ is a Gaussian random variable, different for each state (average 0 and variance given by measurement data in *i* state)

C.3 Non-Conversational Applications

Non-real time applications typically present an asymmetrical nature, as they refer mostly to specific requests for information done by end users to remote machines. The most known applications are Web browsing, FTP and E-mail.

Numerous models are studied and proposed to characterise Web browsing but as the present work does not intend to focus specifically on traffic models, modelling of these applications is based on [ETSI98]. Figure A-1 illustrates a typical Internet surfing session, which consists of a sequence of packet calls. During a packet call several packets may be generated, which means that the packet call is composed of a bursty sequence of packets, Figure C.1.

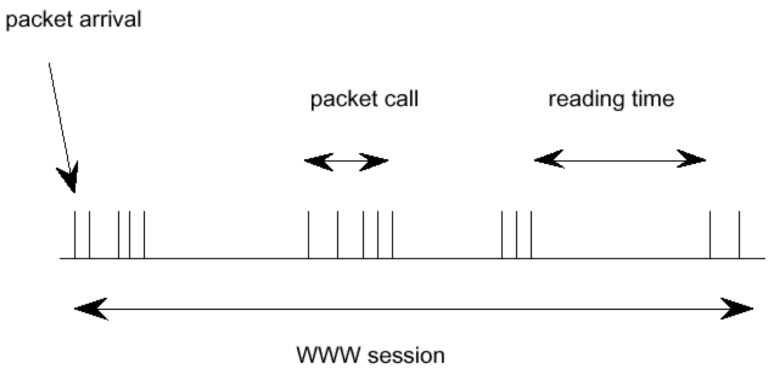

Figure C.1 Typical WWW session (adapted from [ETSI98]).

A packet service session is modelled as a Poisson process. It typically contains one or several packet calls, depending on the application. In a WWW browsing session for instance, a packet call corresponds to the downloading of a web page, and after the document has entirely arrived to the terminal, the user takes some time for analysing the information, which is often called the reading time. On the other hand, in a FTP session, it is likely that the session contains only one packet call.

In order to fully characterise a packet session, the following parameters must be modeled [ETSI98]:

- **The number of packet call requests per session**, N_{pc}: This is a geometrically distributed random variable with mean μN_{pc}.
- **The reading time between two consecutive packet call requests in a session**, D_{pc}: This is geometrically distributed random variable with a mean μD_{pc}. Note that the reading time starts when the last packet of the packet call is completely received by the user. The reading time ends when the user makes a request for the next packet call.
- **The number of packets within a packet call**, N_d: Although different statistical distributions can be used to generate the number of packets, it is assumed that $N_d$can be a geometrically distributed random variable, with mean μN_d.
- **The time interval between two consecutive packets inside a packed call**, D_d: This is a geometrically distributed random variable with a mean μD_d. Naturally, if there is only one packet in a packet call, this is not needed.
- **The Packet size**, Ps:The packet size distribution model is based on Pareto distribution that suits best for the traffic case under study; Pareto distribution with cut-off is used.

C.4 Streaming Model

Audio and Video Streaming services commonly present an asymmetrical nature, as they refer mostly to the download of large files. Contrary to VoIP, Streaming is more flexible to end-to-end transmission delays and to its variations, although the implementation of buffering is required to accommodate those fluctuations. Although Video traffic can be transported either with a constant bitrate or with a variable bitrate, variable bitrate is the one expected to be the prime example, as it has several potential advantages over constant bitrate, particularly the possibility for implementing statistical multiplexing, allowing for improved channel allocation. IEEE has not standardised a specific codec for Video Streaming yet. On the contrary, 3GPP has specified the use of both MPEG-4 and H.263 codecs for Video Streaming services [3GPP02].

For simplification reasons, the present work uses of the non-conversational applications model presented in C.3 also for Streaming service, although with different parameters, namely the mean values μN_{pc}, μN_d and packet size (parameters of Pareto distribution).

Annex D

Link Budget

Path loss calculation and its influence in the link budget is done in this annex.

In order to calculate the maximum path loss in a cell, which is useful to estimate the sector antenna's range, the link budget has to be taken into account.

According to [Corr13], the power received by the UE is given by the following expression:

$$P_{r[\mathrm{dBm}]} = P_{EIRP[\mathrm{dBm}]} + G_{r[\mathrm{dBi}]} - L_{p,total[\mathrm{dB}]} \tag{D.1}$$

where:

- $P_{r[\mathrm{dBm}]}$: power available at the receiving antenna;
- $P_{EIRP[\mathrm{dBm}]}$: Effective Isotropic Radiated Power (EIRP);
- $G_{r[\mathrm{dBi}]}$ gain of the receiving antenna;
- $L_{p,total[\mathrm{dB}]}$: path loss.

The EIRP depends on the link. In DL, it is defined as follows:

$$P_{EIRP[\mathrm{dBm}]} = P_{Tx[\mathrm{dBm}]} - L_{c[\mathrm{dB}]} + \mathrm{G}_{t[\mathrm{dBi}]} \tag{D.2}$$

where:

- $P_{Tx[\mathrm{dBm}]}$: transmitter output power;
- $L_{c[\mathrm{dB}]}$: losses in the cable between the transmitter and the antenna;
- $\mathrm{G}_{t[\mathrm{dBi}]}$: gain of the transmitting antenna.

In UL, the EIRP is defined by the following expression:

$$EIRP_{[\mathrm{dBm}]} = P_{Tx[\mathrm{dBm}]} - L_{u[\mathrm{dB}]} + G_{[\mathrm{dBi}]} \tag{D.3}$$

where:

- $L_{u[\mathrm{dB}]}$: losses due to the user, which take a value between 0 and 3 dB for data.

The power at the receiver, in DL, is given by:

$$P_{Rx[\mathrm{dBm}]} = P_{r[\mathrm{dBm}]} - L_{u[\mathrm{dB}]} \tag{D.4}$$

where:

- $P_{Rx[\mathrm{dBm}]}$: power at the input of the receiver.

The power at the receiver, in UL, is given by:

$$P_{Rx[\mathrm{dBm}]} = P_{r[\mathrm{dBm}]} - L_{c[\mathrm{dB}]} \tag{D.5}$$

The average noise power can be estimated from the following expression, according to [Corr13] and [SeTB11]:

$$N_{[\mathrm{dBm}]} = 10\log\left(\frac{k[\mathrm{JK}^{-1}] \times T_{[\mathrm{K}]}}{10^{-3}}\right) + 10\log(B_{N\,[\mathrm{Hz}]}) + F_{[\mathrm{dB}]} \quad \text{(D.6)}$$

where:

- k: Boltzmann's constant
- T: temperature of the receiver (assumed to be, hence, 288.2 K);
- B_N: noise bandwidth;
- F: noise figure.

The noise bandwidth is given by:

$$B_{N\,[\mathrm{Hz}]} = N_{RB} \times B_{RB[\mathrm{Hz}]} \quad \text{(D.7)}$$

where:

- N_{RB}: number of RBs;
- B_{RB}: bandwidth of one RB, which is 18kHz.

The total path loss can be calculated by:

$$L_{p,total[\mathrm{dB}]} = L_{p[\mathrm{dB}]} + M_{SF[\mathrm{dB}]} + M_{FF[\mathrm{dB}]} \quad \text{(D.8)}$$

where:

- L_p: path loss from the COST-231 Walfisch-Ikegami model;
- M_{SF}: slow fading margin;
- M_{FF}: fast fading margin.

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