Performance evaluation of VoIP and web services in HSDPA

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Gratitude

First of all I would like to thank my partner Marc, for his love and support. Also I would like to mention the important support and affection I have received from my family.

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1 Introduction 9
2 HSDPA Principles 11
2.1 Introduction to mobile communications 11
2.2 High Speed Downlink Packet Access 13
2.3 HSDPA standardisation in 3GPP 14
2.4 HSDPA versus Release 99 15
2.5 3G/HSDPA Network Architecture 16
2.5.1 User Equipment 17
2.5.2 UMTS Terrestrial Radio Access Network 17
2.5.3 Core Network 18
2.6 HSDPA Operation Principle 19
2.7 HSDPA Channels 21
2.7.1 New Transport Channel, HS-DSCH 21
2.7.2 High-Speed Shared Control Channel, HS-SCCH 25
2.7.3 High-Speed Dedicated Physical Control Channel, HS-DPCCH 25
3 Design of the Simulator 26
3.1 Introduction 26
3.1.1 Voice over IP Traffic 26
3.1.2 Web Traffic 27
3.2 Inputs 27
3.2.1 Creation of the CQI tables 28
3.2.2 Queue_matrix 30
3.2.3 Inputs to model the web traffic 31
3.3 The simulator 34
3.3.1 Cell Creation 36
3.3.2 Path Loss Calculation 37
3.3.3 Initialization of the Variables 37
3.3.4 RSCP calculation 39
3.3.5 Actualization of transmission queues and looking for users that want to transmit 41
3.3.6 Scheduler 42
3.3.7 EcIo Calculation 43
3.3.8 CQI calculation 43
3.3.9 Maximum number of codes and transport block size 44
3.3.10 Probability of retransmission 44
3.3.11 Transmission 45
3.3.12 Retransmission 45
3.4 Outputs 46
4 Simulation Results 47
4.1 Only VoIP 47
4.2 Only Web 50
4.3 Mixed Traffic 54
5 Conclusions 61
5.1 Project conclusions 61
5.2 Future work lines 62
6 Appendices 63
6.1 CQI Tables 63
6.2 Probability Distributions 68
6.2.1 Exponential distribution 68

6.2.2 Geometric distribution 68
6.2.3 Pareto Distribution 69
6.3 CQI Distributions 70
6.4 Percentage of users above threshold for VoIP simulations 81
6.5 Percentage of users above threshold for WEB simulations 83
6.6 Percentage of users above threshold for mixed simulations 85
6.6.1 VoIP Users 85
6.6.2 Web Users 88
6.7 Bibliography 90

Abbreviations

1G- First Generation
2G- Second Generation
3G- Third Generation
3GPP- Third Generation Partnership Project
AMC- Adaptive Modulation & Coding
CDMA- Code Division Multiple Access
CN- Core Network
CQI- Channel Quality Indicator
CRC- Cyclic Redundancy Check
CS- Circuit Switch
DCH- Dedicated Channel
DSCH- Downlink Shared Channel
EDGE- Enhanced Data rates for GSM Evolution
FACH- Forward Access Channel
GGSN- Gateway GPRS Support Node
GMSC- Gateway MSC
GPRS- General Packet Radio Services
GSM- Global System for Mobile Communications
H-ARQ- Hybrid Automatic Repeat Request
HO- Handover
HSDPA- High Speed Downlink Packet Access
HS-DPCCH- High speed Dedicated Physical Control Channel
HS-DSCH- High Speed Downlink Shared Channel
HS-PDSCH- High Speed Physical Downlink Shared Channel
HS-SCCH- High Speed Shared Control Channel
HSUPA- High Speed Uplink Packet Access
IP- Internet Protocol
LTE- Tong Term Evolution
ME- Mobile Equipment
MSC- Mobile Switching Centre
PS- Packet Switch
PTT- Post Telegraph & Telephone
QAM- Quadrature Amplitude Modulation
QPSK- Quadrature Phase Shift Key
RNC- Radio Network Controller
RNS- Radio Network Sub-system
RSCP- Received Signal Code Power
SF- Spreading Factor
SGSN- Serving GPRS Support Node
TBS- Transport Block Size
TFCI- Transport Format Combination Indicator
TTI- Transmit Time Interval
UE- User Equipment
UMTS- Universal Mobile Telecommunications System
USIM- UMTS Subscriber Identity

UTRAN- UMTS Terrestrial Radio Access Network
VoIP- Voice over IP
WBTS- WCDMA Base Transceiver Station
WCDMA- Wide-Band Code Division Multiple Access

1 Introduction

During the last years, the mobile communications market in the western developed countries has reached a standstill, with a market penetration higher than 100% in most countries such as, Spain 115%, United Kingdom 120% and Italy with more than 150%.

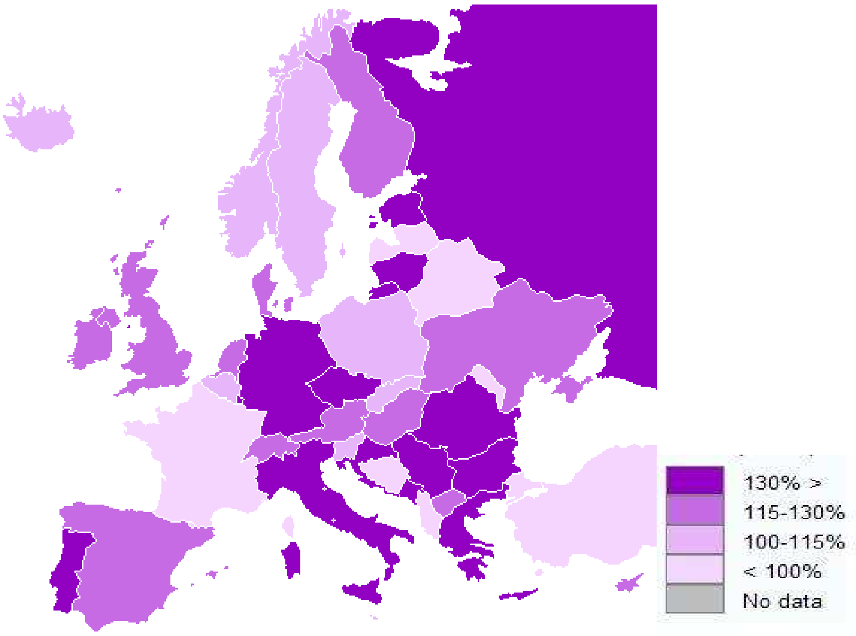


Figure 1-1. Mobile communications market penetration [1]

So the network operators have focused their strategy in moving into a new market still developing, the data, that could help the operators to increase their saturated income. The solution mostly adopted by the operators is to offer the mobile broadband offering flat rate schemes to the customers for a monthly fee. The mobile broadband appeals to wide range of users, such as corporate users and especially to students which give them the liberty to access internet across different locations such as university, cafe or shared accommodation.

That creates a set of challenges to the operators having to improve their network in order to cope with this huge new demand for high-speed data, to successfully satisfy these requirements, Third Generation networks must support high user data rates, especially on the downlink direction of the communication path due to its heavier load. For these reason, the 3GPP standardized in Release 5 the technology HSDPA (High Speed Downlink Packet Access). The HSDPA provides a cost effective solution to provide high-speed data to the customers specially focused to increase the overall cell capacity thanks to the fact that the resources are shared among the users.

This project has consisted on the design of a HSDPA simulator and on the evaluation of the performance of VoIP and web browsing traffic in HSDPA.

The idea of this project started on an industrial placement in Orange UK, in the department of Access Network. Between other tasks the CQI (Channel Quality Information) were modelled using samples taken from the live network.

In order to do a more theoretical analysis and make use of the CQI modelling, after the placement, the design of the complete simulator has been made under the direction of Professor Ferran Casadevall.

The objective of this project is then to simulate an HSDPA cell in different conditions, specially focused in different cell loading conditions such as:

- Different number of users, up to 100 users per cell.
- Different traffic profiles, choosing between VoIP and web users.

Once the simulations have been carried out, the results have been analysed in detail, offering figures and facts of how the throughput and the traffic delay changes with different load conditions. As both the VoIP and web traffic have different thresholds of maximum delay defined by international entities, the simulations could be used to asses to define, the maximum number of HSPDA users in cell. This will help to capacity planners to decide the rollout strategy, based on the simulation results.

This project has been developed with the tool Matlab. This tool has been chosen because it allows an effective code development and at the same time it is very useful to produce graphics and to compute difficult numerical calculations.

This document has been divided in five chapters; the structure of the project is:

- Chapter 1 Introduction: Where the objective and contents of the project are exposed.

- Chapter 2 HSDPA Principles: Is a theoretical chapter which explains the main concepts of HSDPA.

- Chapter 3 Design of the Simulator: In this chapter the design and main functions of the simulator are explain.

- Chapter 4 Simulation Results: Where the results of different simulations are explained.

- Chapter 5 Conclusions: This chapter defines the project conclusions and possible future lines of work.

2 HSDPA Principles

This chapter covers High-Speed Downlink Packet Access (HSDPA) principles for Wide-band Code Division Multiple Access (WCDMA).

The HSDPA technology appeared for the first time in the 3GPP Release 5 specification and its main improvements are the following:

- Increased data rates up to 14Mbps (future releases will allow higher speeds).
- Shorter service response times, due to shorter TTI.
- Adaptive modulation scheme based on received radio conditions.

2.1 Introduction to mobile communications

In Europe, during the 1980s a number of commercial analogue mobile networks were in operation, these are known as First Generation (1G). They had a very limited capacity and did not support roaming as they were based on different standards.

Due to the projected growth in mobile telecommunications, the European Post Telegraph and Telephone (PTT) departments decided to develop a system using digital technology. This system is known as Global System for Mobile communications (GSM) and it is a Second Generation (2G) system. International roaming and security are important features of GSM, along with increased system capacity thanks to the TDMA/FDMA technology and voice quality improvements.

As stated in [2], nowadays GSM is a well established and still a relevant technology used in 219 countries and with more than 3 billion subscribers around the world.

In a stand-alone GSM network, only Circuit Switched data services are offered. Modern data services, such as IP based services, are packet based and are poorly and inefficiently served by this circuit based arrangement.

For this reason the standard General Packet Radio Services (GPRS) was developed. GPRS was the first network design to offer Packet Switched services, working alongside a GSM network. GPRS enables higher data rates than GSM with the added benefit that resources are only used when packets are actually being sent. GPRS is often referred to as 2.5G.

EDGE (Enhanced Data rates for GSM Evolution) also known as Enhanced GPRS is an evolution of the standard GPRS that allows improved data transmission rates using higher modulation schemes, working as an improvement of the existing GPRS network.

Third generation systems (3G) are the next step in cellular mobile communications systems. The main purpose for the creation of the 3G network was to provide in top of voice and messaging services, higher bitrates to allow multimedia services such as video conferencing or high-speed internet access.

The 3G networks are based on the CDMA technology, achieving a higher spectral efficiency. For that fact a new Radio Access network had to be defined, named UTRAN (UMTS Terrestrial Radio Access Network). In addition the 3G network are also able to provide different services at the same time such as voice and Internet usage.

The HSDPA (High Speed Downlink Packet Access) is a 3G evolution which enables even higher data rates using higher modulations schemes and using a shared channel instead of the dedicated channels used in 3G.

In the near future LTE (Long Term Evolution) networks could be a fierce rival to 3G networks, as those networks will provide several improvements. However the LTE networks are not yet fully commercially available, so its degree of implementation it is still unknown.

The figure 2-1 shows the evolution of the different mobile communications technologies.

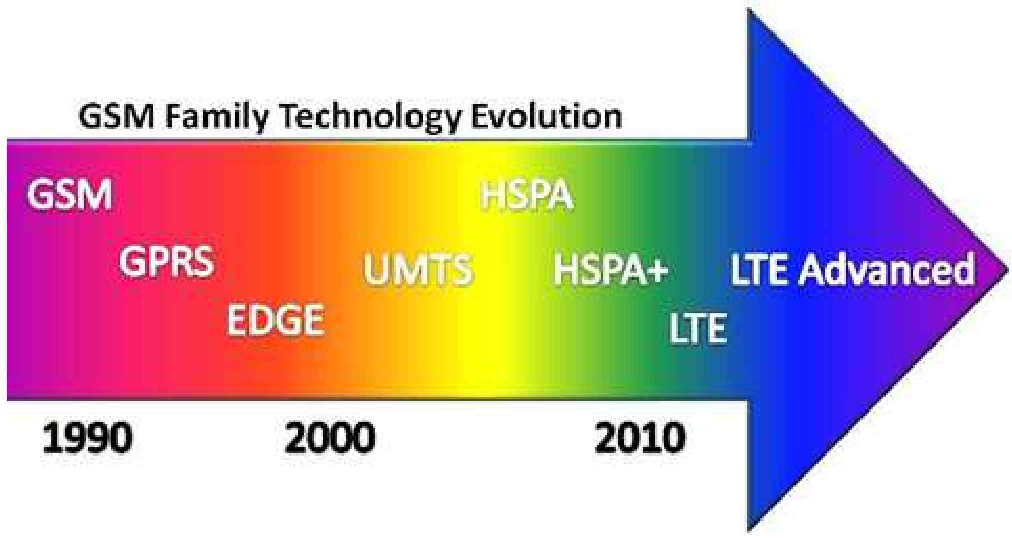


Figure 2-1. Mobile communications evolution

2.2 High Speed Downlink Packet Access

HSDPA is the evolution of the third generation (3G) technologies and it is considered the previous step before the fourth generation (4G), the future all IP, high-speed mobile networks.

Both HSDPA and HSUPA (High-Speed Uplink Packet Access) are the components of the HSPA (High-Speed Packet Access) family.

The purpose of HSDPA is to increase the downlink data rates (current HSDPA deployments support downlink speeds of 1.8, 3.6, 7.2, 10.8 and 14.4 Mbps), improve the quality of service and enhance the spectral efficiency which allows more users to be able to use high data rates on a single carrier.

The fundamental techniques used in HSDPA to achieve these improvements are AMC (Adaptive Modulation and Coding), extensive multi-code operation and a fast and spectrally efficient retransmission strategy.

The TTI (Transmission Time Interval) has been reduced from 10ms to 2ms, so the packet services latency has been reduced substantially and working along side with the adaptive modulation feature is able to react better to the changing nature of the radio conditions caused by the fading.

HSDPA is able to satisfy the most demanding multimedia applications such as high speed email attachments, voice over IP, online gaming or web page browsing. Even though speeds achieved by HSDPA could reach up to 14.4 Mbps, nowadays most networks operators have only deployed commercially 7.2 Mbps solutions.

The figure 2-2 shows a map with the current HSDPA coverage.

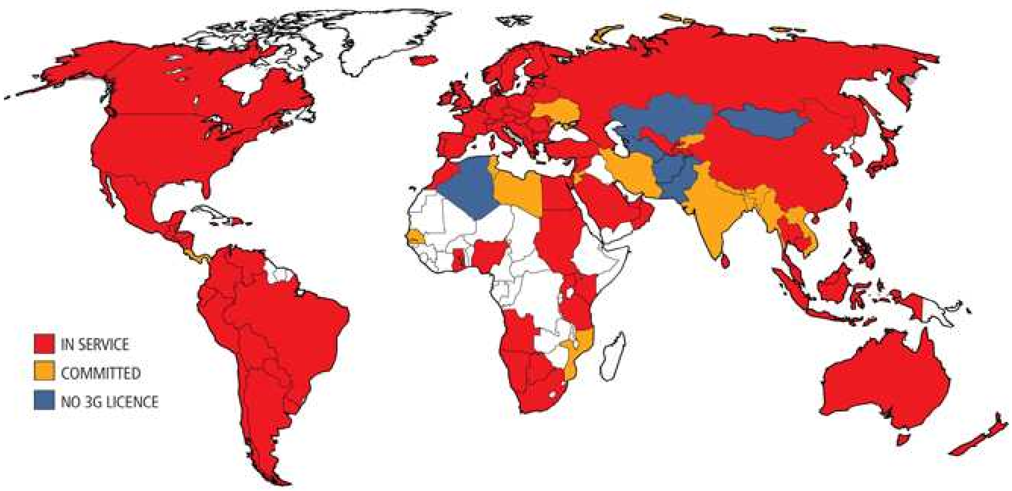


Figure 2-2. HSDPA coverage [2]

2.3 HSDPA standardisation in 3GPP

HSDPA was standardized as part of the 3GPP Release 5, being the first specification version in March 2002. The first commercial networks were available at the end of 2005 [3].

Many enhancements have been introduced in the following releases (R6, R7 and R8), figure 2-3 shows the evolution of the downlink peak data rates:

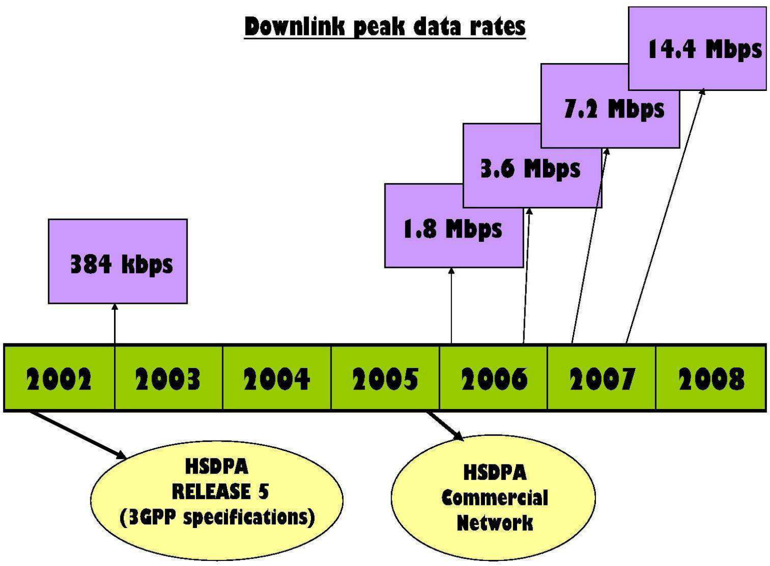


Figure 2-3. HSDPA peak data rates

HSDPA is deployed on top of the WCDMA R99 networks and both of them share all the network elements in the Core Network and in the Access Network. However significant changes in the both RNC and Node B software are required, in addition to that the Iub capacity will need to be significantly increased to cope with the increased bit rates provided by HSDPA.

2.4 HSDPA versus Release 99

In R99, the three different channels that can be used for downlink packet data are:

- Dedicated Channel (DCH).
- Downlink-Shared Channel (DSCH).
- Forward Access Channel (FACH).

The DCH can be used for any type of service and it has a fixed SF (Spreading Factor) in the downlink. This means that it reserves the code space capacity according to the peak data rate of the connection.

The DSCH always operates together with the DCH. It has dynamically varying SF, which is informed on a 10 ms basis with the TFCI (Transport Format Combination Indicator) signalling carried in the associated DCH. The DSCH channel can be shared by several users. This channel though, has hardly been implemented in most of the commercially available solutions in the R99 variant.

The FACH can be used for low speed downlink packet data as well although usually is used to carry control information.

In HSDPA, the transport channel used to carry user data is the HS-DSCH (High-Speed Downlink Shared Channel). A comparison of the fundamental properties of the R99 DCH and the HSDPA HS-DSCH can be seen in figure 2-4:

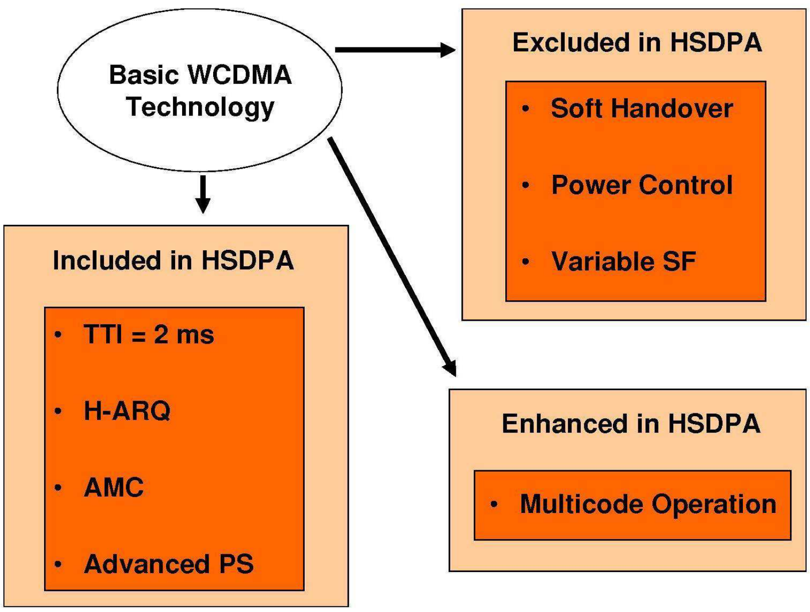


Figure 2-4. Features included and excluded in the HSDPA HS-DSCH

As the figure 2-4 shows, in HSDPA, two of the main characteristics of WCDMA have been excluded: Closed loop power control, variable spreading factor and specially Soft Handover. On the other hand, some features have been included: AMC (Adaptive Modulation and Coding) and shorter TTI.

2.5 3G/HSDPA Network Architecture

HSDPA is deployed on top of the WCDMA network either on the same carrier or, for high-capacity and high bit rate solution, using another dedicated carrier. But in both cases HSDPA and R99 share all the network elements. Because of the shared infrastructure, the cost of upgrading from R99 to HSDPA is relatively low in comparison with building a new standalone data network.

The UMTS (Universal Mobile Telecommunications System) is the most used third-generation (3G) mobile phone technology. The initial release set by the 3GPP, R99 defined the first UMTS 3G network.

The UMTS network can be divided in three main parts:

- UTRAN (UMTS Terrestrial Radio Access Network): that handles all the radio related issues.
- CN (Core Network): is responsible for switching and routing calls and data connections to external networks.
- UE (User Equipment): interfaces with the user and the radio interface.

The figure 2-5 [4] shows this architecture:

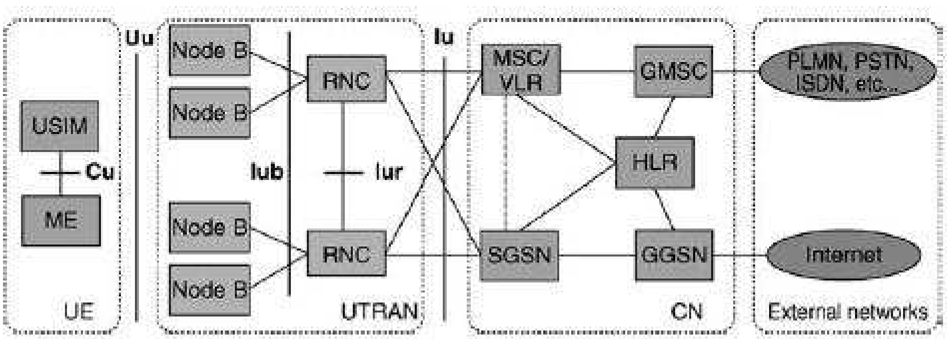


Figure 2-5. UMTS Network

2.5.1 User Equipment

The UE is what is commonly called phone; this part enables the user to access to the mobile telephony network. The UE is divided into two main parts:

- ME (Mobile Equipment): is the terminal used for the radio communications over the air interface.

- USIM (UMTS Subscriber Identity): is a smartcard which is inserted in a 3G phone. This card holds the subscriber identity, performs the authentication algorithms and stores authentication and encryption keys and some subscription information that is needed at the terminal.

2.5.2 UMTS Terrestrial Radio Access Network

The UMTS Terrestrial Radio Access Network (UTRAN) is the access network for the UMTS technology and its main function is to manage the radio resources and the mobility procedures.

The UTRAN is composed by one or more RNS (Radio Network Sub-systems) that are connected to the CN using the Iu interface, each RNS is responsible for the transmission and reception over a set of UMTS cells. The UTRAN architecture and network elements are shown in figure 2-6.

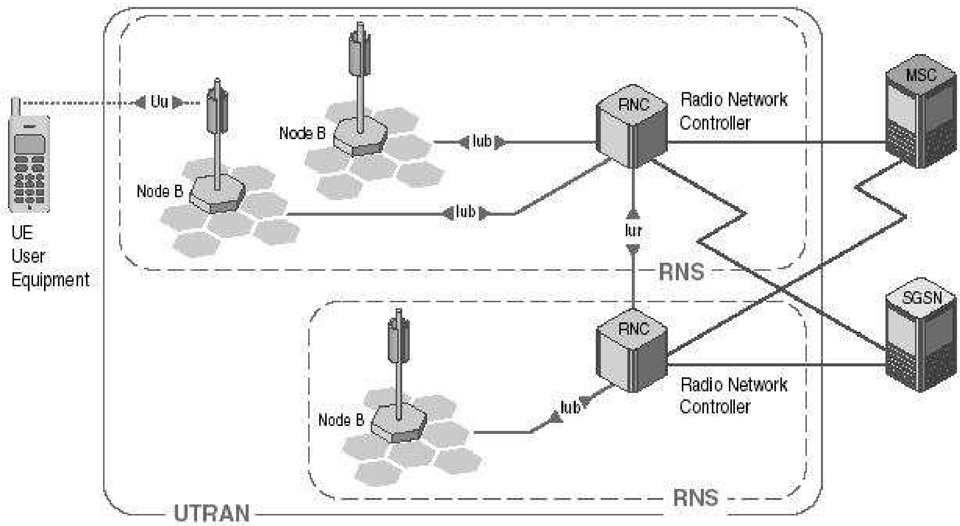


Figure 2-6. UTRAN architecture [5]

The RNS is a sub-network within UTRAN and consists of one Radio Network Controller (RNC) and one or more Nodes Bs.

The RNC (Radio Network Controller) is the network element responsible for the radio resource management and mobility management for all the cells that are in the Nodes Bs connected to the RNC. Its functionality is very similar to the functionality of a BSC in a GSM network.

These are some of the most important operations done by the RNC.

- Outer Loop Power.
- Load control.
- Admission Control.
- Packet scheduling.
- Handover control.
- Signal combining in Inter-Node B Handovers.
- Security functions.

The main function of the Node B is to provide the air interface physical layer processing, doing several operations such as: channel coding and interleaving, rate adaptation and spreading. It also performs some other basic operations such as the inner loop power control and signal combination in Softer HO situations (Handover between two cells from the same Node B).

With HSDPA the retransmission of packets is also handled on the Node B for lower response times.

2.5.3 Core Network

The Core Network (CN) is the switching part of the UMTS network. It provides call control and performs mobility and high-level security functions such as location updating and authentication. The CN also has some functionality related to the charging and billing of the services provided by the network operator.

The figure 2-7 shows the CN architecture:

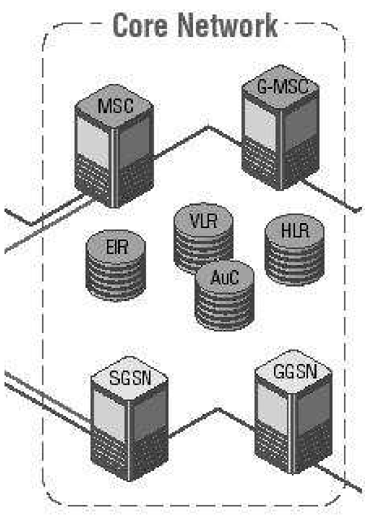


Figure 2-7. Core Network

The CN is divided into two domains depending on whether circuit switched (CS) or packet switched (PS) traffic is considered:

- CS Domain: It provides circuit-switched connections such as, the telephony service or the video telephony service. Its main elements are the MSC (Mobile Switching Centre) and the GMSC (Gateway MSC).

  The MSC is a sophisticated telephone exchange which provides circuit-switched calling, mobility management and services to the mobile phones roaming within the area that it serves. This means voice, data and fax services, as well as SMS and call divert. The MSC also supports the authentication and charging functionality.

  The GMSC is the switching point where UMTS network is connected to others CS networks, such as the public telephony network.

- PS Domain: it provides connections for data packet services that could be for example Internet Browsing, Email synchronization or multimedia downloads. Its main elements are the SGSN (Serving GPRS Support Node) and the GGSN (Gateway GPRS Support Node).

  The SGSN is responsible for the delivery of data packets from and to the mobile stations within its geographical service area. Its tasks include packet routing and transfer, mobility management (attach/detach and location management), logical link management, and authentication and charging functions.

  The GGSN is the network node that acts as a gateway between a GPRS wireless data network and other networks such as the Internet or other private networks.

2.6 HSDPA Operation Principle

HSDPA is based on fast Node B scheduling where the Node B estimates the channel quality of each active HSDPA user on the basis of the physical layer feedback received in the uplink from the UE. Scheduling and link adaptation are then conducted on a fast pace depending on the scheduling algorithm and the user prioritization scheme. The general HSDPA operation principle is show in figure 2-8.

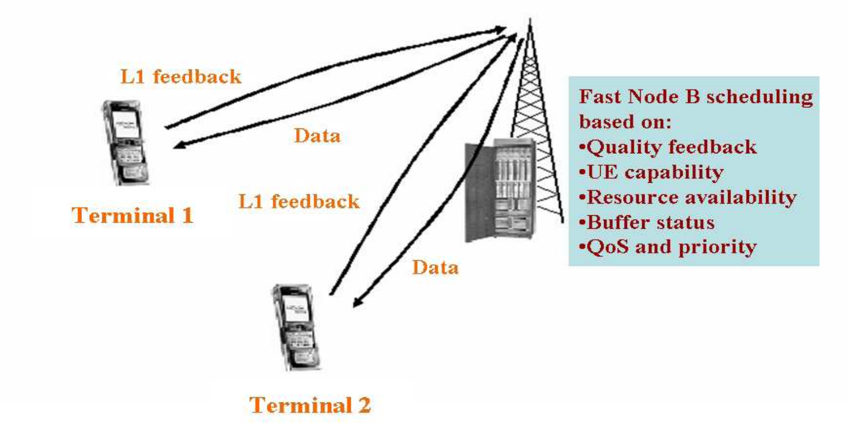


Figure 2-8. HSDPA Operation Principle

The other new key technology is the physical layer retransmissions. In R99, when the data is not received correctly it is necessary to retransmit from the RNC using RLC retransmissions. With HSDPA the packet is first received in the Node B buffer and is kept there even after it has been send to the user so in case of failure, retransmission automatically takes place from the Node B without any involvement from the RNC, that fact increases the response time of the network in case of packet retransmission. The figure 2-9 shows the difference between R99 and HSDPA retransmissions.

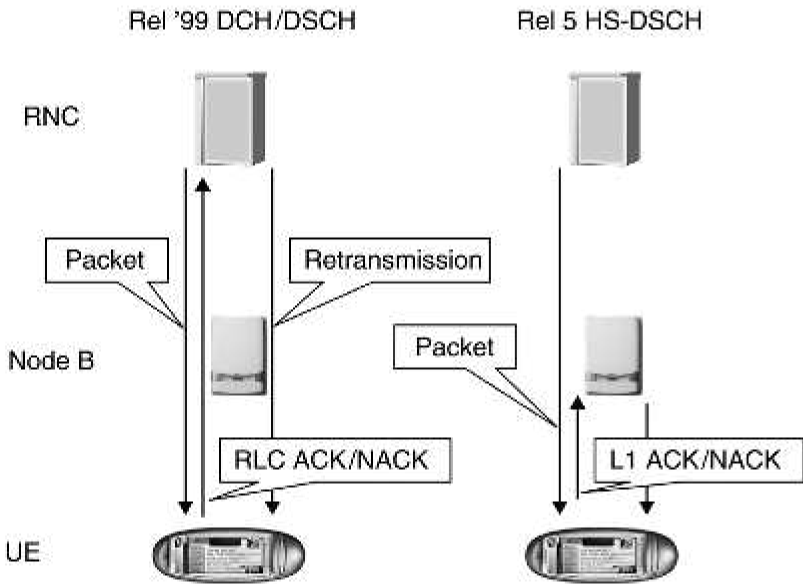


Figure 2-9. R99 and HSDPA Retransmissions

2.7 HSDPA Channels

Several new channels have been introduced for HSDPA operations. Those are shown on figure 2-10:

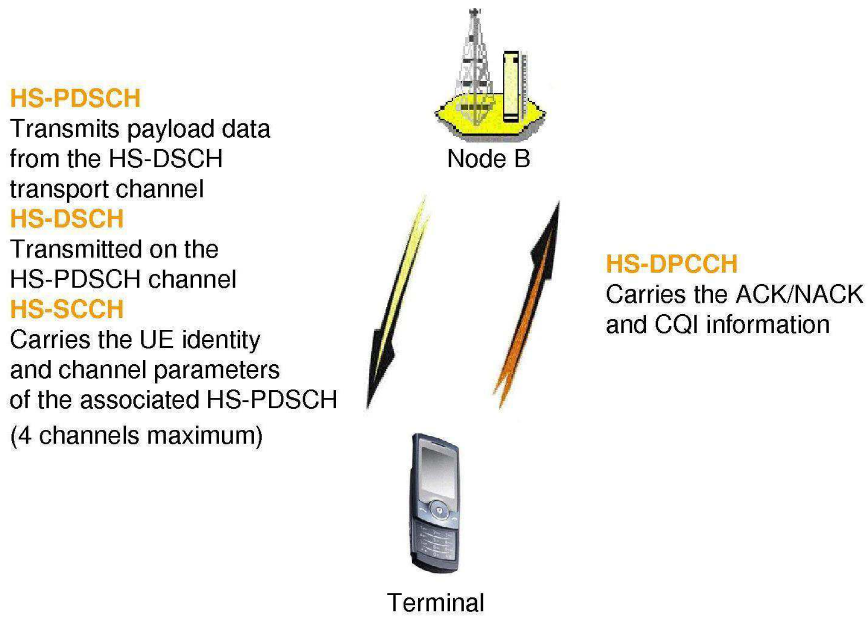


Figure 2-10. HSDPA Channels

2.7.1 New Transport Channel, HS-DSCH

The HS-DSCH (High-Speed Downlink Shared Channel) is a downlink transport channel which carries the actual user data. In the physical layer, the HS-DSCH is mapped onto the HS-PDSCH (High-Speed Physical Downlink Shared Channel).

The TTI (Transmission Time Interval) has been reduced to 2ms in order to achieve a shorter Round Trip delay.

The SF (Spreading Factor) is fixed to 16, the users can use more than one code at the same time. In addition to that several users can use different codes during the same TTI, so the resources could be shared among the cell users.

There are 15 codes available, and these are allocated to different users for each TTI. The number of codes allocated to each user is determined by the UE (User Equipment) category and by the quality of the channel indicated with the CQI (Channel Quality Indicator). This process is shown in the figure 2-11:

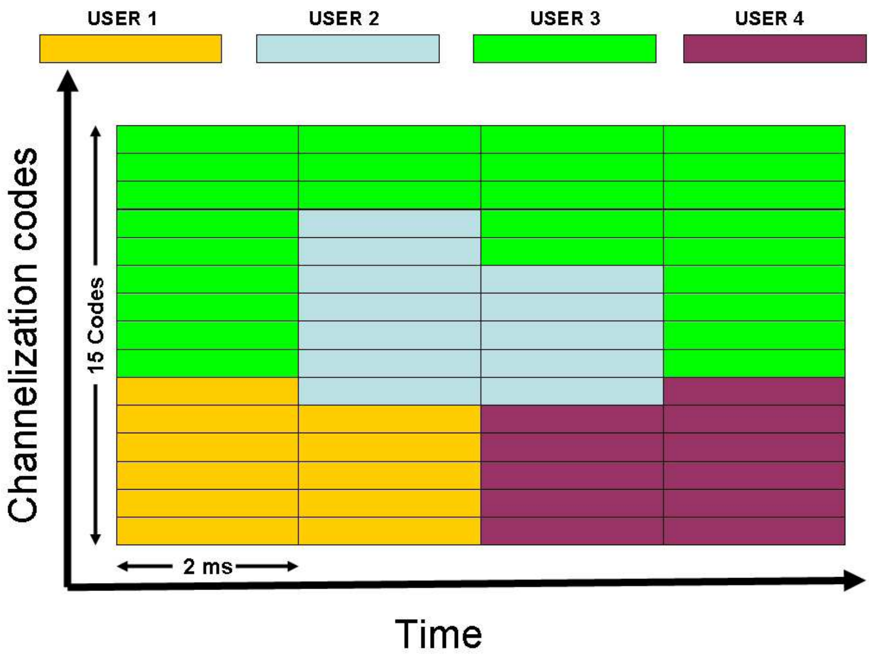


Figure 2-11. HS-DSCH Code and Time shared

2.7.1.1 Adaptive Modulation and Coding

HSDPA adapts the modulation, the coding rate and the number of channelization codes to the instantaneous perceived radio conditions. The combination of the first two mechanisms is called AMC (Adaptive Modulation and Coding).

The modulation 16 QAM (Quadrature Amplitude Modulation) is introduced in HSDPA in addition to the R99 QPSK (Quadrature Phase Shift Key). HS-DSCH incorporates this new modulation for users that have good radio conditions. Figure 2-12 shows both modulations:

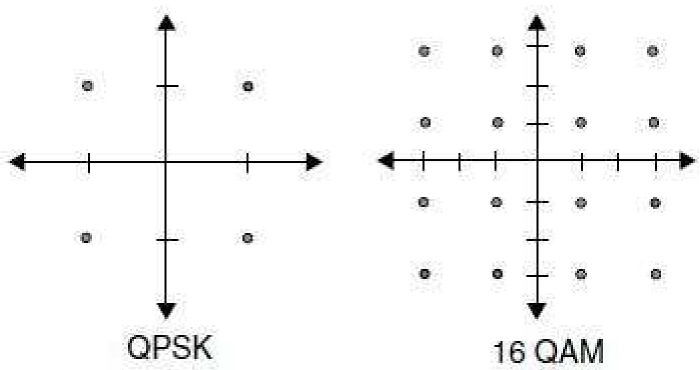


Figure 2-12. QPSK and 16QAM modulations

2.7.1.2 HS-DSCH Link Adaptation

In addition to the scheduling decision, the Node B will also decide every 2 ms which coding and modulation combination to transmit. Link adaptation is based CQIs (Channel Quality Indicator) being provided by the terminal. For the purpose of HS-DSCH link adaptation, the user therefore sends every 2ms a CQI to the serving HS-DSCH cell on the uplink HS-DPCCH (High-Speed Dedicated Physical Control Channel). The figure 2-13 shows this procedure:

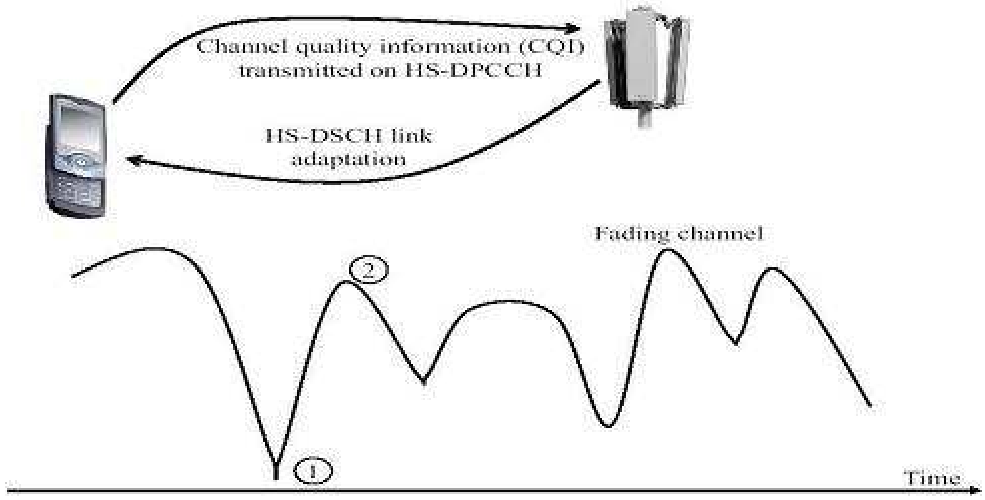


Figure 2-13. Link Adaptation

The CQI is a number between 0 and 30 that indicates the channel quality. The higher the number, the better the radio conditions will be. When a UE reports a particular CQI, it is reporting that under the current radio conditions, the UE is able to receive at the correspondent maximum data rate with a BLER lower than 0.1. The Maximum data rates associated to each CQI are calculated with the following formula:

$$Maximum\_data\_rate = \frac{TBS}{TTI}$$

The relation between the TBS (Transport Block Size) and the CQI depends on the mobile category; this relation is specified in the 3GPP specifications [6] and can be found on the tables in the Appendix 6.1.

2.7.1.3 Hybrid Automatic Repeat Request

HSDPA incorporates a physical layer retransmission functionality that adds robustness against errors and improves the performance.

The H-ARQ (Hybrid Automatic Repeat Request) functionality is operated in two different ways:

- Identical Retransmissions or Soft Combining: every retransmission is a replica of the first transmission. The decoder at the receiver combines these multiple replicas of the transmitted packet.

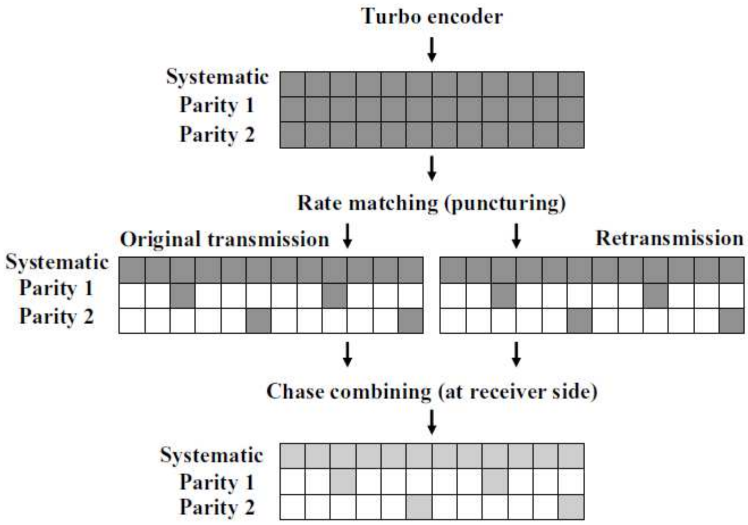


Figure 2-14. H-ARQ Identical Retransmissions

- Non-Identical Retransmissions or Incremental Redundancy: Uses a different rate matching between retransmissions. The retransmissions include additional redundant information.

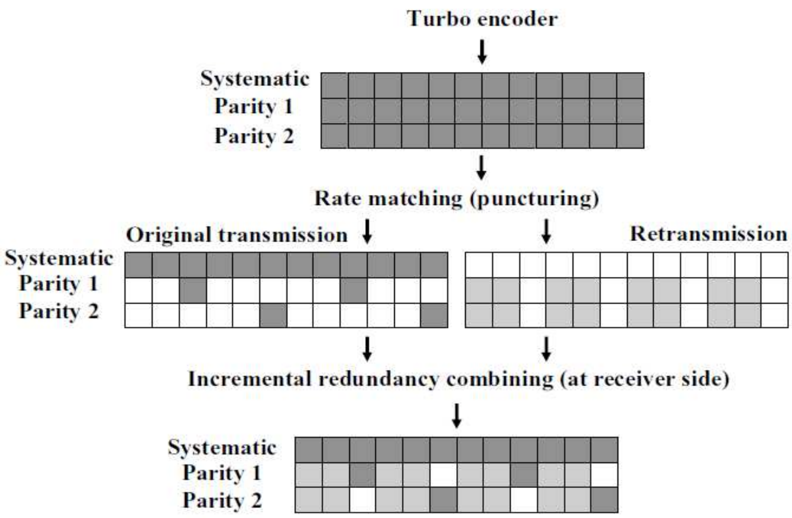


Figure 2-15. H-ARQ Non-Identical Retransmissions

2.7.2 High-Speed Shared Control Channel, HS-SCCH

The HS-SCCH carries the key information necessary for the HS-DSCH demodulation. The number of HS-SCCH channels is determined by the maximum number of users that will be code-multiplexed in the same TTI.

Each HS-SCCH block is divided in three slots. The first slot contains time-critical information, which is needed to start the demodulation. The next two slots contain less time critical parameters, such as the CRC (Cyclic Redundancy Check) and H-ARQ information.

The HS-SCCH uses a Spreading Factor of 128 which allows 40 bits per slot to be carried.

It carries the following information:

- Codes to despread.
- Modulation (QPSK or 16QAM).
- H-ARQ related information.

2.7.3 High-Speed Dedicated Physical Control Channel, HS-DPCCH

The HS-DPCCH is an uplink channel that carries the ACK/NACK information for the physical layer retransmissions and the CQI (Channel Quality Indicator) to indicate which TBS (Transport Block Size), modulation type and number of codes could be received correctly, that is with a BLER smaller than 0.1.

The HS-DPCCH uses a fixed spreading factor of 256 and has a 2ms/three-slot structure. The first slot is used for the HARQ information and the two remaining slots are for CQI use. This structure is shown in figure 2-16:

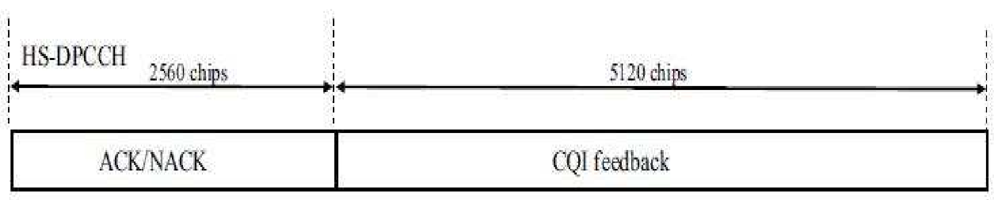


Figure 2-16. HS-DPCCH structure

3 Design of the Simulator

This chapter explains the design of the Simulator, it describes how it has been designed, how it works and its programming implementation. The Simulator is used to simulate the traffic in a HSDPA cell in different situations such as different number of users. These users can be either Voice over IP or Web browsing. The Simulator has been programmed with Matlab.

3.1 Introduction

The HSDPA simulator is used to simulate traffic in a HSDPA cell with VoIP and web users. Its main purpose is to find the user throughput and the delay in different situations, depending on the number of users and different distributions of VoIP and web users.

The output of the simulator is a group of vectors that represent the received queues, one for each user, and they contain information about the number of bits received and the delay of those. Then the correspondent throughput of each user can be calculated dividing the number of received bits by the simulation time and the average delay for each user can be found dividing the sum of all the delays by the number of bits received by that user.

3.1.1 Voice over IP Traffic

Voice over Internet Protocol (VoIP) refers to voice communications over IP networks such as Internet or other packet-switched networks.

The Voice over IP traffic is modelled in ON/OFF states: During the ON state there are voice bursts and during the OFF silence. The duration of both ON and OFF states has an exponential distribution with an average of 3 seconds. The total duration of the call is exponentially distributed with an average length of 60 seconds [7] [8].

The characteristics of the exponential distribution can be found in the Appendix 6.2

During the ON state, packets are generated every 20ms, each VOIP packet has 304 bits. During the OFF state no packets are generated.

The ITU-E states that when the mouth-to-ear delay exceeds 250ms the voice quality rapidly deteriorates. When the delay induced by the voice encoder/decoder and other nodes is subtracted, about 80 to 150 ms remains for the node B processing and scheduling [9]. The delay budget is the time available for scheduling and in this simulator is considered to be 80ms.

3.1.2 Web Traffic

Web browsing traffic is modelled as the download of a file every certain time. The model is divided in these two levels:

- Burst Level: Each packet session is formed by one/many packet/file call(s) and each packet/file represents a web page. The inter-arrival of these packets/files is called reading time, and represents the time that the user needs to read the web site, that is, it starts when the users receives the complete website, and lasts until it clicks a link to download another website.

- Packet Level: Each packet/file call is composed of a number of data packets. The distributions of the packet size and of the inter-arrival time are specified at this level.

In the simulator, the following parameters have been picked to represent the web traffic, according to existing studies [10]:

- Reading time: A geometric distribution with average 2 seconds.

- Number of packets per call: Geometric distribution with average equal to 25.

- Packet inter-arrival time: This parameter is fixed to 2ms.

- Packet size: Pareto distribution (Pareto [1.2, 81.5]) with average 480 bytes.

The characteristics of the geometric and Pareto distributions can be found in the Appendix 6.2

Based on [11] a good quality of service indicator is defined by a packet call (web page) delay no longer than 2- 4 seconds.

3.2 Inputs

This is the list of the inputs to the function ‘simulator’:

- Number of users: Number of HSDPA active users in the cell.

- Radius of the cell.

- Frequency.

- Transmission antenna height.

- Reception antenna height.

- Transmission power.

- Standard deviation: This input is used in the shadowing calculation. Typical values are between 6 and 12 dB. In very dense areas with a lot of buildings or with mountains this value is high; on the other hand in very flat areas this value is close to 6 dB.

- Voip_web: It is a vector that has the same length as the number of users, that is it has one position for each user, and it contains zeros and ones: a ‘0’ means that the user is a VoIP and ‘1’ means that that it is a web user.

- Cat: It is a vector that also has one position for each user and it contains the category of the UE of the correspondent user.

- CQI0…CQI30: These matrices have one row for each possible EcIo value and one column for each possible RSCP value and each element is the number of samples that have that particular CQI for that EcIo and RSCP combination. Details on these matrices are explained in the following section 3.2.1.

- Queue_matrix: It is a matrix that is used to update the transmission queues of the VoIP users during the simulation. This procedure is explained in more detail in the point 3.2.2.

- Shadowingtable: This input is used to calculate the shadowing.

- Reading_time_table: Used to calculate the reading time of the web users.

- Packet_size_table: This input is used to decide the packet size of the web applications according to the statistical model.

- Number_packets_table: This table is used to calculate the file size of the web applications.

3.2.1 Creation of the CQI tables

The CQI tables show the relation between the CQI values and the EcIo and RSCP parameters. The EcIo is the signal to interference ratio and the RSCP is the received signal level of the CIPCH.

This relation is not unique, that is for one EcIo and RSCP combination there are different possible CQI values with different probabilities. It is important to keep in mind that the CQI value that a UE terminal reports does not only depend on the EcIo and RSCP but also on other variables, such as the multipath environment, the terminal receiver type and the HSDPA power availability. This is then an approximation of the CQI values based only on the EcIo and RSCP information.

The first step to find the relation between the EcIo and RSCP with the CQI values is to analyse some samples taken from the Orange UK network. These logs were taken in the top ten UK cities and can be processed with the software “Actix”. From all the information available in these samples, the values related with the EcIo, RSCP and CQI are extracted and imported to Matlab for further analysis.

This information is in fact a table, containing in the first column all the possible EcIo values, in the second column all RSCP values, and in the following 31 columns, which is one for each CQI possible value, the number of samples that have that CQI value for that specific EcIo and RSCP combination. This structure is shown in figure 3-1:

EcIo	RSCP	CQI0	CQI1	CQI2	…	CQI30
EcIo1	RSCP1	N° samples	N° samples	N° samples	…	N° samples
EcIo2	RSCP2	N° samples	N° samples	N° samples	…	N° samples
.						
.						
.						

Figure 3-1. Table containing EcIo, RSCP and CQI combinations

Then the minimum number of samples needed is calculated for each EcIo/CQI and RSCP/CQI combination, and the worst case is taken. The percentage of samples that have a specific EcIo/CQI combination is calculated as the number of samples is incremented then when this percentage is stabilized the minimum of samples needed for that combination is found.

This process is done for all the EcIo/CQI combinations and then for all the RSCP/CQI combinations, and then the worst case is chosen as the minimum number of samples, which is: 1.4959e+008 samples.

Using the minimum number of samples, the 31 CQI matrices are created, one for each CQI value. These matrices have one row for each EcIo possible value (from -1dB to -25dB) and one column for each RSCP value (from -40dBm to -120dBm) and contain the number of samples that have that combination of EcIo, RSCP and CQI. As said before, the relation between the EcIo, RSCP and CQI is not unique, so for a certain EcIo and RSCP combination, a histogram of possible CQIs is obtained (different CQI values are possible for the same combination) and one value is chosen taking into account the probabilities of all the possible CQIs.

The figure 3-2 shows the histogram of possible CQI values for the combination EcIo=-10 dB and RSCP=-60 dBm.

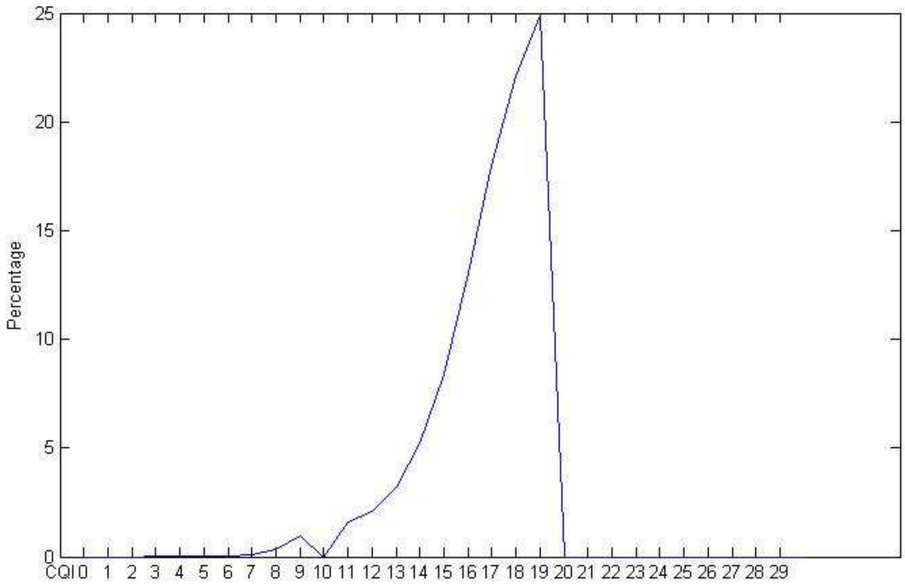


Figure 3-2. CQI histogram for EcIo=-10dB and RSCP=-60dBm

The figures showing the relationship between the EcIo, the RSCP and the CQI can be found in the appendix 6.3. Figure 3-3 represents the distribution for CQI 20:

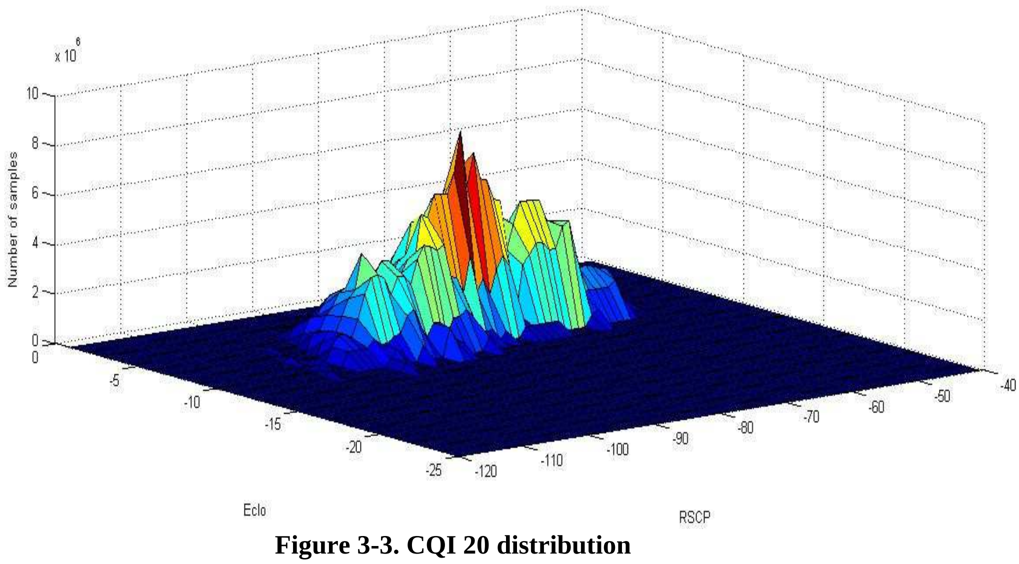


Figure 3-3. CQI 20 distribution

3.2.2 Queue_matrix

'Queue_matrix' is one of the inputs to the simulator and it is used to update the transmission queues of the VoIP users. The application sends packets to the transmission queue and this process has to be done while the simulator is running. As the VoIP traffic is not interactive, this information can be given as an input; this saves time when running the simulator as solving the traffic distribution equations to model the traffic adds delay. The input ‘queue_matrix’ contains the information needed to update the transmission queues of the VoIP users every TTI.

This matrix has one row for each VoIP user, and one column for every TTI of the simulation (20 seconds simulations= 10.000TTIs=10.000 columns) and indicates when new bits need to be added in the transmission queue for that specific user in that TTI. The elements of the matrix are ones and zeros. Zero meaning no changes and one meaning that a new VoIP packet, which is 304 bits, is added to the transmission queue for the correspondent user to that row and in the TTI correspondent to that column. The structure of ‘queue_matrix’ can be seen in figure 3-4.

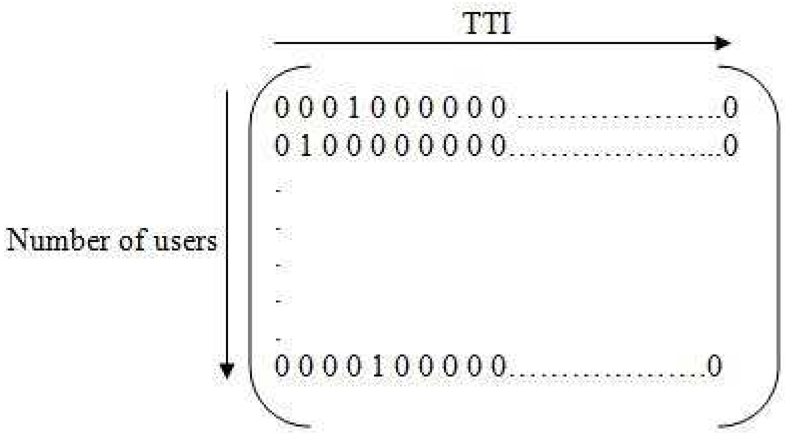


Figure 3-4. queue_matrix

The rows that form this matrix are calculated separately using a function called ‘voip’. The output of this function is a vector, which is one of the ‘queue_matrix’ rows correspondent to one VoIP user.

This function calculates the rows that form the ‘queue_matrix’ using the model explained in 3.1.1 to represent VoIP traffic. This input must be filled with the number of rows equal to the number of VoIP users in the simulation. For each snapshot, a new ‘queue_matrix’ is calculated.

3.2.3 Inputs to model the web traffic

As the web traffic is interactive it can not be modelled entirely with an input. So this traffic is modelled during the simulation, this is a time consuming process but some inputs are provided to improve this. These inputs are the ‘reading_time_table’, the ‘packet_size_table’ and the ‘number_packets_table’. They are used to model the web traffic without having to solve every time the equations to calculate the reading time, the packet size and the number of packets that form an entirely web site. Instead an element of the tables is picked randomly to represent the correspondent parameter.

3.2.3.1 Reading_time_table

The reading time starts when the last packet of a packet call is completely received by the user and ends when the user makes a request for the next packet call. It is defined as a Geometric distribution with average 2 seconds. Instead of calculating the reading time every time a user finishes a transfer, it has been calculated a hundred times, and the obtained values are in this table. Then when the reading time needs to be calculated a value from this table is picked randomly saving time as solving equations is a slow process.

The simulator has been designed to work with a time reference of TTIs instead of continuous time, the reading time contained in this table is then in number of TTIs, which is the reading time divided by 2 ms.

Figure 3-5 shows the values of the reading time table:

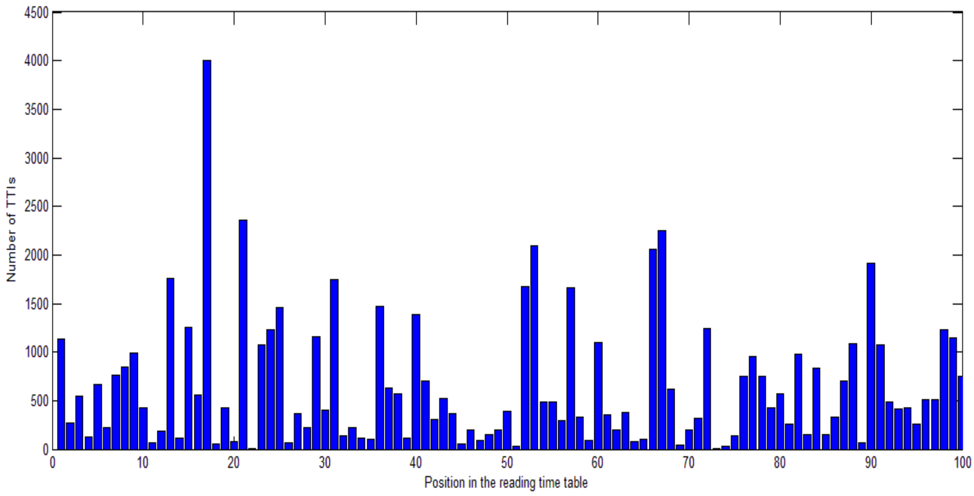


Figure 3-5. Values on the reading_time_table

3.2.3.2 Packet_size_table

The packet size is the number of bytes that conforms a packet, several packets conforms a web site. Based on [7] the packet size is defined as a Pareto distribution with average 480 bytes, the shape parameter equals to 1.2 and the minimum possible value is equal to 81.5 bytes. As explained before in the 3.2.3.1, solving equations is time consuming, and for the same reason this table is created. Every time a packet is send the packet size needs to be calculated just picking one element from the table randomly.

This table contains 100 possible packet sizes, and is represented in figure 3-6:

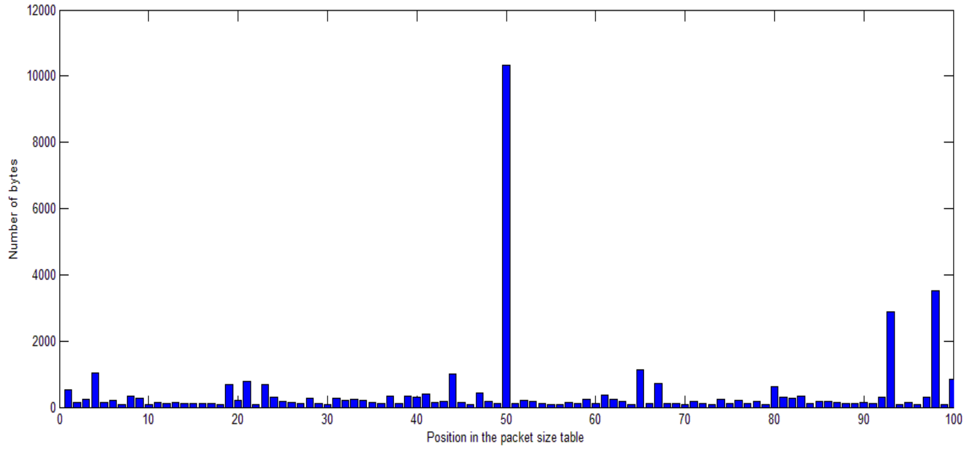


Figure 3-6. Values on packet_size_table

3.2.3.3 Number_packets_table

This input contains information about the number of packets that form a web site. The number of packets follows a geometric distribution with an average equal to 25 packets.

As explained in the previous points 3.2.3.1 and 3.2.3.2 this table contains a hundred possible values for the number of packets and these are shown in figure 3-7:

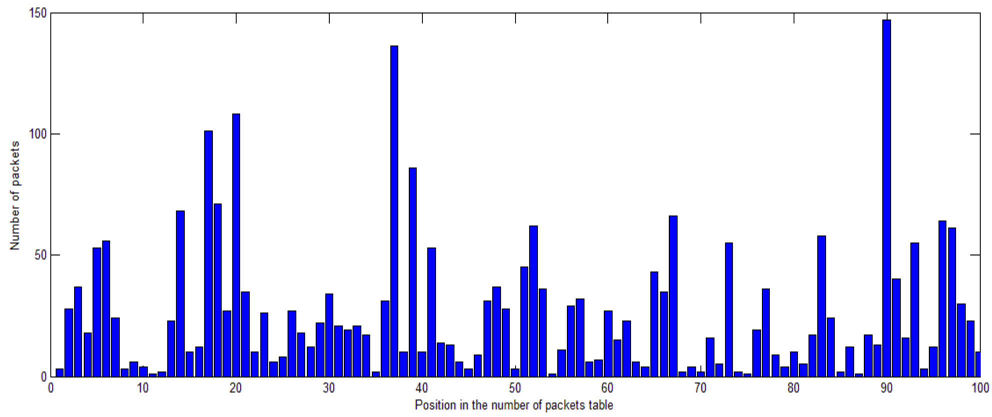


Figure 3-7. Values on number_packets_table

3.3 The simulator

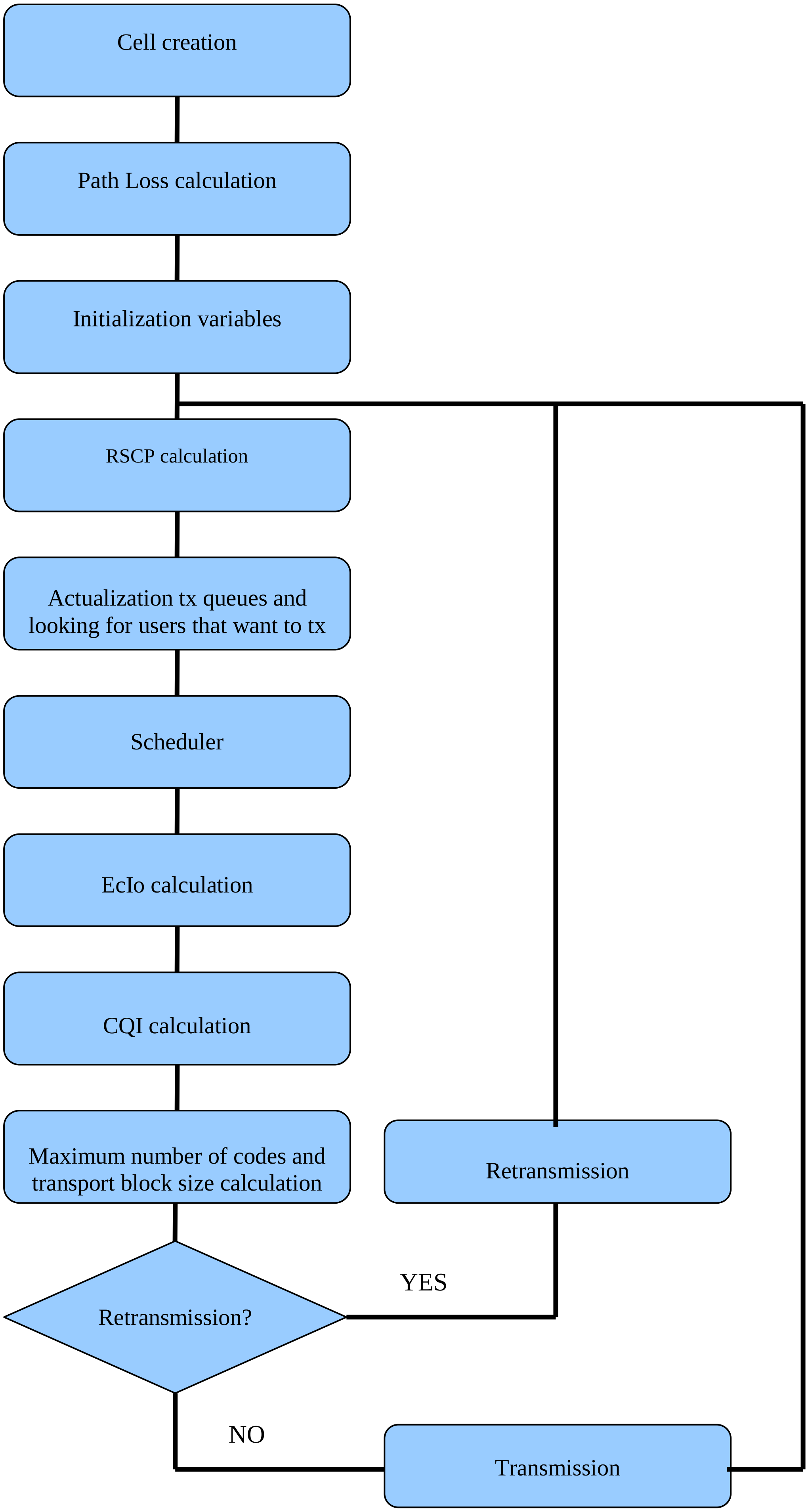
The simulator is the main function, it is used to obtain the throughput and the delay in the reception queues of the users. It simulates an HSDPA cell with the conditions indicated in the inputs. Then the output is a group of vectors, one for each user, that indicate the number of bits and the delay of these at the end of the simulation time.

The delay is considered the time that has passed since that bit is put on the transmission queue until it is received by the user.

This time is measured in TTIs as the simulation has one iteration for every TTI and for each iteration:

- The RSCP is calculated taking into account the shadowing fading effect.
- The transmission queues are updated.
- The scheduler decides which users will transmit and which will not.
- The EcIo is calculated.
- With the EcIo and the RSCP the CQI is calculated for the users that will transmit.
- The number of bits that will be transmitted is assigned.
- The reception queues of the users that have transmit in that TTI are updated if the frame is received ok, if not these bits are retransmitted after a waiting time of at least 6 TTIs, that is 6x2ms=12ms.

The following diagram shows the steps that the simulator follows:


Cell creation
Path Loss calculation
Initialization variables
RSCP calculation
Actualization tx queues and looking for users that want to tx
Scheduler
EcIo calculation
CQI calculation
Maximum number of codes and transport block size calculation
Retransmission
YES
Retransmission?
NO
Transmission

3.3.1 Cell Creation

This step creates a hexagonal cell with the number of users and the radius indicated in the inputs.

The edges of the cell are given by the equations:

1. y=R
2. y=-R
3. y=-2x+2R
4. y=2x-2R
5. y=-2x-2R
6. y=2x+2R

Then the users are allocated randomly within the edges of the cell, this is done creating a random position in the x axe, between -R and R and then a random y position between –R and R but this one is only taken if it is inside the edges, if not another possible y position is calculated.

Figure 3-8 shows a cell with Radius equal to 1km and 10 users:

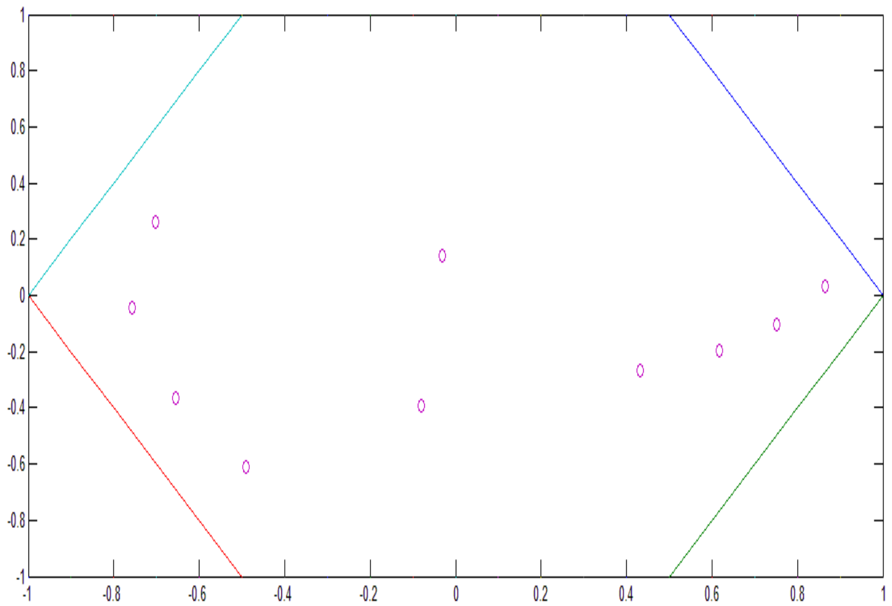


Figure 3-8. Cell with Radius=1km and Number of users=10

3.3.2 Path Loss Calculation

The path loss is the reduction of the signal level as it propagates through space. In this design, the path loss has been approximated with the Okamura-Hata formula, which is:

$$L(dB) = 69.55 + 26.16 \cdot \log(f) - 13.82 \cdot \log(ht) - a(hm) + \left(44.9 - 6.55 \cdot \log(ht)\right) \cdot \log(d)$$

Where

f is the frequency in MHz
ht is the height of the transmitting antenna
hm is the height of the receiver
d is the distance in km

And a(hm) is defined as:

- Small/medium city:

$$a(hm) = \left(1.1 \cdot \log(f) - 0.7\right) \cdot hm - \left(1.56 \cdot \log(f) - 0.8\right)$$

- Big city with a frequency smaller or equal to 400MHz:

$$a(hm) = 8.29 \cdot \left(\log(1.54 \cdot hm)\right)^2 - 1.1$$

- Big city with a frequency greater than 400MHz:

$$a(hm) = 3.2 \cdot \left(\log(11.75 \cdot hm)\right)^2 - 4.97$$

3.3.3 Initialization of the Variables

This step is used to initialize all the variables used in the simulator, these variables are:

- position_queue_tx: Indicates the position in the transmission queues, this position indicates the next bits to transmit, it has one element for each user which points to the position in the transmission queue of that user and it is initialized with all its elements equal to one. Usually this position is one, but when there is a retransmission, the position is moved to the next bits to be transmitted, and after 6 TTIs is moved forward again to include the bits that need to be retransmitted. This procedure is explained in more detail in the section of retransmission.

- tx: This variable indicates when the users want to transmit, when a user wants to transmit, the correspondent element of this vector is set to one, if it does not have bits to transmit is set to zero. It is set to zero for its initialization.

- web_matrix: This matrix is used to update the transmission queues of the web users, it contains the correspondent information of a packet call for each user, that is the number of packets and the packet size of these. One row of this matrix has 150 elements, the first X elements contain packet sizes, extracted from the packet_size_table and the rest are set to 0. X is calculated extracting a value from the number_packets_table. When a user finishes the transmission of a web site, the row is updated with the correspondent value of the next web site. It has one row for each web user and it is initialized with the correspondent values of the first web sites. The structure of this matrix can be seen in figure 3-9.

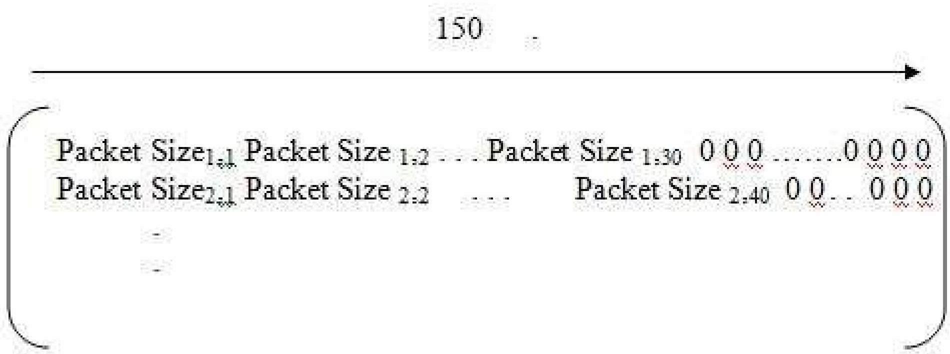


Figure 3-9. Structure of 'web_matrix'

- position_web_matrix: This vector has one row for each web user, and it points to the correspondent position in the web matrix, knowing then, how many bits need to be added to the transmission queue of that user. It is initialized to a random number, so not all the web users start transmitting at the same time.

- reading_time: This variable is a vector that has one element for each web user, it contains the reading time. In the initialization, it is set to zero.

- waiting: This vector has one element for each web user and it is initialized to zero. During the simulation its value can be:

  - Zero: The user is transmitting.
  - One: The lasts bits of a web site have been already added to the transmission queue, but have not yet been received by the user.

- Two: The user has received the entire web site and is ‘reading it’ so we are waiting for the reading time to finish and then start the next transmission.

- Web_user: This vector has one element for each user, if the user is voip it contains a zero and if the user is web it contains the counter or user number that reference it with the rest of the web information. For example if there are 20 users, and the first 10 are voip and the next 10 are web, that means that the user 11 is the first web user, then it will contain a one in the 11 position of this vector.

- queue_tx1…queue_tx100: These matrices represent the transmission queues, there is one for every possible user (a maximum of 100 users) if the simulation is done for less users, not all of these matrices are used. These matrices are updated every TTI if there are bits from the applications to be added in the transmission queues. These matrices have four rows: The first row contains the TTI when the bits are added to the queue, the fourth row contains the number of bits that are added on that TTI. The second and third rows are used when there is a retransmission, then the correspondent element in the third row is set to one, indicating that those bits need to be retransmitted and the third row is then set to the TTI when this bits have been send but not received ok. The retransmission process is explained in more detail in the retransmission section. In the initialization, all these queues are set to zero.

- queue_rx1…queue_rx100: These are the matrices that represent the reception queues, as for the transmission queues, there is one for every possible user (the maximum number of users is 100) but when the simulation is done for less users, only the needed ones are used. These matrices have two rows, the first one contains the delay that the bits received have and the second one the number of bits with that delay, these are updated with every transmission and these process is explained in more detail in the transmission section. These variables are set to zero during the initialization.

3.3.4 RSCP calculation

The Received Signal Code Power (RSCP) is defined as the received power of the pilot channel; it is then calculated with the CPICH transmitting power (33dBm), the path loss and the shadowing.

The shadowing, also known as slow fading, is the variation on the signal level due to the shadowing of obstacles affecting the wave propagation. Figure 3-10 shows the effect of the shadowing as a mobile station moves around an omnidirectional antenna [12]:

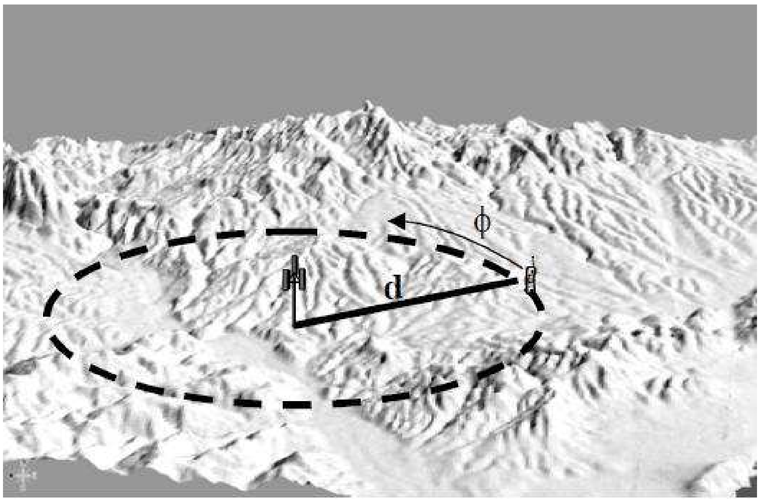


Figure 3-10. Mobile Station moving aroung an omnidirectional antenna

Although the mobile is all the time at the same distance, because of the shadowing of the different obstacles in the environment such as mountains, the received power changes. Figure 3-11 shows the average power and the variation on the received power as the mobile station moves around the omnidirectional antenna.

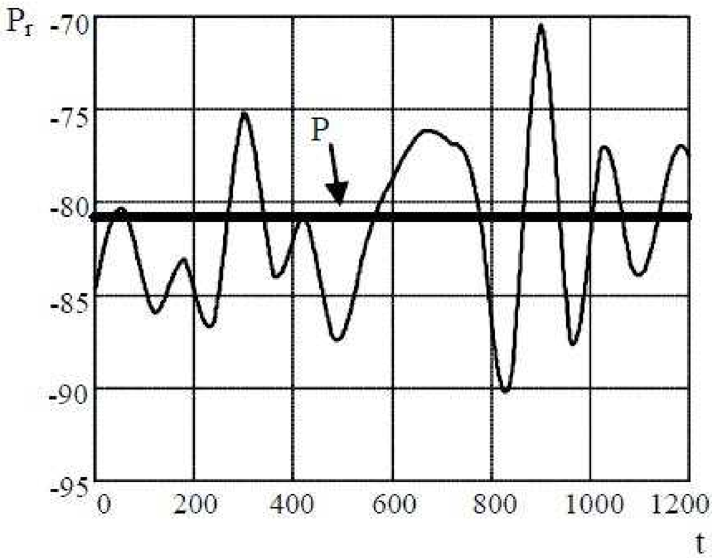


Figure 3-11. Average received power and received power

The effect of the shadowing is modelled with the formula:

$$f(\mathrm{Pr}) = \frac{1}{\partial\sqrt{2\Pi}}\exp\left(-\frac{(\mathrm{Pr}-P)^2}{2\partial^2}\right)$$

Where Pr is the average received power, which is the transmit power minus the path loss, and ∂ is the standard deviation.

The standard deviation is defined by the propagation environment and its values are between 6 and 12dB. Areas with mountains and high buildings show greater standard deviation values. One the other hand flat fields show values around 6dB.

As explained before, solving a formula is a slow process and it will increase the time that the simulator needs to run significantly. For this reason, the input table ‘shadowingtable’ is used. This table contains a hundred shadowing possible values for each possible value of average received power.

3.3.5 Actualization of transmission queues and looking for users that want to transmit

The applications send packets to the transmission queues and these are updated every TTI. For the VoIP users, the transmission queues are updated with the values on the ‘queue_matrix’ and for the web users, with the values on the ‘web_matrix’.

When there are new bits to be added on the transmission queue of a VoIP user, the element on ‘queue_matrix’ correspondent to that user and that TTI is one. Then 304 bits are added to the transmission queue of that user. As explained before, the transmission queues are represented with a matrix that has four rows, the first row contains the TTI and the fourth the number of bits, the second and the third are used for retransmissions. Then when this happens, a new column is added to the transmission queue of that user, the first row containing the correspondent TTI and the element on the forth row being 304. The second and third row elements are left with the value zero.

When the web application is sending packets to the transmission queue of a web user, the number of bits indicated on the ‘web_matrix’ for that user are updated in the transmission queue and as for the VoIP user, a new column is added containing the TTI and the number of bits.

When the transmission queues are updated, it is time to look for which users have bits to transmit. This includes looking for retransmissions too.

When there is a retransmission, the elements on the transmission queue that need to be retransmitted are updated, this is done indicating in the second row that those bits need to be retransmitted, which is done putting that value to one, and the third row is updated putting that value to TTI+6. This indicates when those bits can be retransmitted which is after 6 TTIs. Then the value for that user in the ‘position_queue_tx’ is updated pointing to the next bits to transmit.

Then when looking for users to transmit, the users that have bits that need to be retransmitted are checked, if the value on the third row is greater or equal to the current TTI, the value in ‘position_queue_tx’ for that user is updated pointing to the bits that need to be retransmitted. Then the element for that user in the vector ‘tx’, is updated with a one as it has bits to retransmit.

Also all the transmission queues are checked, and ,if they have bits to transmit, the ‘tx’ vector is updated with a one indicating that the user wants to transmit.

This function also updates the 'web_matrix' when a web user finishes a packet call and updates the variables waiting and reading time to indicate when to start the next transmission.

3.3.6 Scheduler

The scheduler decides which users will transmit on that particular TTI. A maximum of three users are able to transmit on the same TTI. The total number of codes in HSDPA is 15, if only one user wants to transmit, it will have available all the codes and will take the number of codes correspondent to its radio conditions.

On the other hand, if more users want to transmit the total number of codes will be divided between those users. There are different types of schedulers, in this simulator, the Round Robin has been developed.

3.3.6.1 Round Robin

In this scheduler, the users are served in a cyclic order. The Round Robin scheduler allocates equal resources to all the users regardless of their radio conditions. This method outstands for its simplicity while ensuring that all the users in the cell have the same possibilities to transmit. The following figure shows the Round Robin scheduling for 10 users:

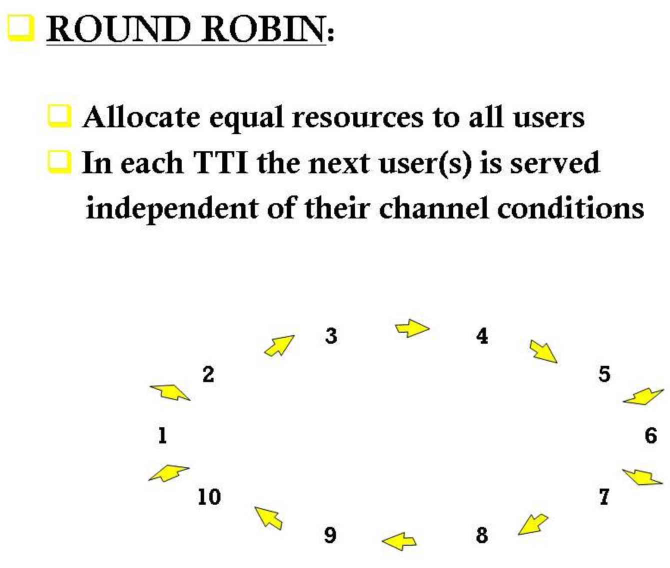


Figure 3-12. Round Robin

The last user that has transmitted is indicated on the variable 'last_tx', to maintain the order for the next TTI. For a particular TTI, starting with 'last_tx +1', the users are checked, and the three first ones that want to transmit are scheduled.

3.3.7 EcIo Calculation

The EcIo is the signal to interference ratio, which is calculated as the relation between the CPICH and the total power received by the UE:

$$Ec / Io = \frac{RSCP}{RSSI}$$

Where the RSCP is the received power of the pilot channel and the RSSI (Received Signal Strength Indicator) is the total power received by the UE.

The RSSI contains power received from the same cell and from the neighbour cells then it can be defined as:

$$RSSI = P_{Intra} + P_{Inter}$$

P_{Intra} has been approximated as the maximum transmitting power (43dBm) with the correspondent path loss and shadowing applied.

As the neighbour cells are not simulated, the P_{Inter} is approximated as a percentage of P_{Intra}:

$$P_{Inter} = P_{Intra} \cdot F$$

Where F is the inter cell interference factor and it is defined as 1.2 in the cell centre and 1.6 in the cell edge. For the rest of the positions F is calculated as the proportional value with their distance to the cell centre.

3.3.8 CQI calculation

The Channel Quality Indicator (CQI) is approximated using the input tables aCQI0…aCQI30. As explained before these tables have been created using information from the real network and contain the percentage of that CQI value for a particular EcIo and RSCP combination.

Then having already calculated the EcIo and the RSCP the CQI value can be easily calculated.

As explained on 3.2.1 a histogram containing the probabilities of each CQI value for that particular EcIo and RSCP combination is made.

Then the cumulative probability function is calculated adding the probabilities of each CQI, then a random number r between 0 and 1 is chosen, and the CQI that contains this number on the cumulative function is chosen, this process is explained in the following figure for the case of EcIo= -5 dB and RSCP = -60 dBm:

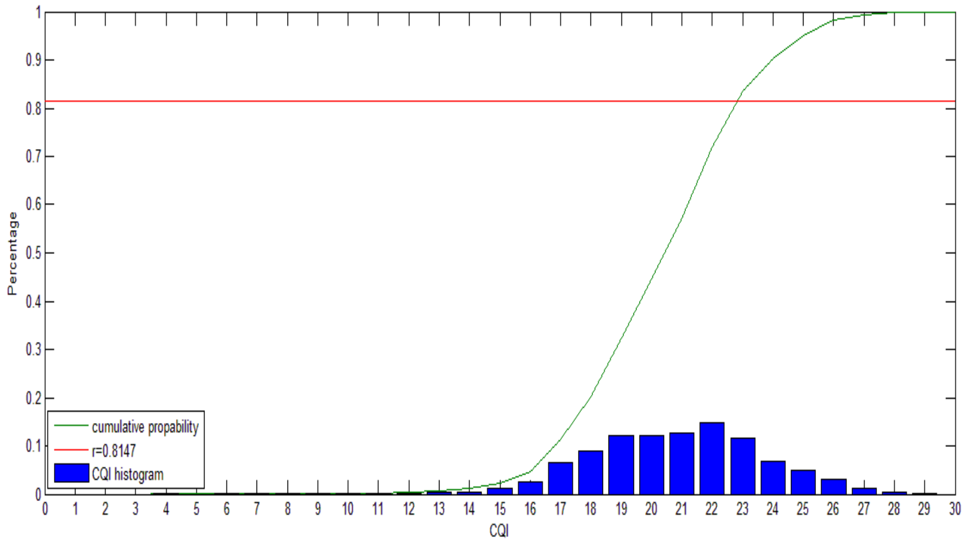


Figure 3-13. CQI election

In this case, the random number r is 0.8147, and then the CQI that would be chosen is 23.

3.3.9 Maximum number of codes and transport block size

The maximum number of codes and transport block sizes are defined by the UE category and the radio conditions indicated in their CQI reports.

The tables that indicate the maximum number of codes and transport block size are defined in the 3GPP specifications [10], and can be found in the appendix 6.1.

It is important to notice that these values are not necessarily the ones assigned to the transmission but the maximum ones. If only one user is served, then it will have all the resources and it will uses the maximum number of codes and the maximum TBS, but if more users want to transmit then the recourses are divided between the users and it may not get the number of codes wanted. Then the data rate assigned to that user for the transmission will be the maximum data rate multiplied by the number of codes assigned and divided by the maximum number of codes.

$$dr = \frac{TBS}{TTI} \cdot \frac{NCodes}{MNCodes}$$

Where, *NCodes* is the number of codes assigned, and *MNCodes*, are the maximum number of codes for that CQI and mobile category.

3.3.10 Probability of retransmission

The probability of retransmission is set to 0.01, which means that one of every hundred transmissions will need to be retransmitted.

Then every TTI a random number between 0 and 1 is picked, if this number if smaller or equal to 0.01 that block is considered to be received with errors and it needs to be retransmitted.

If this number is greater than 0.01 the transmission is considered not erroneous and the function to execute the transmission is called.

3.3.11 Transmission

During the transmission, three main steps are done:

- A data rate is assigned to each user that is going to be served.

- The reception queue of the users that have been served is updated with the number of bits received and the delay of these.

- The transmission queue of that user is updated, deleting the bits that have already been transmitted and with ‘position_queue_tx’ pointing to the next bits to be transmitted.

As explained before, there are 15 codes available; if only one user is served then it can take all the desired codes as all are available for it.

If two users want to transmit, the codes are divided between them. Unless one or both of the users wants less codes, the assignment made is 8/7 codes for each.

And if three users are being served, the codes are divided between them if all the users want more that 5 codes, the codes assigned to each users are 5/5/5, if one or more users ask for less than 5 codes, they are given the codes that they have asked, and the rest of codes are divided between the other users.

Then for each of the served users, the data rate is calculated with the number of codes assigned and the transport block size knowing then the number of bits that can be transmitted in one TTI.

Then the reception queues are updated with the number of bits received, and its delay. The delay is calculated as the current TTI minus the TTI indicated in the first row of the transmission queue, which indicates the time when these bits have arrived to the transmission queue.

Finally, the columns on the transmission queue that represents the bits that have been send are deleted and the variable ‘position_queue_tx’ is updated to indicate the next bits to transmit.

3.3.12 Retransmission

When there is a retransmission the data rate correspondent to each user, is calculated as explained for the case of transmission in 3.3.11. Then the number of bits that are needed to be retransmitted are obtained, and then for these bits, the values in the second row and the third row are updated accordingly to indicate the need of retransmission.

The second row is updated with a ‘1’, to indicate that those bits need to be transmitted again, and the third row is updated with the current TTI, indicating when those bits have been received wrong. Then the ‘position_queue_tx’ is updated pointing to the next bits to be transmitted.

After six TTIs these bits can be transmitted again; every TTI the simulator will check the third row of the transmission queues, when the current TTI is higher or equal to the one indicated in the transmission queues plus six it will update the variable ‘position_queue_tx’ to point to those bits, and then they will be transmitted again.

3.4 Outputs

The outputs of the simulator are the variables queue_rx1…queue_rx100, which represent the number of bits received by that user and the delay of these.

With the received number of bits, the throughput can be calculated as the number of bits divided by the simulation time.

The delay indicates the quality of the received bits. As explained before, in the case of VoIP a maximum delay of 80 ms which is 40 TTIs is considered, and or browsing users, a maximum web page delay of 2 seconds, which is 1000 TTIs.

4 Simulation Results

This chapter shows the results obtained in different simulations. The chapter has been divided in three main blocks:

- Only VoIP: In this block the results for the simulations with only VoIP users in the cell are shown.

- Only web: In this section the results for simulations when only web users are in the cell are shown.

- Mixed: When VoIP and web browsing users are in the cell in a 50/50 situation.

All the simulations have been done with the following parameters:

- Radius of the cell equal to 1Km.

- Frequency equal to 2150Mhz.

- Transmission antenna height equal to 50m.

- Reception antenna height equal to 1.5m.

- Transmission power equal to 43dBm.

- Standard deviation equal to 10.

- Mobile categories equal to 8.

The results shown in this chapter are the average of 10 snapshots. For each of these snapshots, the positions of the users have been fixed, so the positions have been calculated once for each snapshot, and have been maintained for the different simulations. The simulation time is set to 20 seconds, which is 10.000 TTIs.

4.1 Only VoIP

In this section the results from the simulations for the case when only VoIP users are in the cell are explained. Different simulations for different number of users have been run, showing then the results in the throughput and the delay when the number of users is incremented. VoIP applications have a low throughput but they have thigh delay requirements. Figure 4-1 shows the average throughput as the number of users is incremented:

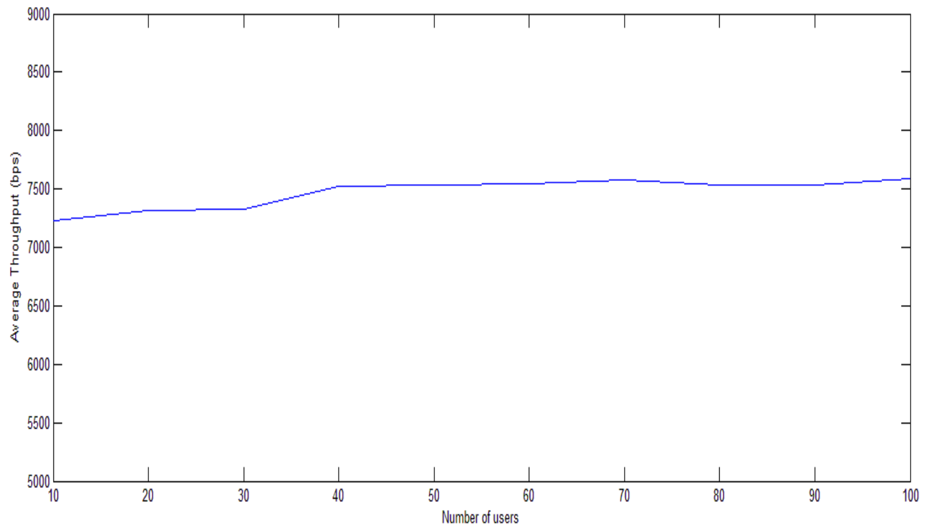


Figure 4-1. Average Throughput

As it can be seen the average throughput of the users is maintained stable around 7.5Kbps. The throughput of each user is very different due to the nature of VoIP. It is important to understand that this difference of throughput between users is not due to the users starving of data but the lack of bits to transmit, that is the VoIP application of that particular user has that throughput too or in other words all the bits in the transmission queues are received.

Figure 4-2, shows a histogram of all the individual throughputs of the users for the 10 snapshots and for the case of 100 users in the cell.

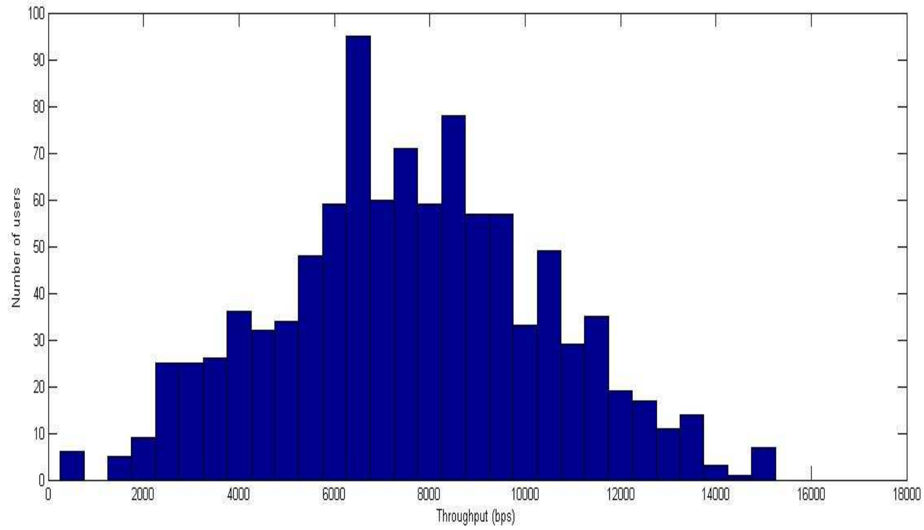


Figure 4-2. Individual Throughput

Figure 4-3 shows the total cell throughput; for users to be starved of data, the total cell throughput must arrive to 10Mbps, as with 15 codes and CAT 8 User Equipments this is the maximum throughput.

As it can be seen in this figure, as the VoIP applications have low throughput not even for the case of 100 users in the cell the maximum cell throughput is achieved and then the total cell throughput increases with the number of users:

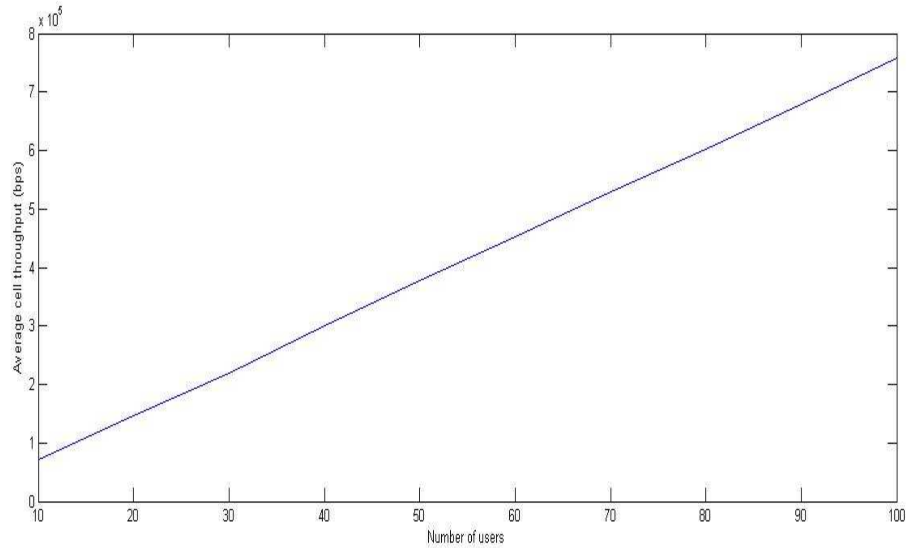


Figure 4-3. Total Cell Throughput

As explained before, VoIP has tight delay requirements, and the delay should not be higher than 80ms. Figure 4-4 shows the average delay as the number of users is incremented:

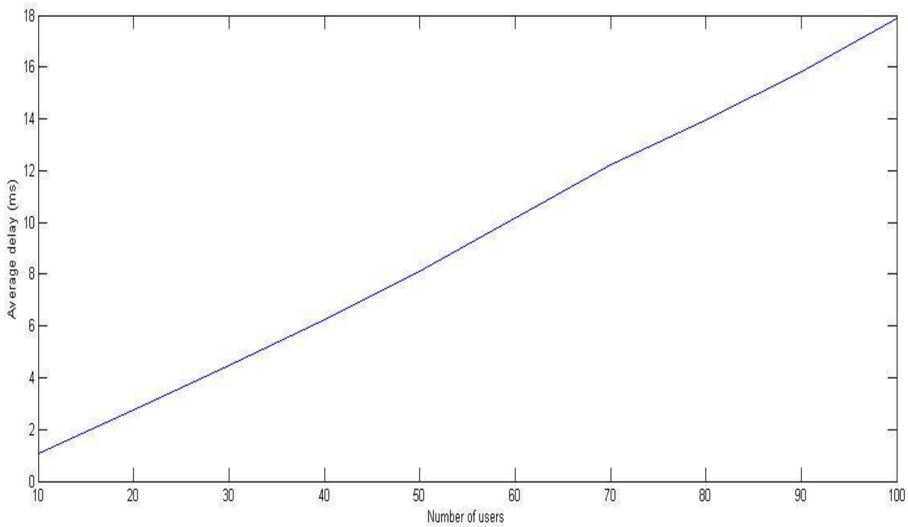


Figure 4-4. Average Delay

As it can be observed, the higher the number of users in the cell, the higher the delay is. The reason for this to happen is that then the users have to wait more time to be scheduled.

As it can be seen, the average delay is smaller than the threshold of 80ms even for the case of a hundred VoIP users in the cell that though does not mean that all the bits are received with a small delay. The percentage of bits received with more delay than the threshold of 80ms, named ‘p’, has been calculated for all the users. In the following figure it can be seen the average percentage of users in the cell that have p≥2% as the number of users is incremented. The tables showing the values for each snapshot can be seen in the appendix 6.4.

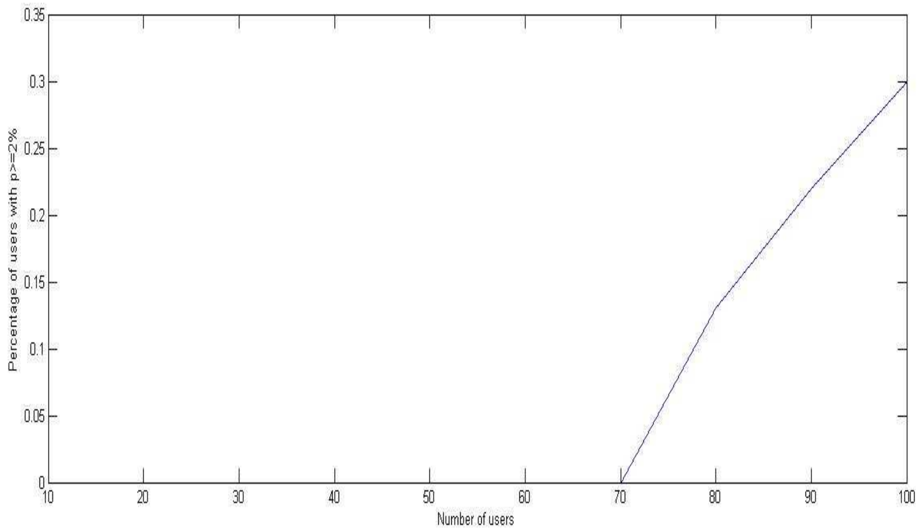


Figure 4-5. Percentage of users with p≥2%

As it can be seen, the percentage of users with bits received above the delay threshold increments as the number of users in the cell is incremented. For the case of only VoIP users in the cell, the number of bits received over the threshold is kept low. The percentage of users receiving more than 2% of the bits with higher delay than the threshold is small, and as it can be seen in the tables on the appendix 6.4 no user is receiving more than 5% of the bits with a delay over the threshold. It can be said then, that the overall results for this situation are good.

4.2 Only Web

In this section a cell with only web users is considered. Different simulations are run incrementing the number of users and the changes in the throughput and delay are explained. Figure 4-6 shows the average throughput as the number of users in the cell is incremented:

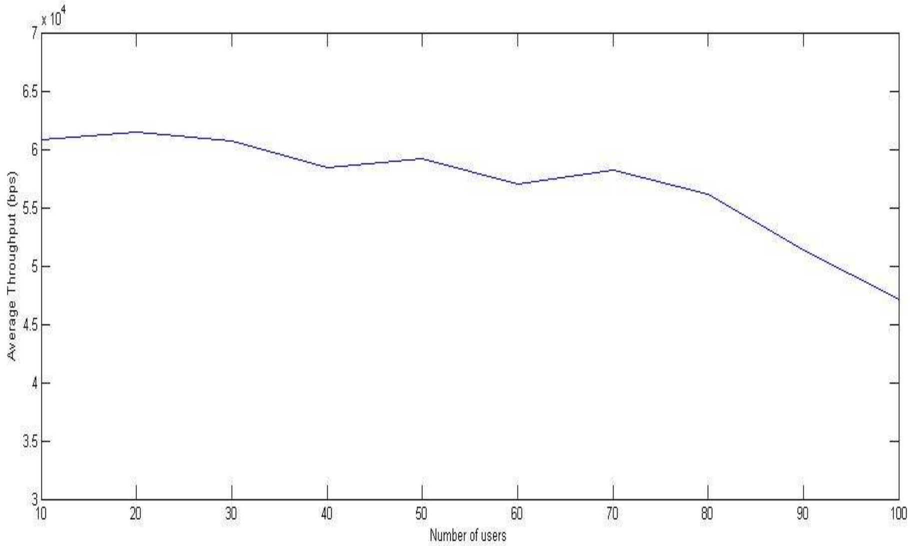


Figure 4-6. Average Throughput

As this figure shows, the average throughput decreases as the number of users is incremented, the reason of this is that the web browsing traffic is interactive, and the higher the delay of the web sites, the smaller the amount of traffic generated over a period of time. Users wait to receive all the bits of one web site, then they read it and after that reading time they click to download another website. So when the delays are high, the number of websites that are downloaded over a simulation time is smaller.

The individual throughput of the users is very variable, and this is due to the nature of the browsing traffic. Figure 4-7 shows a histogram of all the individual throughputs for the simulation case of a 100 users are in the cell:

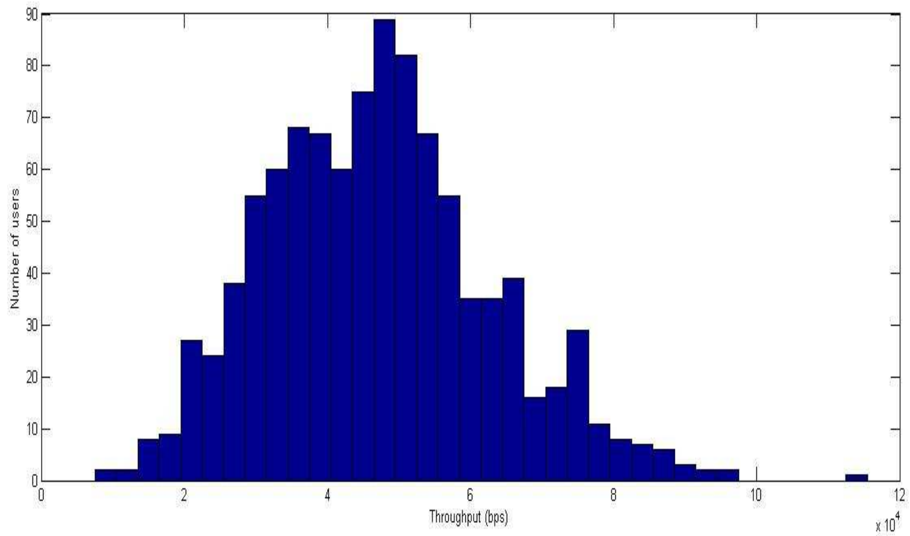


Figure 4-7. User throughput distribution

With the total cell throughput we can see if all the traffic can be handled, the maximum total cell throughput for this conditions is 10Mbps, figure 4-8 shows the total cell throughput as the number of users is incremented and it can be seen that not even for the case of a hundred users in the cell the maximum throughput is achieved, which means that all the generated traffic is handled.

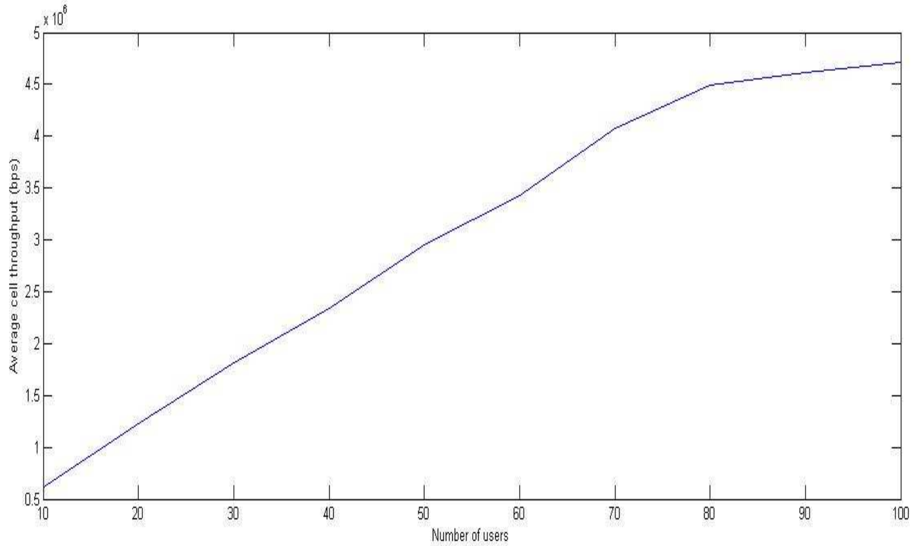


Figure 4-8. Total cell throughput

It can be observed that the average total cell throughput increases more rapidly from 10 to 80 users in the cell and from that point it still increases but at a lower rate, this is due to the decrease in the average user throughput explained before.

Figure 4-9 shows the average delay as the number of users is incremented:

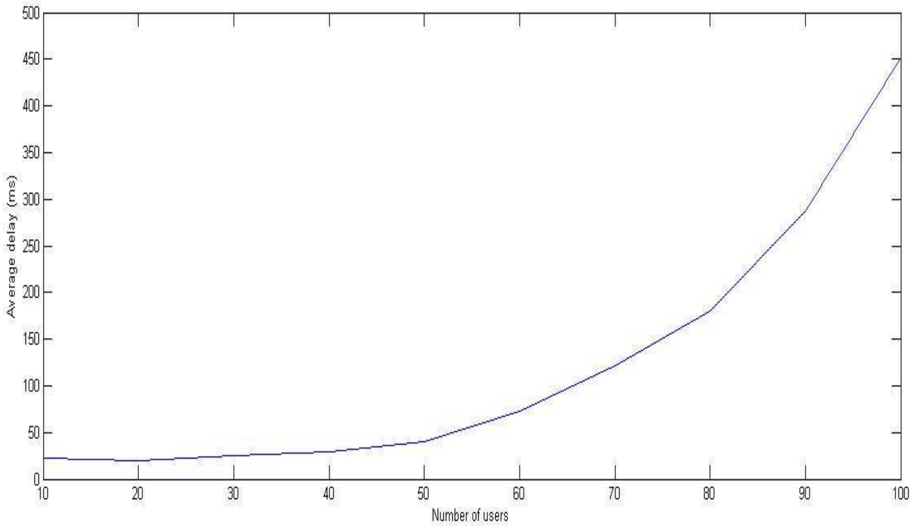


Figure 4-9. Average Delay

As this figure shows, the average delay increases as the number of web users in the cell is incremented. Although the average delay is relatively low for all the cases, some of the bits are actually received with high delays. A good QoS indicator for web traffic is that the delay for downloading a web site should be lower than 2-4 seconds.

The variable 'pw' is defined as the percentage of websites received by one user with more delay than the threshold of 2 seconds. The average number of users with pw≥5%, pw≥10% and pw≥20% for different number of users in the cell is shown in the following figures, and the correspondent values can be seen in the tables of the appendix 6.5:

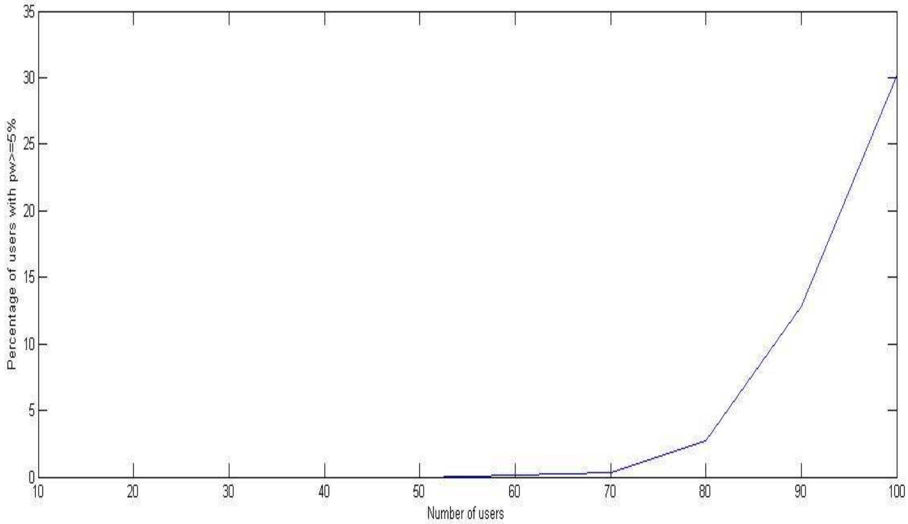


Figure 4-10.Percentage of users with pw≥5%

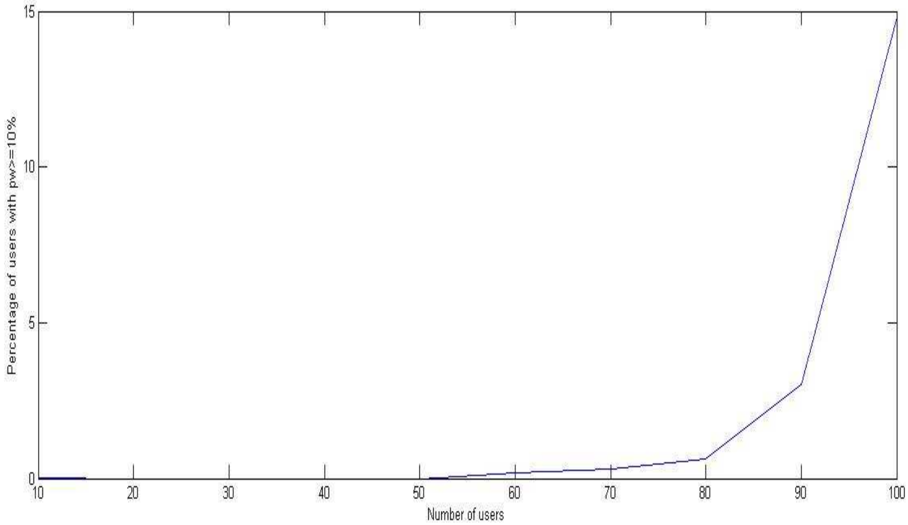


Figure 4-11. Percentage of users with pw≥10%

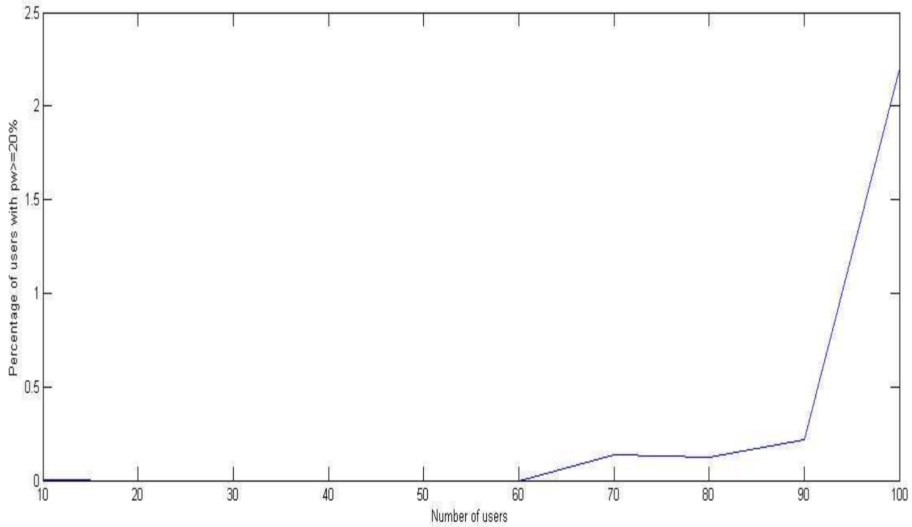


Figure 4-12. Percentage of users with pw≥20%

As these figures show, the number of users receiving websites over the delay threshold increments as the number of users in the cell is incremented. When a 100 are in the cell the percentage of users receiving web sites over the threshold is considerably high, having 30.2% of the users receiving more than 5% of the websites with more delay than the permitted, 14.8% of the users with pw≥10% and 2.2% of users with pw≥20%. Considering as unacceptable to have more than 5% of the users with a pw≥5%, to provide a good quality of service the maximum number of simultaneous WEB users shall be kept lower or equal to 80.

4.3 Mixed Traffic

A cell with VoIP and web traffic is simulated in this section. The number of users is incremented and the throughput and delay for the VoIP and web users are analysed. The number of users in the cell is 50 % VoIP and 50 % web. Figure 4-13 shows the average throughput for the VoIP users and figure 4-14 the average throughput for the web users:

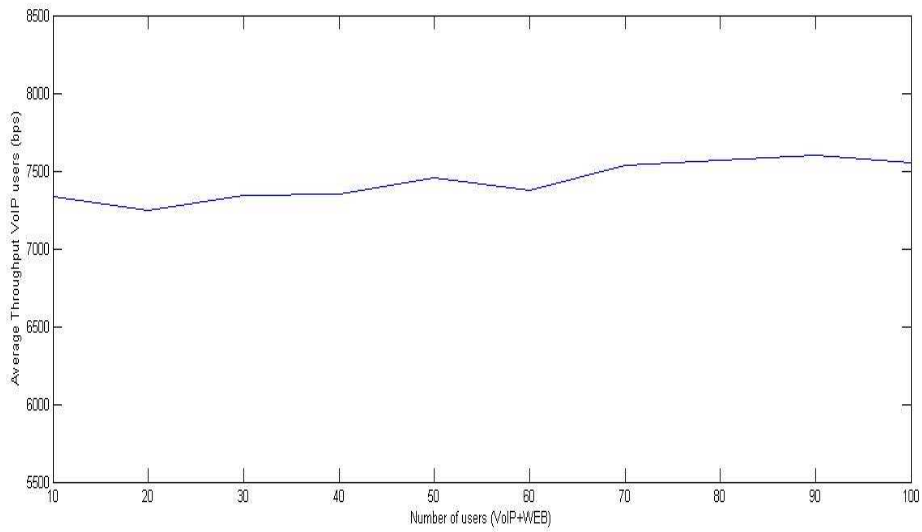


Figure 4-13. Average Throughput VoIP users

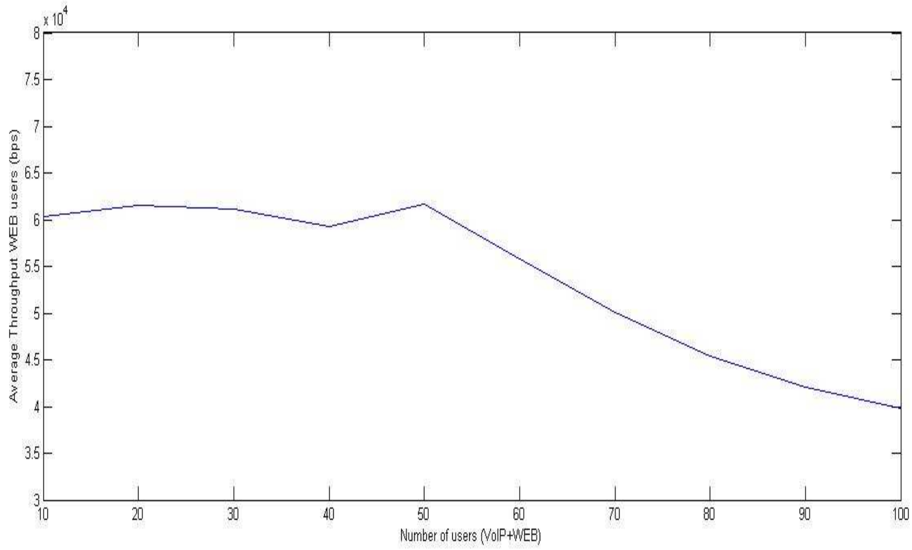


Figure 4-14.Average Throughput web users

The average throughput of the web users decreases with the number of users in the cell, notice though that this is not due to the cell not being able to handle all the generated traffic but to the nature of web traffic, as web traffic is interactive, the higher the delay the less traffic is generated on a certain amount of time (as explained in 4.2).

Figure 4-15 shows the total cell throughput, and it can be seen that all the traffic can be handled even for the case of a hundred users in the cell (50 VoIP users+50 web users) as the total cell throughput is incrementing as the number of users is incremented.

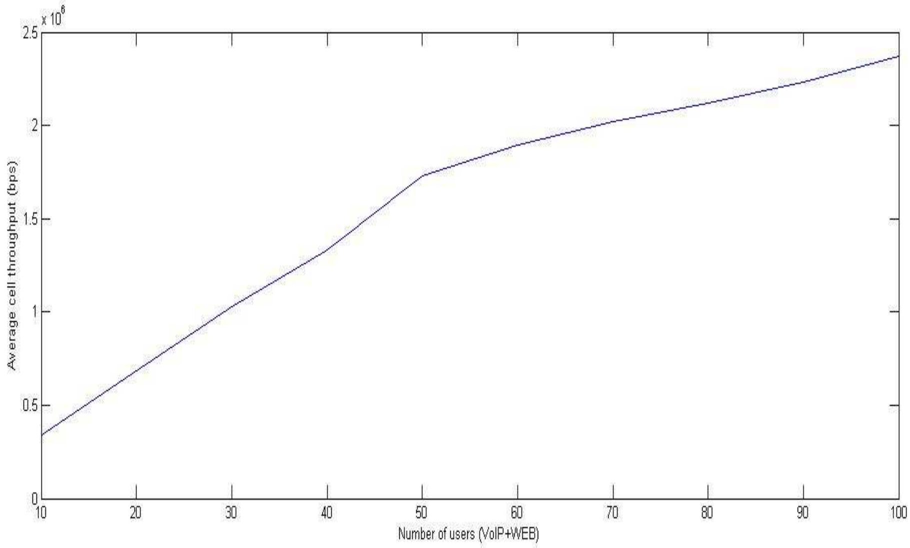


Figure 4-15. Total Cell Throughput

As explained in 4.2, the rate that the average cell throughput is incrementing decreases with the number of users due to the web users generating less traffic.

Figures 4-16 and 4-17 show the average delay for VoIP and web users respectively:

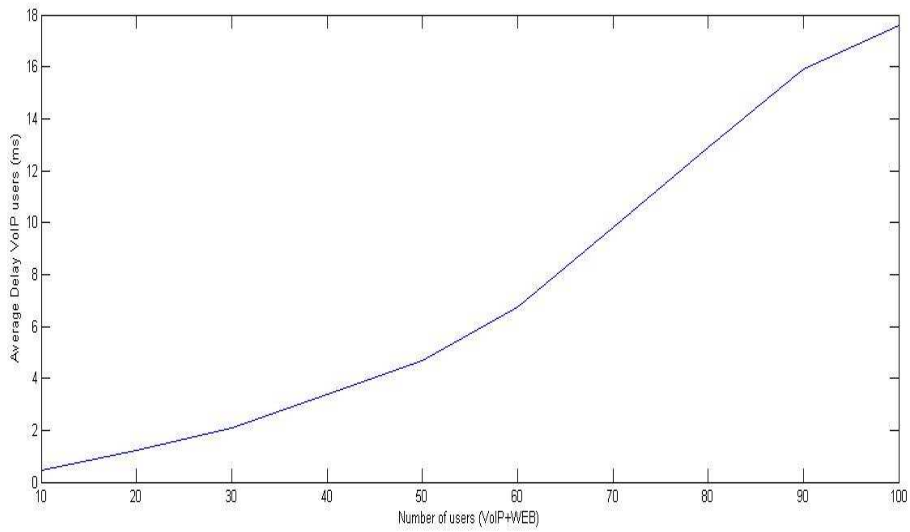


Figure 4-16.Average delay VoIP users

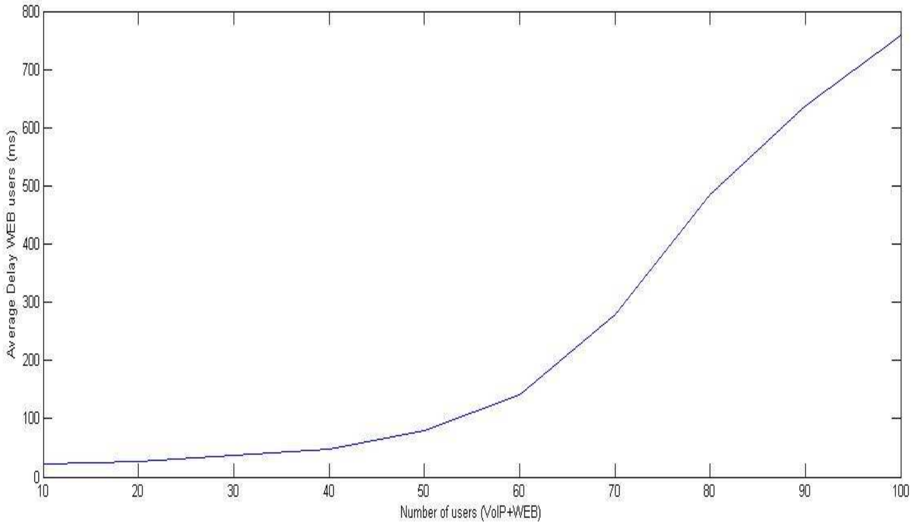


Figure 4-17. Average Delay web users

As it can be seen, the average delay for VoIP and web users is below their respective threshold.

The same way it has been done for the cases of only VoIP users in the cell, the percentage of VoIP users with a percentage of bits received above the threshold of 80ms higher than 2% and 5% has been calculated. If 'p' is defined as the percentage of bits received above the threshold, the following figures show the average number of users with p≥2% and p≥5% as the number of users in the cell is incremented.

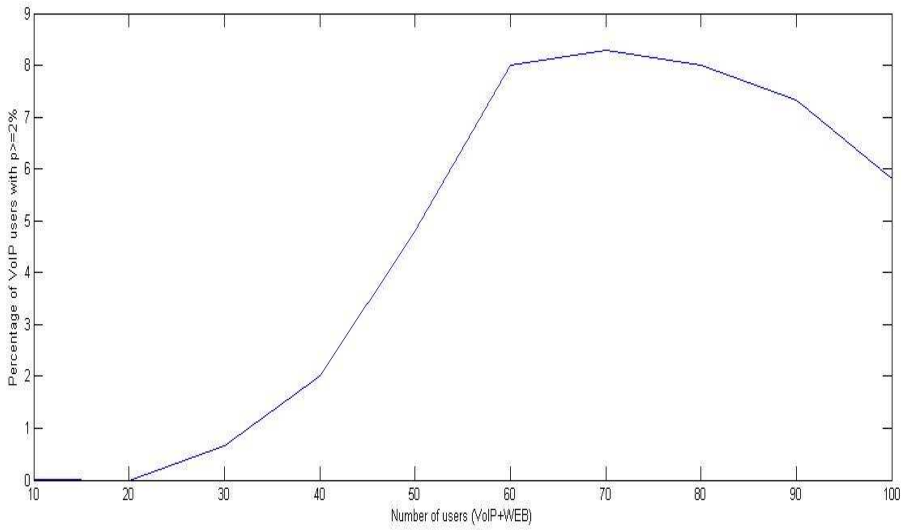


Figure 4-18.Percentage of VoIP users with p≥2%

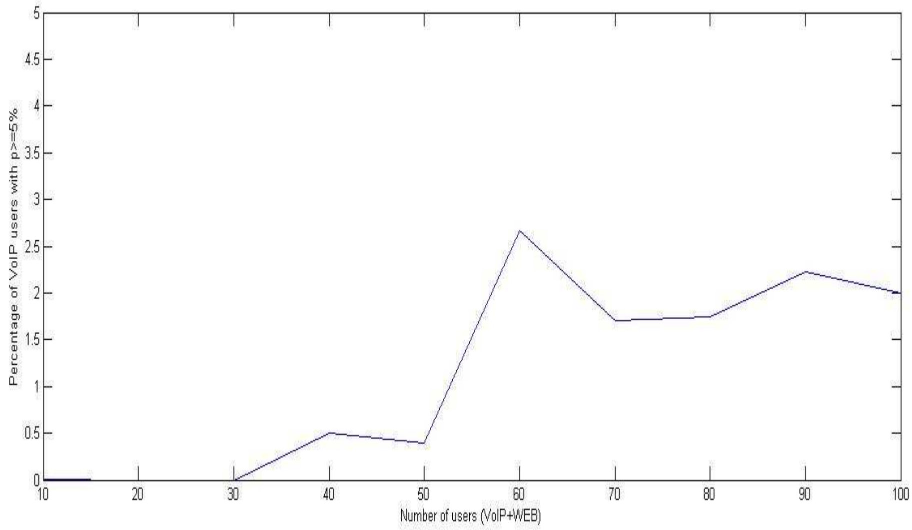


Figure 4-19.Percentage of VoIP users with p≥5%

As it can be seen in these figures the percentage of VoIP users receiving bits with more delay than the threshold is higher than in the case of only VoIP users in the cell.

When 100 users are in the cell (50 VoIP and 50 web users) 5.8% of the VoIP users are receiving more than 2% of their bits with a delay above the threshold and 2% are receiving more that 5% of the bits with more delay that the permitted.

It can be observed, that from the point of 70 users in the cell, there is a small improvement in the QoS, this is due to the decrease in the average throughput of the web browsing users.

The tables corresponding to the calculations of the percentage of VoIP users for the different p values can be found on the appendix 6.6.

For the web users, the same analysis than for the case of only web users in the cell is done, calculating ‘pw’, which is defined as the percentage of websites with a total website delay higher than the threshold of 2 seconds. The average number of users with pw≥5%, pw≥10% and pw≥20% for different number of users in the cell is shown in the following figures:

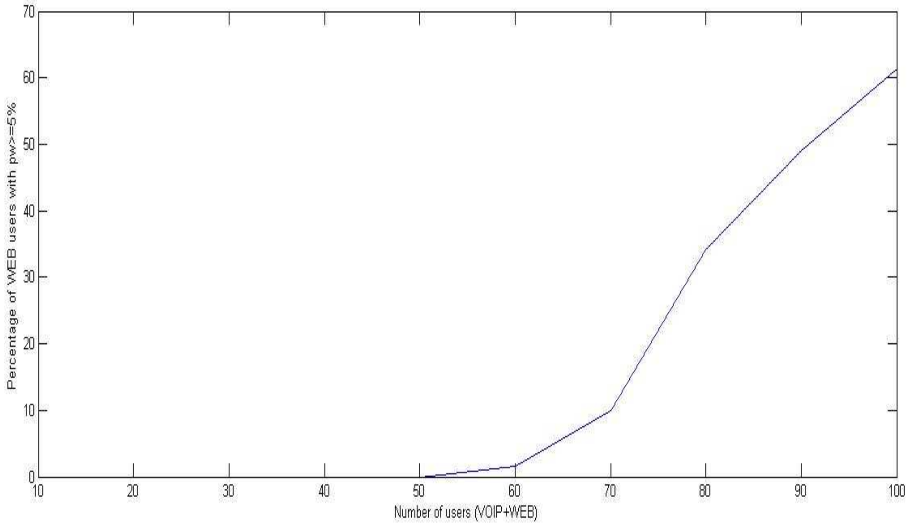


Figure 4-20. Percentage of web users with pw≥5%

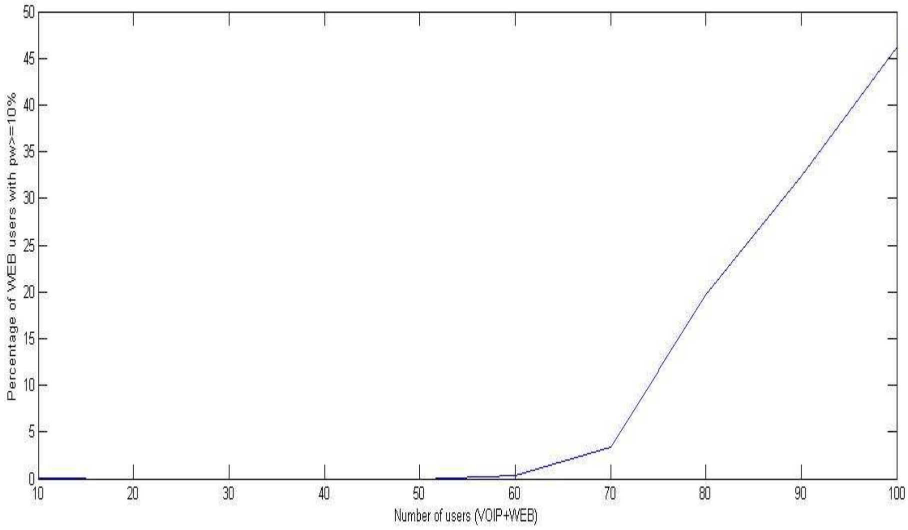


Figure 4-21. Percentage of web users with pw≥10%

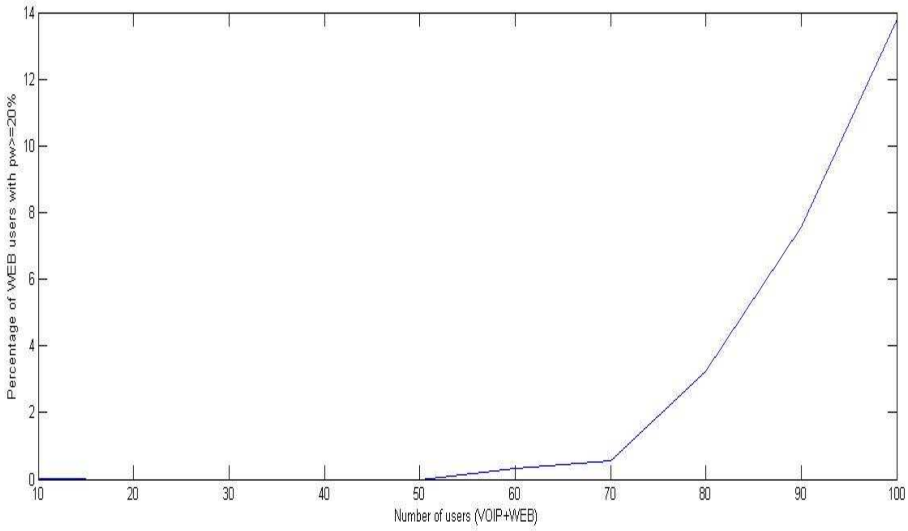


Figure 4-12. Percentage of web users with pw≥20%

As it can be seen in these figures the percentage of web users receiving websites with a delay higher than the threshold of 2s increments as the number of users in the cell is incremented. The percentages are high having for the case of 100 users in the cell 61.4% of the web users with more than 5% of their websites with a delay higher than the threshold, 46.2% receiving 10% or more of the websites with a delay higher than the threshold and 13.8% with pw≥20%.

The tables showing the calculations of these percentages can be seen in Appendix 6.6.

If we consider as unacceptable for a real time service such as VoIP, having more than 5% of the users with a p≥2% and for a web browsing service having more than 5% of the users with pw≥5%, we could realize that in a mixed scenario the VoIP traffic is a more sensitive application than the WEB traffic. Therefore in order to keep a good quality of service for both services the maximum number of simultaneous users shall be maintained below or equal to 50. This reduced simultaneous number of users is due to the fact that the Round Robin scheduling algorithm manages both services in the same way, despite that the VoIP service has higher time restrictions than the WEB service. Obviously other scheduling strategies that assign more scheduling priority to the VoIP users (or any other type of real time traffic) could improve these performances.

5 Conclusions

In this last chapter it will be exposed the final conclusions of the project and some future lines will be described as well.

5.1 Project conclusions

The objective of this project has been twofold:

1. To take advantage of my industrial placement in the department of Access Network in Orange UK, where realistic CQI (Channel Quality Information) statistical models where extracted.

2. By using the CQI statistical models, to design and implement a Matlab program that simulates VoIP and web browsing traffic over a HSDPA cell. This program has been used to evaluate the system performances in terms of throughput and delay of the bits received by the users.

Simulations have been run for different load conditions, that is for different number of users on the cell and different traffic profiles: only VoIP users in the cell, only web users in the cell and mixed traffic with half of the users VoIP and the other half web.

From the obtained results the following conclusions can be extracted:

- It has been seen shown that the average delay increments as the number of users is incremented, the reason for this to happen is that the more number users in the cell the more they have to wait be scheduled.

- The cell throughput does not arrive to its maximum for any of the simulations, not even for the case when a hundred users are in the cell, this can be seen looking at the evolution of the total cell throughput as the number of users increases, if the maximum is achieved, from that point its value would be stable, and will not increment any more as the number of users is incremented.

- For both single service and mixed service scenarios, the average user throughput for the VoIP users stays stable for all the simulations as the maximum cell throughput is not achieved. The average throughput for the web browsing users decreases with the number of users even though the maximum cell throughput is not achieved. This happens because web users ask for a new website when the transmission of the previous website is over and a reading time of 2 seconds in average has expired. As the number of users is incremented, the time between websites increments, as the reading time does not start until all the bits from the website have been received. Then the number of websites that a user downloads during the simulation time decreases as the delay increases.

In this project, a delay threshold of 80 ms for the VoIP traffic and a total website delay threshold of 2 seconds for the web browsing traffic have been considered.

- If we considerer as unacceptable for VoIP to have more than 5% of the users with p≥2% and for web browsing to have more than 5% of the users with a pw≥5%, the maximum number of simultaneous users for each situation are:

  - When only VoIP users are in the cell, more than 100 simultaneous users can be accepted.
  - When only web users are in the cell, the maximum number of simultaneous users in the cell is 80.
  - When a mixed traffic scenario is considered, the maximum number of simultaneous users in the cell is 50. Notice that this result could be improved using other schedulers that give priority to real time services such as VoIP.

5.2 Future work lines

There are many enhancements that can be done to this simulator to make it more realistic. Some examples of these are:

- R99 users could be added to the cell, this would make the simulator more realistic as this situation is very common in live network HSDPA cells.

- Instead of static simulations, movement could be added to the users.

- Neighbouring cells could be modelled to have a better approximation of the inter-cell interference, and also users could move from one cell to another.

- Different schedulers could be developed, for example the Weighted Proportional Fair, which shows good results in mixed traffic environments.

- Limitations on the Iub link capacity could be considered, as this is a common bottleneck in the operator’s network.

- New functionalities from HSDPA release 7 could be added, such ad 64 QAM and MIMO.

As this simulator has been designed in modules, it is very easy to add new functionalities to it.

6 Appendices

6.1 CQI Tables

- UE Categories from 1 to 6:

CQI value	Transport Block Size	Number of HS-PDSCH	Modulation	Reference power adjustment Δ	N_{IR}	X_{RV}
0	N/A	Out of range				
1	137	1	QPSK	0	9600	0
2	173	1	QPSK	0		
3	233	1	QPSK	0		
4	317	1	QPSK	0		
5	377	1	QPSK	0		
6	461	1	QPSK	0		
7	650	2	QPSK	0		
8	792	2	QPSK	0		
9	931	2	QPSK	0		
10	1262	3	QPSK	0		
11	1483	3	QPSK	0		
12	1742	3	QPSK	0		
13	2279	4	QPSK	0		
14	2583	4	QPSK	0		
15	3319	5	QPSK	0		
16	3565	5	16-QAM	0		
17	4189	5	16-QAM	0		
18	4664	5	16-QAM	0		
19	5287	5	16-QAM	0		
20	5887	5	16-QAM	0		
21	6554	5	16-QAM	0		
22	7168	5	16-QAM	0		
23	7168	5	16-QAM	-1		
24	7168	5	16-QAM	-2		
25	7168	5	16-QAM	-3		
26	7168	5	16-QAM	-4		
27	7168	5	16-QAM	-5		
28	7168	5	16-QAM	-6		
29	7168	5	16-QAM	-7		
30	7168	5	16-QAM	-8		

- UE Categories 7 and 8:

CQI value	Transport Block Size	Number of HS-PDSCH	Modulation	Reference power adjustment Δ	N_{IR}	X_{RV}
0	N/A	Out of range				
1	137	1	QPSK	0	19200	0
2	173	1	QPSK	0		
3	233	1	QPSK	0		
4	317	1	QPSK	0		
5	377	1	QPSK	0		
6	461	1	QPSK	0		
7	650	2	QPSK	0		
8	792	2	QPSK	0		
9	931	2	QPSK	0		
10	1262	3	QPSK	0		
11	1483	3	QPSK	0		
12	1742	3	QPSK	0		
13	2279	4	QPSK	0		
14	2583	4	QPSK	0		
15	3319	5	QPSK	0		
16	3565	5	16-QAM	0		
17	4189	5	16-QAM	0		
18	4664	5	16-QAM	0		
19	5287	5	16-QAM	0		
20	5887	5	16-QAM	0		
21	6554	5	16-QAM	0		
22	7168	5	16-QAM	0		
23	9719	7	16-QAM	0		
24	11418	8	16-QAM	0		
25	14411	10	16-QAM	0		
26	14411	10	16-QAM	-1		
27	14411	10	16-QAM	-2		
28	14411	10	16-QAM	-3		
29	14411	10	16-QAM	-4		
30	14411	10	16-QAM	-5		

- UE Category 9:

CQI value	Transport Block Size	Number of HS-PDSCH	Modulation	Reference power adjustment Δ	N_{IR}	X_{RV}
0	N/A	Out of range				
1	137	1	QPSK	0	28800	0
2	173	1	QPSK	0		
3	233	1	QPSK	0		
4	317	1	QPSK	0		
5	377	1	QPSK	0		
6	461	1	QPSK	0		
7	650	2	QPSK	0		
8	792	2	QPSK	0		
9	931	2	QPSK	0		
10	1262	3	QPSK	0		
11	1483	3	QPSK	0		
12	1742	3	QPSK	0		
13	2279	4	QPSK	0		
14	2583	4	QPSK	0		
15	3319	5	QPSK	0		
16	3565	5	16-QAM	0		
17	4189	5	16-QAM	0		
18	4664	5	16-QAM	0		
19	5287	5	16-QAM	0		
20	5887	5	16-QAM	0		
21	6554	5	16-QAM	0		
22	7168	5	16-QAM	0		
23	9719	7	16-QAM	0		
24	11418	8	16-QAM	0		
25	14411	10	16-QAM	0		
26	17237	12	16-QAM	0		
27	17237	12	16-QAM	-1		
28	17237	12	16-QAM	-2		
29	17237	12	16-QAM	-3		
30	17237	12	16-QAM	-4		

- UE Category 10:

CQI value	Transport Block Size	Number of HS-PDSCH	Modulation	Reference power adjustment Δ	N_{IR}	X_{RV}
0	N/A	Out of range				
1	137	1	QPSK	0	28800	0
2	173	1	QPSK	0		
3	233	1	QPSK	0		
4	317	1	QPSK	0		
5	377	1	QPSK	0		
6	461	1	QPSK	0		
7	650	2	QPSK	0		
8	792	2	QPSK	0		
9	931	2	QPSK	0		
10	1262	3	QPSK	0		
11	1483	3	QPSK	0		
12	1742	3	QPSK	0		
13	2279	4	QPSK	0		
14	2583	4	QPSK	0		
15	3319	5	QPSK	0		
16	3565	5	16-QAM	0		
17	4189	5	16-QAM	0		
18	4664	5	16-QAM	0		
19	5287	5	16-QAM	0		
20	5887	5	16-QAM	0		
21	6554	5	16-QAM	0		
22	7168	5	16-QAM	0		
23	9719	7	16-QAM	0		
24	11418	8	16-QAM	0		
25	14411	10	16-QAM	0		
26	17237	12	16-QAM	0		
27	21754	15	16-QAM	0		
28	23370	15	16-QAM	0		
29	24222	15	16-QAM	0		
30	25558	15	16-QAM	0		

- UE Categories 11 and 12:

CQI value	Transport Block Size	Number of HS-PDSCH	Modulation	Reference power adjustment Δ	N_{IR}	X_{RV}
0	N/A	Out of range				
1	137	1	QPSK	0	4800	0
2	173	1	QPSK	0		
3	233	1	QPSK	0		
4	317	1	QPSK	0		
5	377	1	QPSK	0		
6	461	1	QPSK	0		
7	650	2	QPSK	0		
8	792	2	QPSK	0		
9	931	2	QPSK	0		
10	1262	3	QPSK	0		
11	1483	3	QPSK	0		
12	1742	3	QPSK	0		
13	2279	4	QPSK	0		
14	2583	4	QPSK	0		
15	3319	5	QPSK	0		
16	3319	5	QPSK	-1		
17	3319	5	QPSK	-2		
18	3319	5	QPSK	-3		
19	3319	5	QPSK	-4		
20	3319	5	QPSK	-5		
21	3319	5	QPSK	-6		
22	3319	5	QPSK	-7		
23	3319	5	QPSK	-8		
24	3319	5	QPSK	-9		
25	3319	5	QPSK	-10		
26	3319	5	QPSK	-11		
27	3319	5	QPSK	-12		
28	3319	5	QPSK	-13		
29	3319	5	QPSK	-14		
30	3319	5	QPSK	-15		

6.2 Probability Distributions

In this appendix the exponential, geometric and Pareto distributions are explained in order to model the VoIP and web traffic.

6.2.1 Exponential distribution

The probability density function (pdf) of an exponential distribution is

$$pdf = \lambda \cdot e^{-\lambda \cdot x} \text{ for } x \geq 0$$

Where $\lambda > 0$ is the parameter of the distribution. The distribution is supported on the interval $[0, \infty)$.

The cumulative distribution function is given by

$$cdf = 1 - e^{-\lambda \cdot x} \text{ for } x \geq 0$$

The mean and the variance of an exponential distribution are:

$$Mean = \frac{1}{\lambda} \qquad Variance = \frac{1}{\lambda^2}$$

6.2.2 Geometric distribution

The probability mass function of a geometric distribution is defined as:

$$pmf = (1-p)^{k-1} \cdot p$$

Where $0<p\leq 1$ and k=1,2,3…

The cumulative distribution function is:

$$cdf = 1 - (1-p)^k$$

The mean and the variance of a geometric distribution are:

$$Mean = \frac{1}{p} \qquad Variance = \frac{1-p}{p^2}$$

6.2.3 Pareto Distribution

A Pareto distribution is often defined as Pareto[α,xm], where α is the shape parameter, and xm is the minimum possible value for x.

The probability density function of a Pareto distribution is:

$$pdf = \frac{\alpha \cdot xm^{\alpha}}{x^{\alpha+1}} \text{ for } x \geq xm$$

The cumulative distribution function is:

$$cdf = 1 - \left(\frac{xm}{x}\right)^{\alpha} \text{ for } x \geq xm$$

The mean and the variance are defined as:

$$Mean = \frac{\alpha \cdot xm}{\alpha - 1} \qquad Variance = \frac{xm^2 \cdot \alpha}{(\alpha - 1)^2 \cdot (\alpha - 2)}$$

When α is smaller or equal to 2, the value of the variance is infinite.

6.3 CQI Distributions

The following figures show the relationship between the EcIo, the RSCP and the CQI. These represent the values on the input tables aCQI0…aCQI30. Notice how in general, the higher the EcIo and the RSCP are the higher the CQI, this is because the CQI represents the radio conditions.

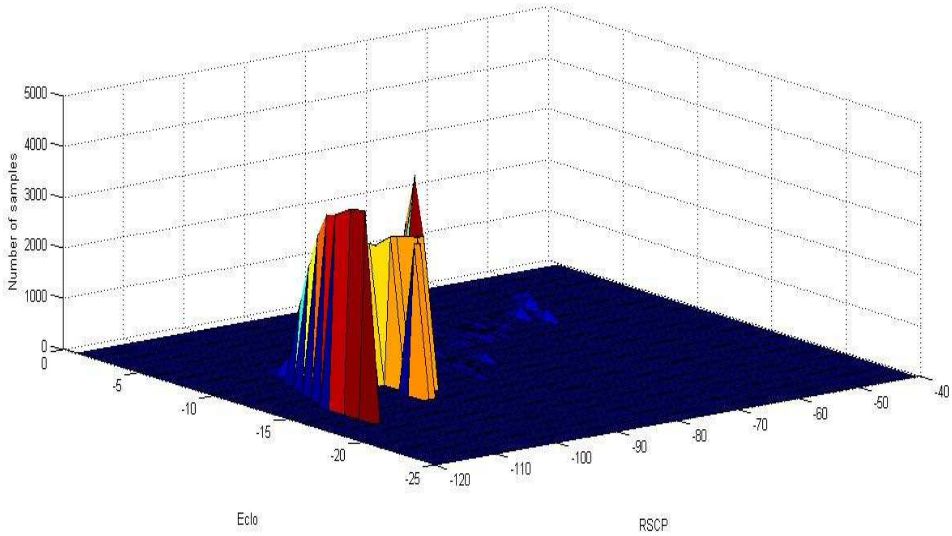


Figure 6-1. CQI0 Distribution

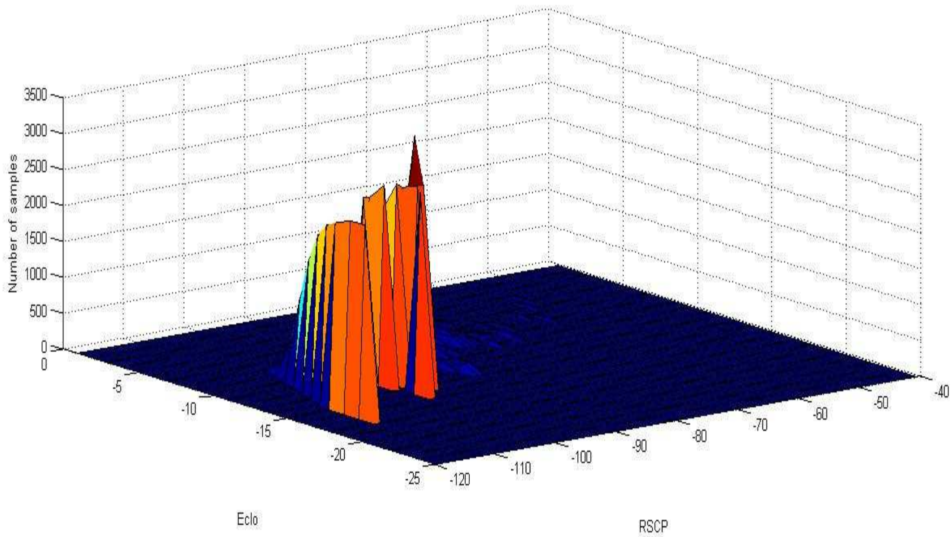


Figure 6-2. CQI1 Distribution

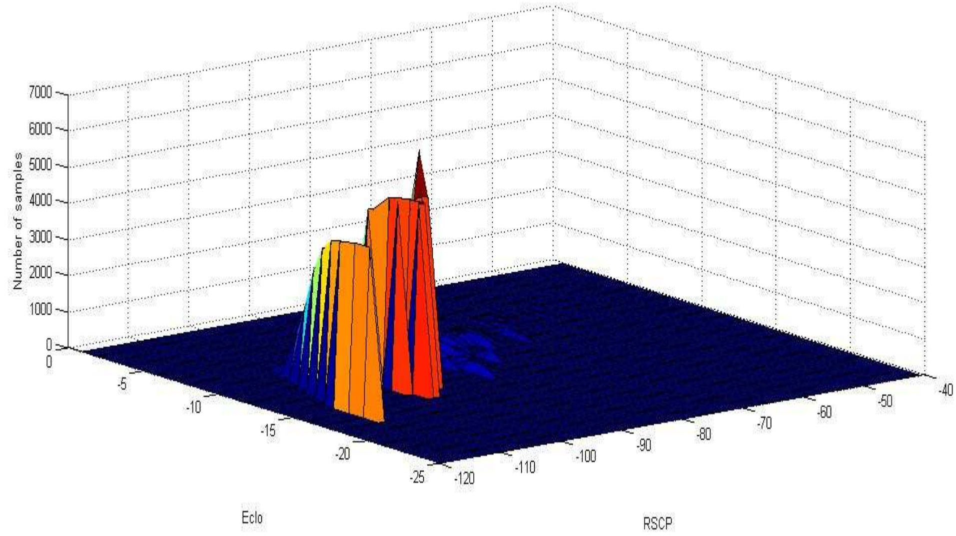


Figure 6-3. CQI2 Distribution

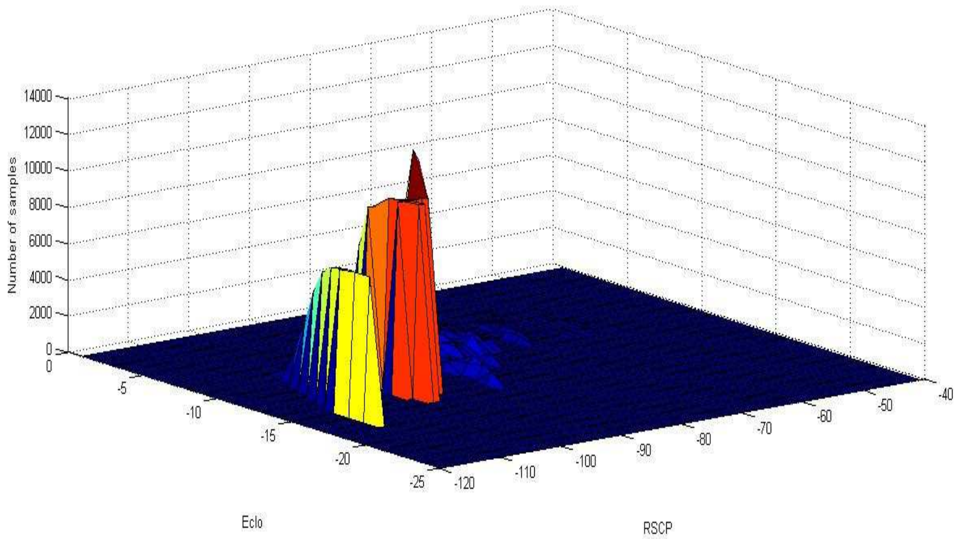


Figure 6-4. CQI3 Distribution

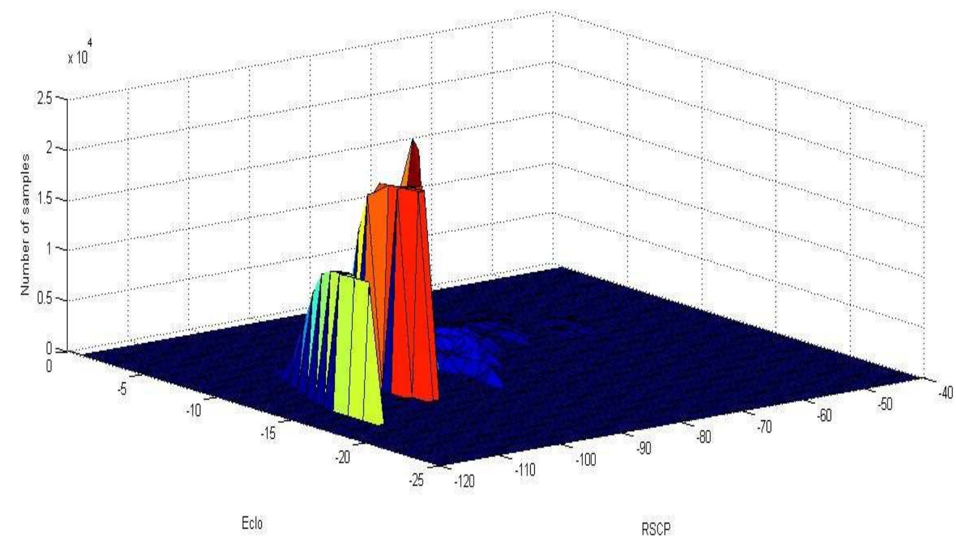


Figure 6-5. CQI4 Distribution

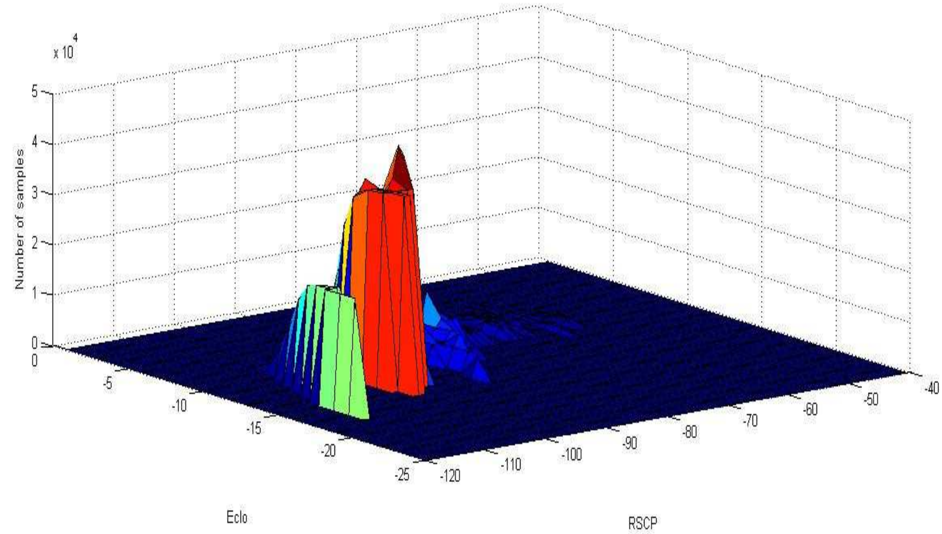


Figure 6-6. CQI5 Distribution

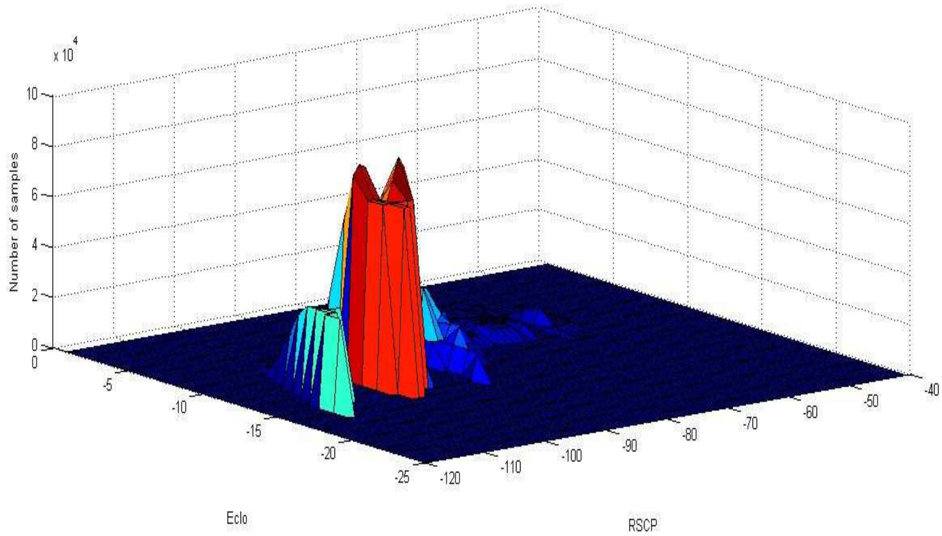


Figure 6-7. CQI6 Distribution

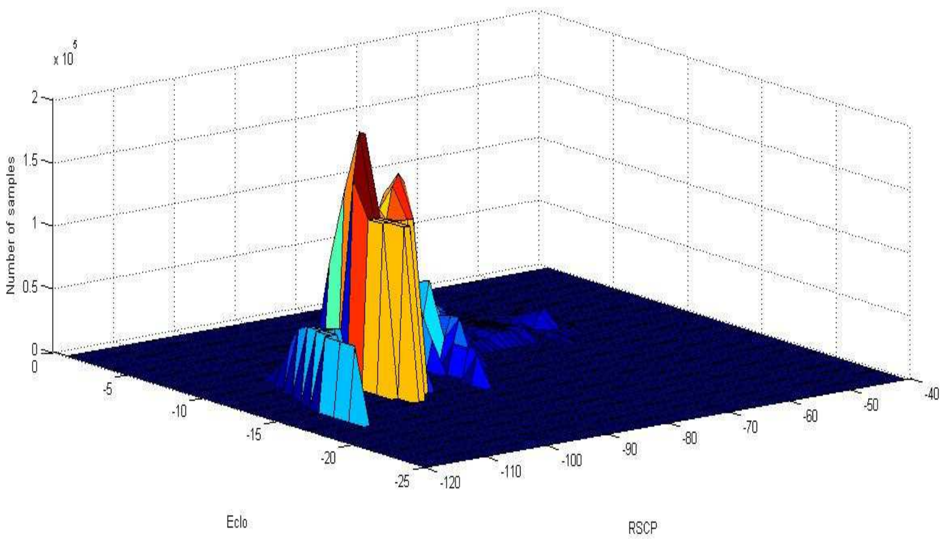


Figure 6-8. CQI7 Distribution

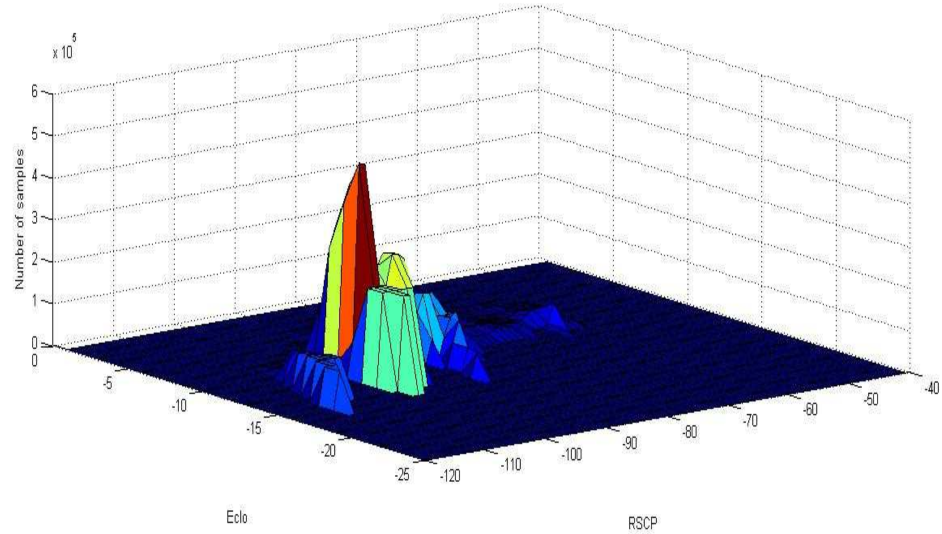


Figure 6-9. CQI8 Distribution

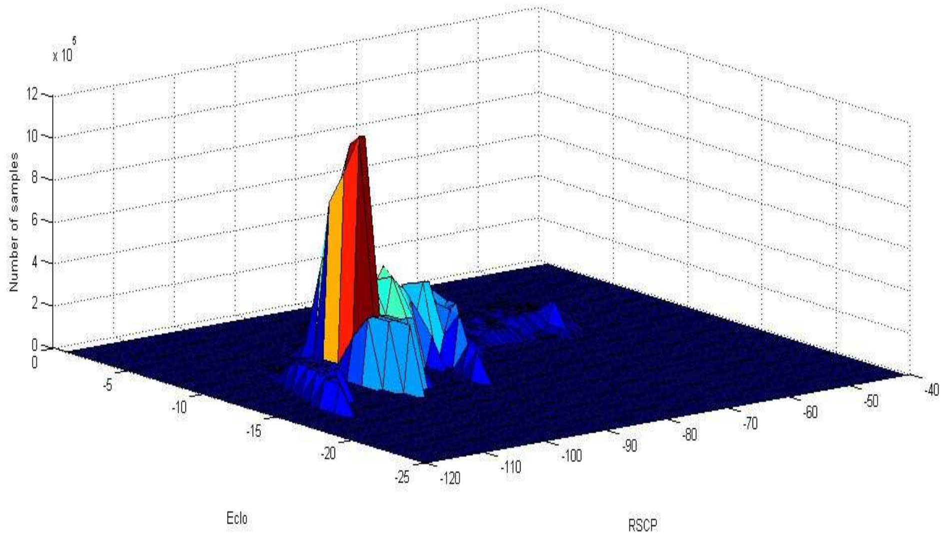


Figure 6-10. CQI9 Distribution

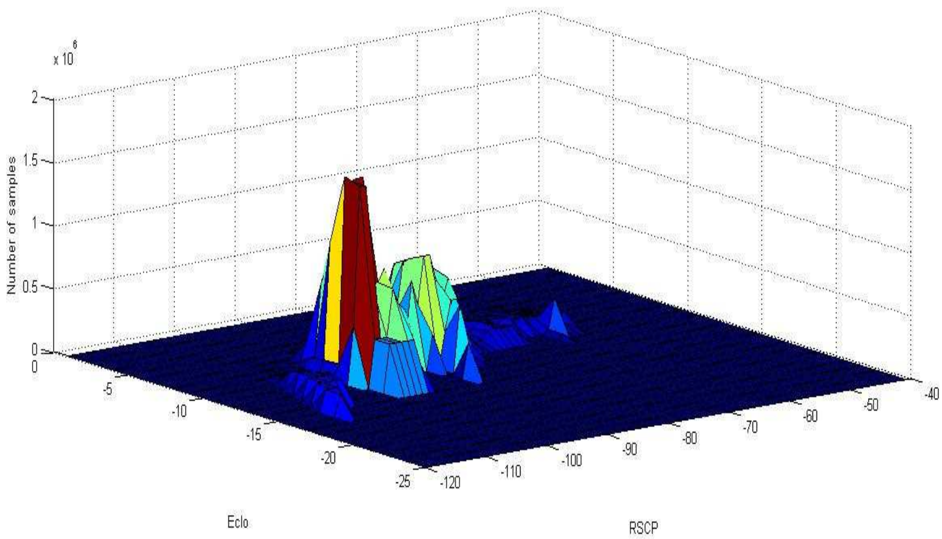


Figure 6-11. CQI10 Distribution

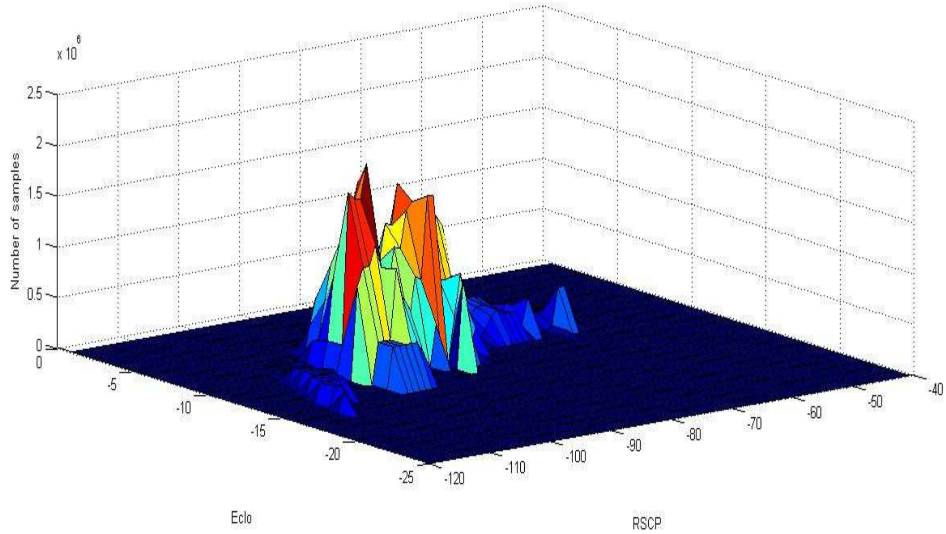


Figure 6-12. CQI11 Distribution

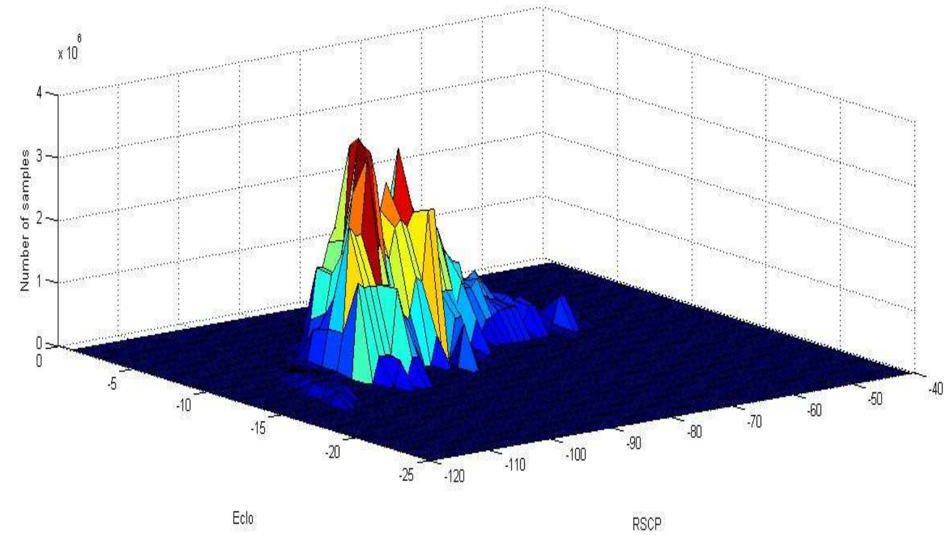


Figure 6-13. CQI12 Distribution

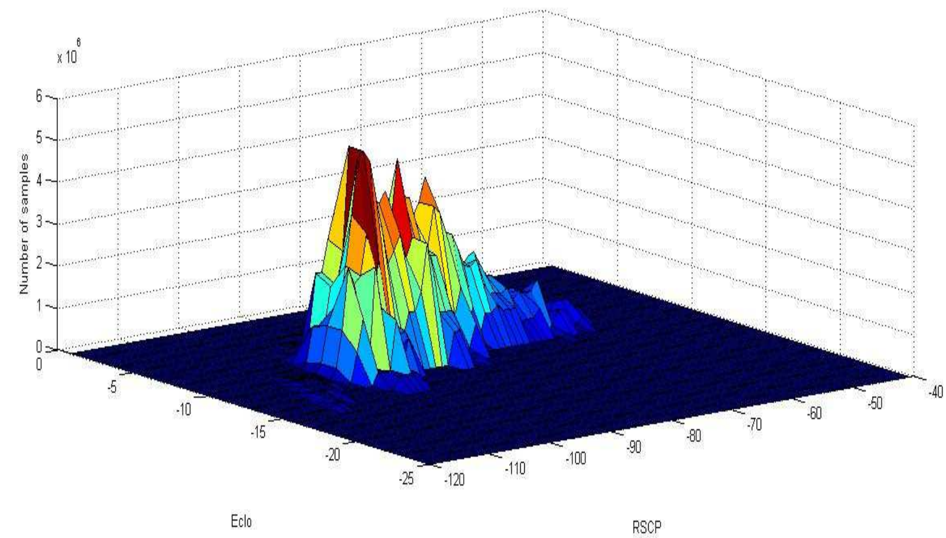


Figure 6-14. CQI13 Distribution

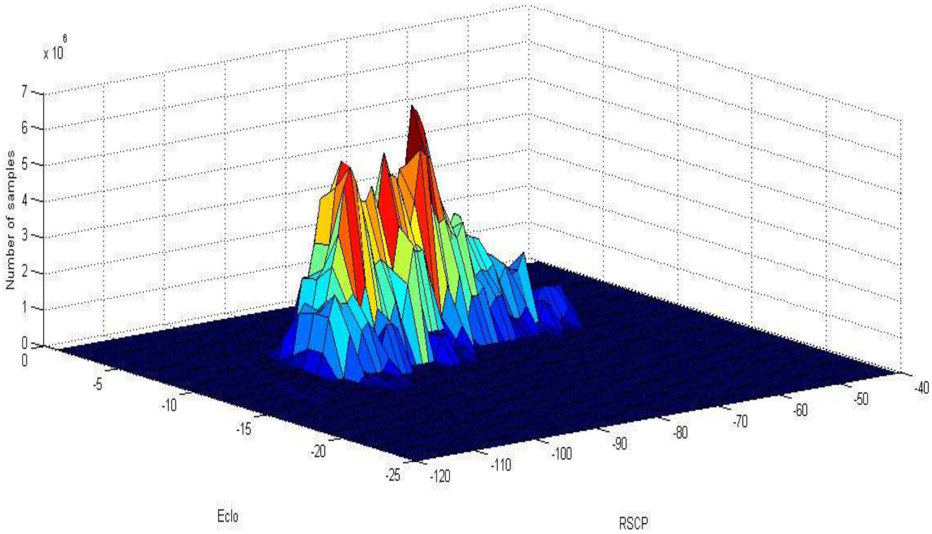


Figure 6-15. CQI14 Distribution

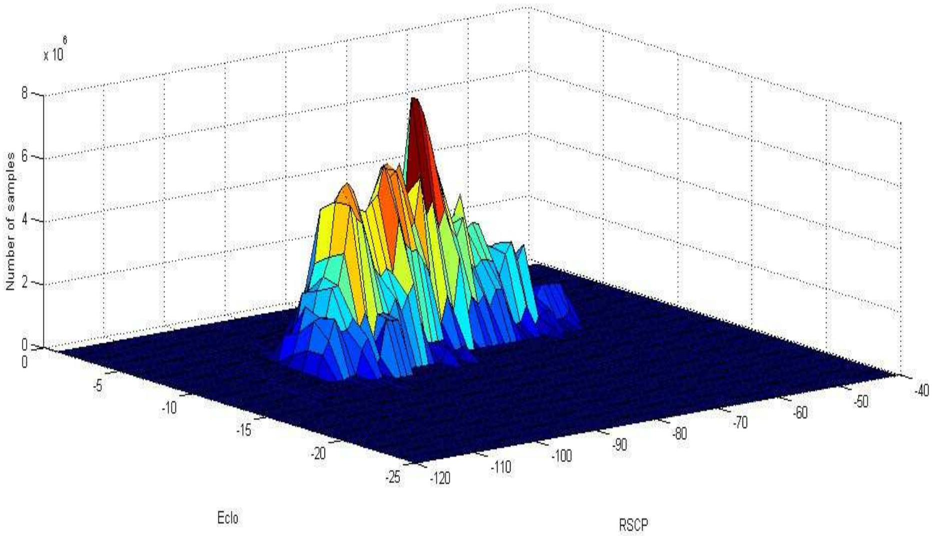


Figure 6-16. CQI15 Distribution

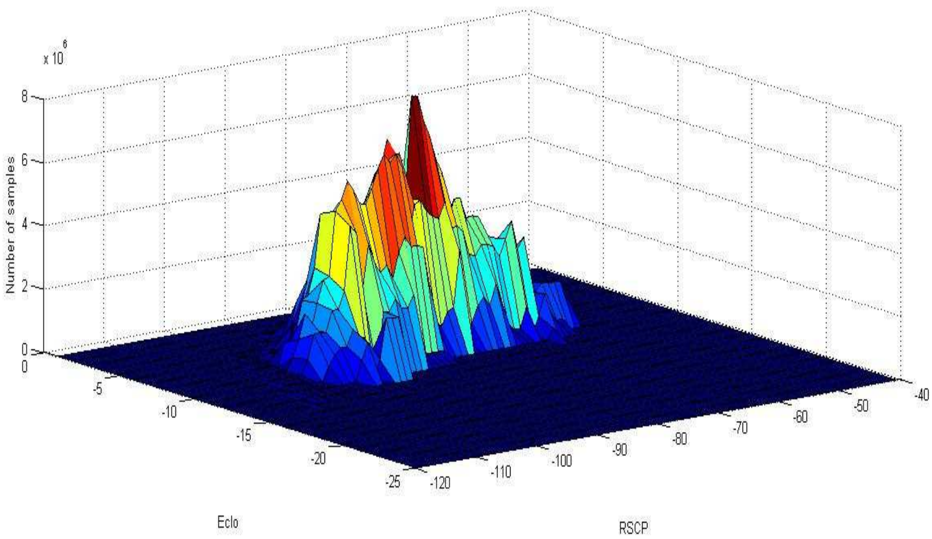


Figure 6-17. CQI16 Distribution

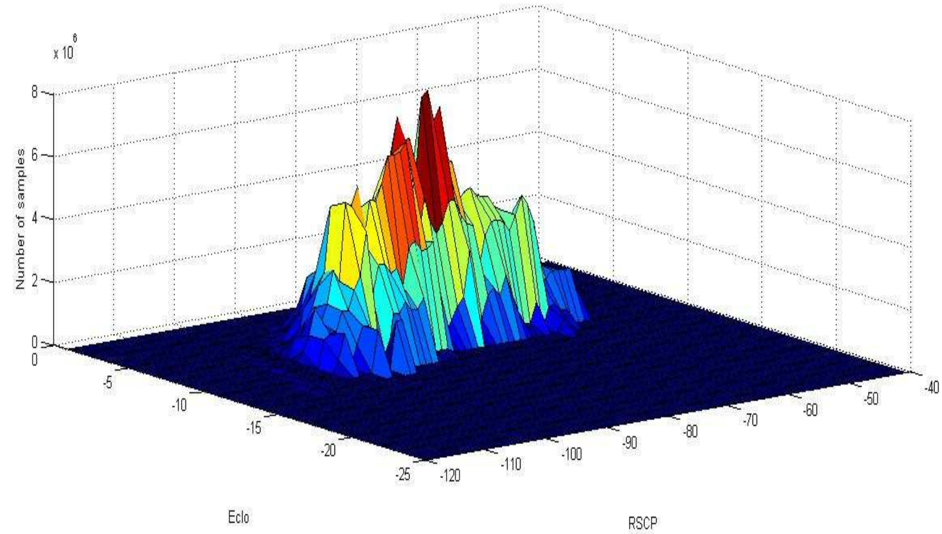


Figure 6-18. CQI17 Distribution

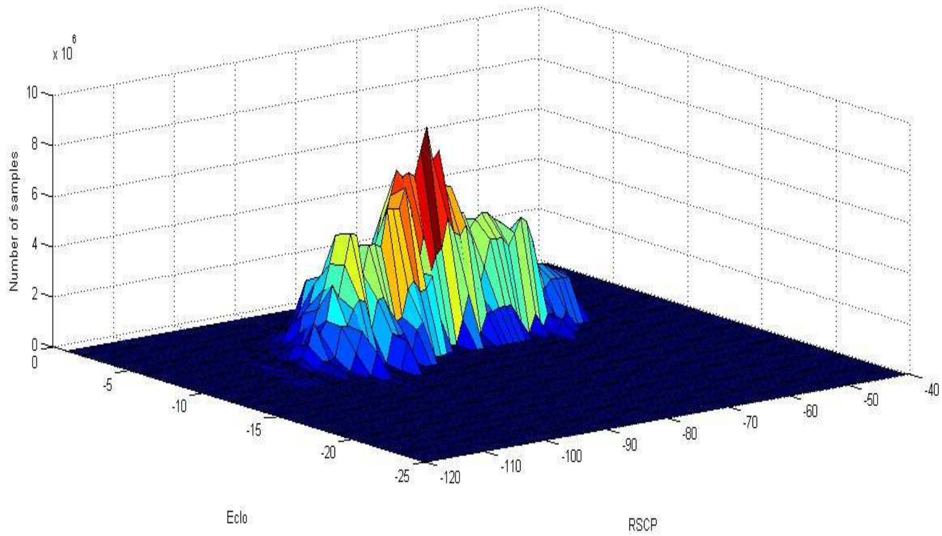


Figure 6-19. CQI18 Distribution

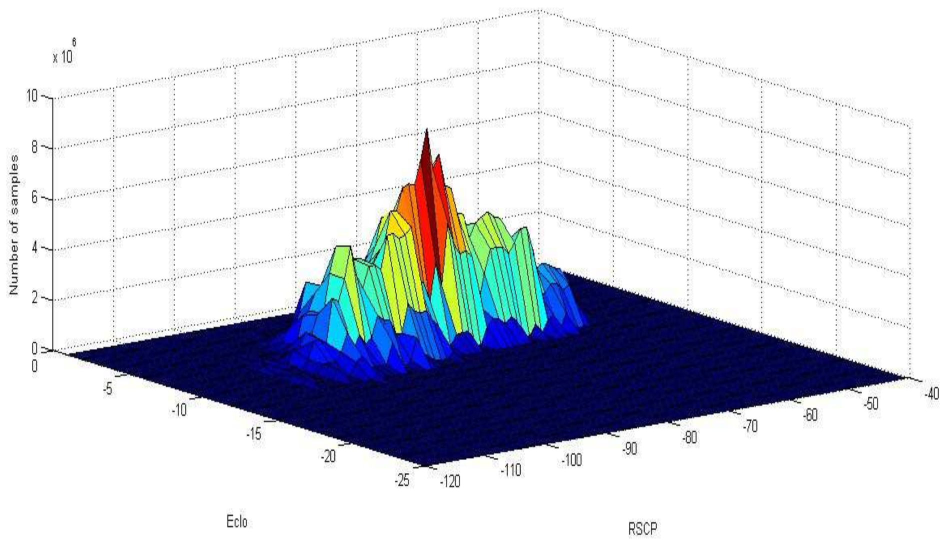


Figure 6-20. CQI19 Distribution

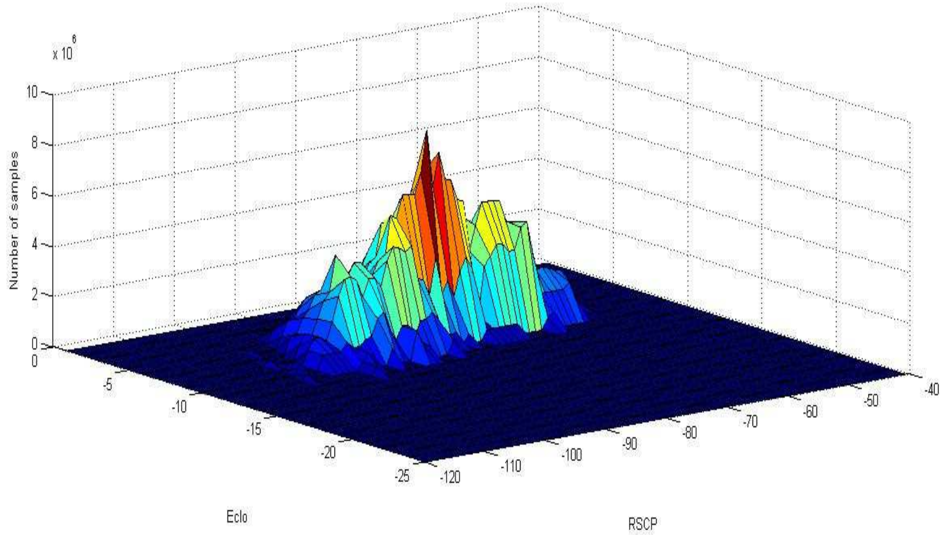


Figure 6-21. CQI20 Distribution

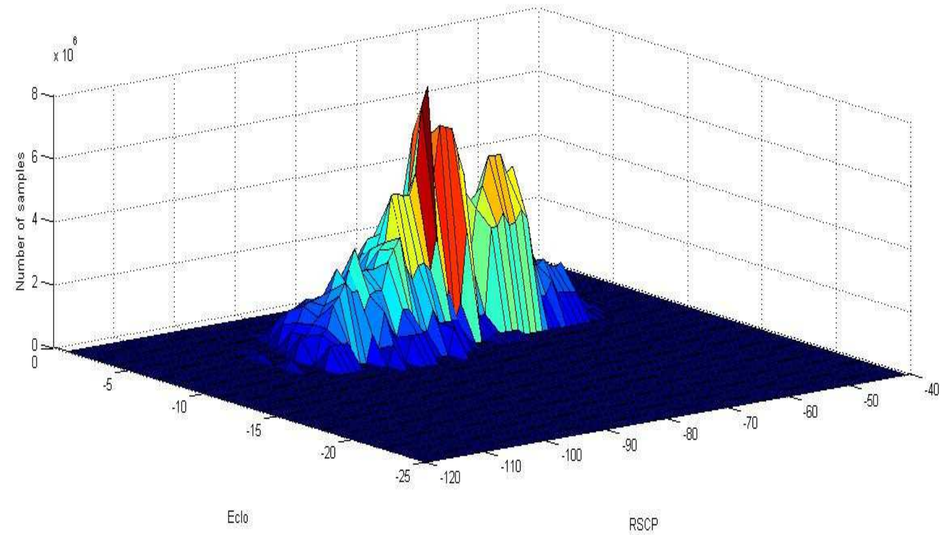


Figure 6-22. CQI21 Distribution

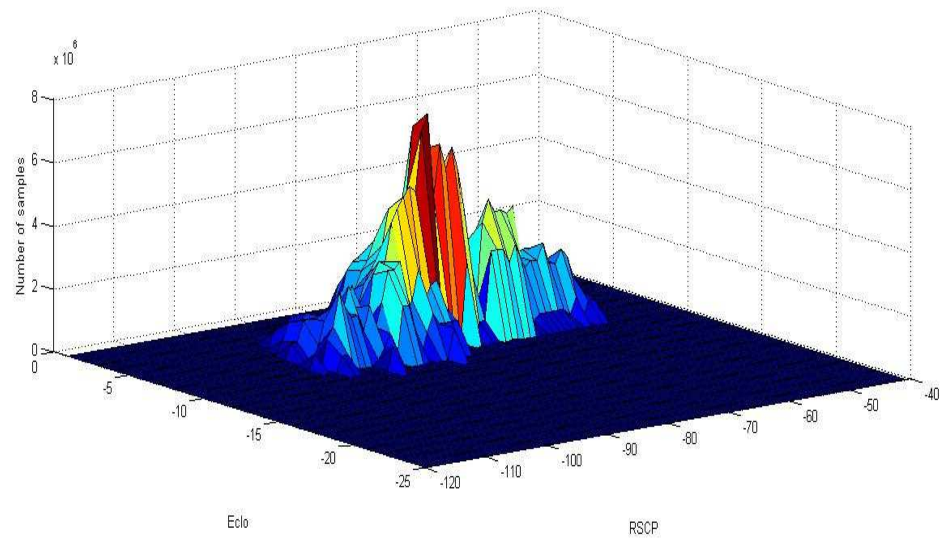


Figure 6-23. CQI22 Distribution

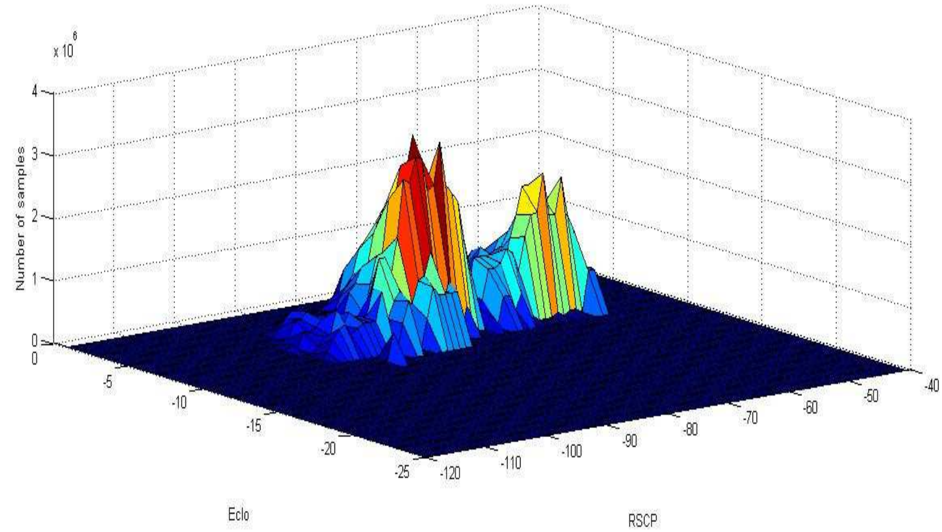


Figure 6-24. CQI23 Distribution

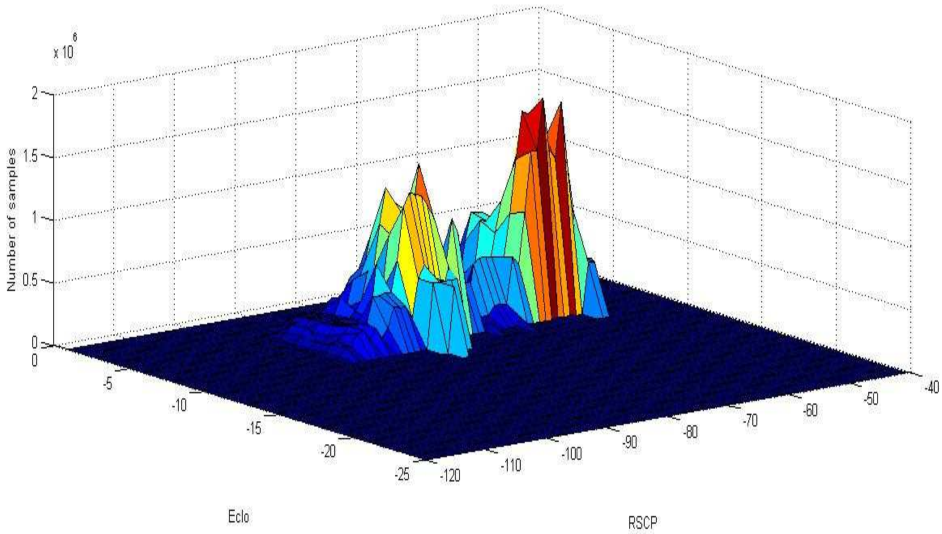


Figure 6-25. CQI24 Distribution

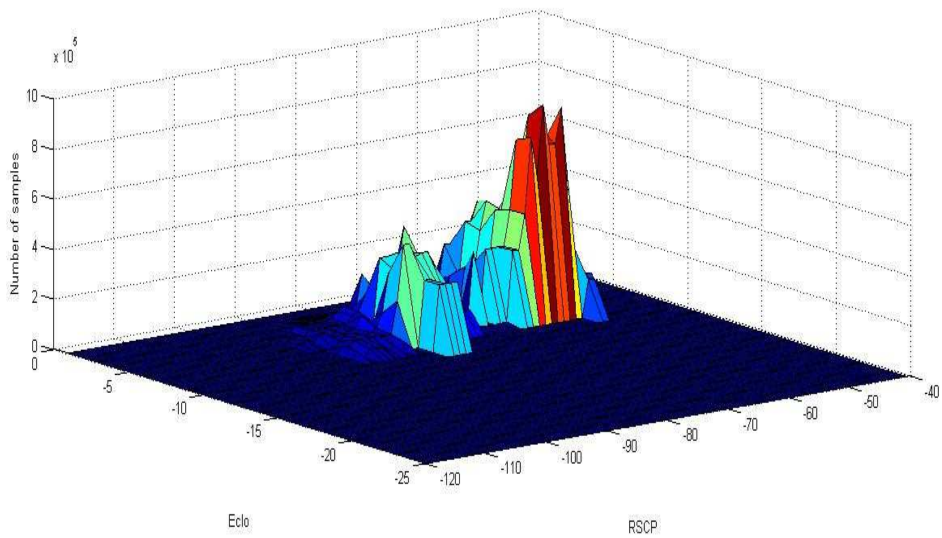


Figure 6-26. CQI25 Distribution

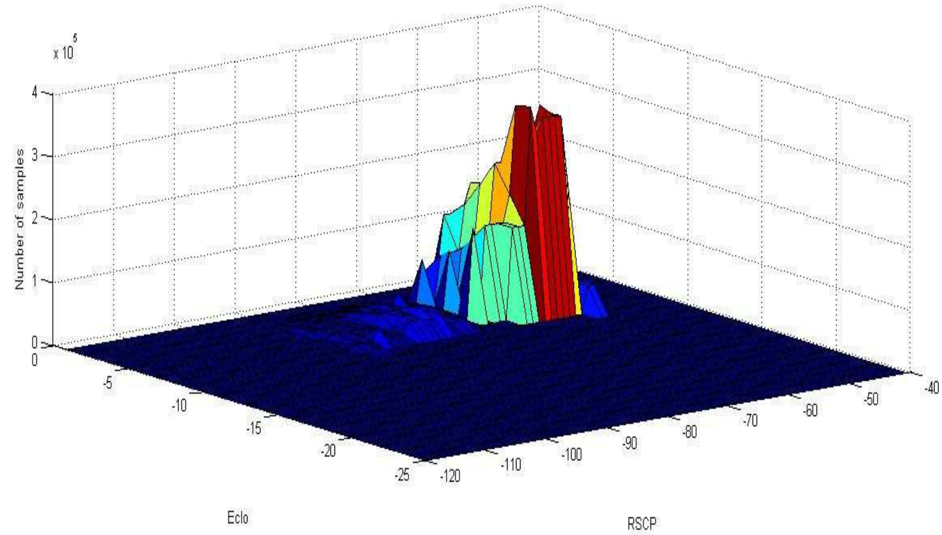


Figure 6-27. CQI26 Distribution

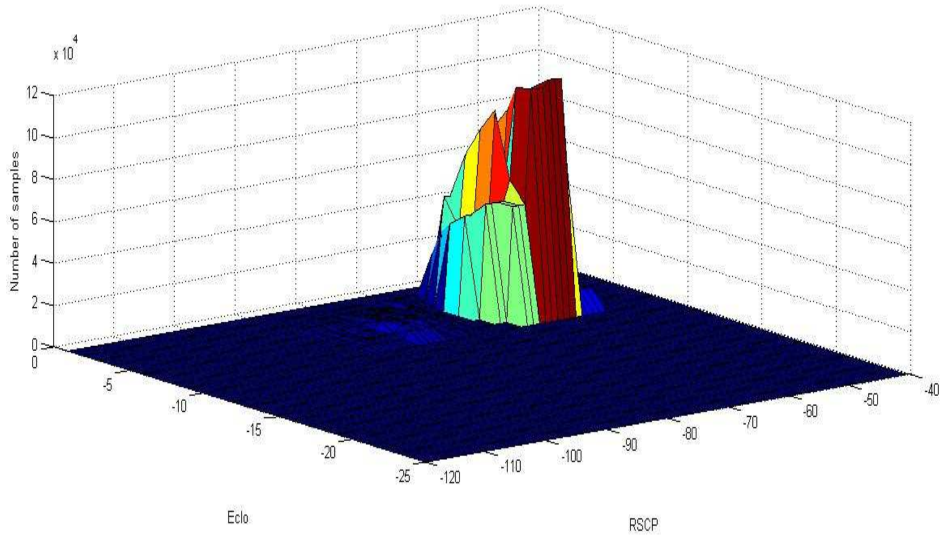


Figure 6-28. CQI27 Distribution

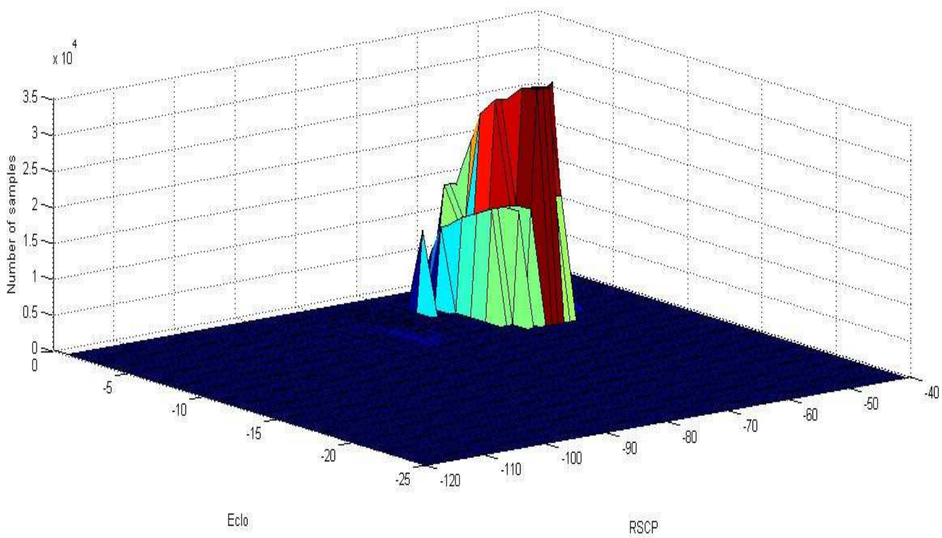


Figure 6-29. CQI28 Distribution

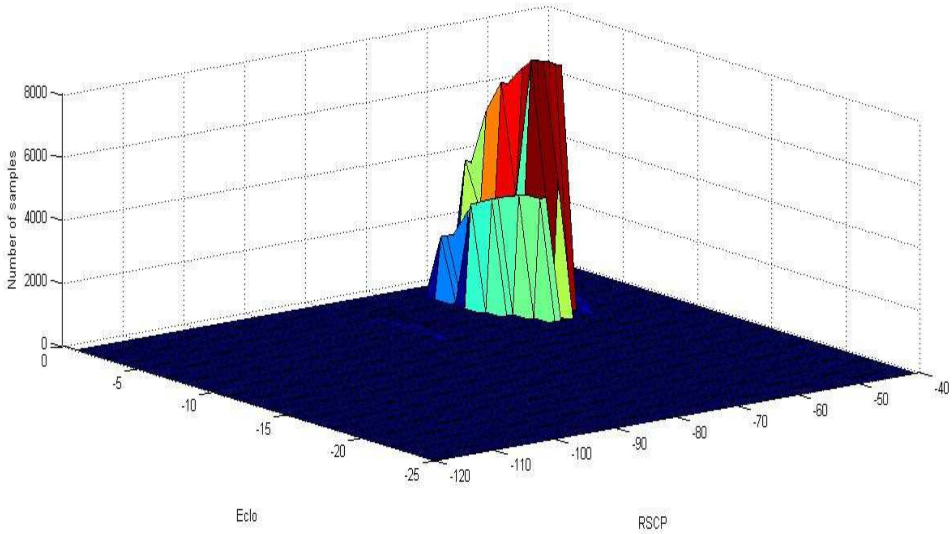


Figure 6-30. CQI29 Distribution

6.4 Percentage of users above threshold for VoIP simulations

In this section, the tables containing the number of users, for each simulation, with 2% or more and 5% or more of bits received above the threshold of 80 ms are shown.

The variable ‘p’ is defined as the percentage of bits received above the delay threshold.

None of the simulations shown users with p≥2% until the case of 80 users in the cell, this is why only the tables for 80 to 100 users in the cell are shown:

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	0	0
Snapshot 2	0	0
Snapshot 3	0	0
Snapshot 4	0	0
Snapshot 5	0	0
Snapshot 6	0	0
Snapshot 7	0	0
Snapshot 8	0	0
Snapshot 9	0	0
Snapshot 10	1	0
Average	0.1	0
Percentage users	0.13	0

Table 6-1. Results simulations 80 VoIP users in the cell

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	0	0
Snapshot 2	0	0
Snapshot 3	0	0
Snapshot 4	0	0
Snapshot 5	0	0
Snapshot 6	0	0
Snapshot 7	0	0
Snapshot 8	0	0
Snapshot 9	1	0
Snapshot 10	1	0
Average	0.2	0
Percentage users	0.22	0

Table 6-2. Results simulations 90 VoIP users in the cell

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	0	0
Snapshot 2	0	0
Snapshot 3	0	0
Snapshot 4	1	0
Snapshot 5	0	0
Snapshot 6	0	0
Snapshot 7	0	0
Snapshot 8	0	0
Snapshot 9	1	0
Snapshot 10	1	0
Average	0.3	0
Percentage users	0.3	0

Table 6-3.Results simulations 100 VoIP users in the cell

6.5 Percentage of users above threshold for WEB simulations

In this section, the tables containing the number of users, for each simulation with 5% or more, 10% or more and 20% or more of websites received with a total delay above the threshold of 2 seconds are shown.

The variable 'pw' is defined as the percentage of websites received above the threshold.

All the websites for the cases of 10 to 50 web users in the cell, are received with a delay below the threshold, the results for the simulations with 60 to 100 users in the cell can be seen on the following tables:

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	0	0	0
Snapshot 2	0	0	0
Snapshot 3	0	0	0
Snapshot 4	0	0	0
Snapshot 5	0	0	0
Snapshot 6	0	0	0
Snapshot 7	0	0	0
Snapshot 8	0	0	0
Snapshot 9	1	1	0
Snapshot 10	0	0	0
Average	0.1	0.1	0
Percentage users	0.17	0.17	0

Table 6-4. Results simulations 60 web users in the cell

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	0	0	0
Snapshot 2	0	0	0
Snapshot 3	0	0	0
Snapshot 4	1	1	0
Snapshot 5	0	0	0
Snapshot 6	0	0	0
Snapshot 7	0	0	0
Snapshot 8	1	1	1
Snapshot 9	0	0	0
Snapshot 10	0	0	0
Average	0.2	0.2	0.1
Percentage users	0.29	0.29	0.14

Table 6-5. Results simulations 70 web users in the cell

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	0	0	0
Snapshot 2	1	0	0
Snapshot 3	0	0	0
Snapshot 4	3	0	0
Snapshot 5	1	0	0
Snapshot 6	1	0	0
Snapshot 7	3	1	0
Snapshot 8	3	0	0
Snapshot 9	0	0	0
Snapshot 10	10	4	1
Average	2.2	0.5	0.1
Percentage users	2.75	0.63	0.13

Table 6-6. Results simulations 80 web users in the cell

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	14	0	0
Snapshot 2	11	5	0
Snapshot 3	14	0	0
Snapshot 4	17	7	1
Snapshot 5	14	3	0
Snapshot 6	7	2	0
Snapshot 7	7	4	1
Snapshot 8	11	5	0
Snapshot 9	14	0	0
Snapshot 10	6	1	0
Average	11.5	2.7	0.2
Percentage users	12.78	3	0.22

Table 6-7. Results simulations 90 web users in the cell

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	34	16	0
Snapshot 2	15	6	1
Snapshot 3	34	16	0
Snapshot 4	13	8	1
Snapshot 5	36	23	3
Snapshot 6	39	18	3
Snapshot 7	29	12	2
Snapshot 8	20	5	3
Snapshot 9	34	16	0
Snapshot 10	48	28	9
Average	30.2	14.8	2.2
Percentage users	30.2	14.8	2.2

Table 6-8. Results simulations 100 web users in the cell

6.6 Percentage of users above threshold for mixed simulations

In this section, the tables containing the percentage of VoIP and web users above the delay threshold are shown.

6.6.1 VoIP Users

Until the case of 30 users in the cell, no users shown more than 2% of their bits received above the threshold, the following tables show the results for the simulations from 30 to 100 users in the cell (VoIP+WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	0	0
Snapshot 2	0	0
Snapshot 3	0	0
Snapshot 4	0	0
Snapshot 5	0	0
Snapshot 6	0	0
Snapshot 7	0	0
Snapshot 8	0	0
Snapshot 9	1	0
Snapshot 10	0	0
Average	0.1	0
Percentage users (VoIP)	0.67	0

Table 6-9.Results simulations 30 users in the cell (15 VoIP + 15 WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	0	0
Snapshot 2	1	0
Snapshot 3	1	1
Snapshot 4	0	0
Snapshot 5	1	0
Snapshot 6	0	0
Snapshot 7	0	0
Snapshot 8	1	0
Snapshot 9	0	0
Snapshot 10	0	0
Average	0.4	0.1
Percentage users (VoIP)	2	0.5

Table 6-10.Results simulations 40 users in the cell (20 VoIP + 20 WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	0	0
Snapshot 2	2	0
Snapshot 3	1	0
Snapshot 4	1	0
Snapshot 5	1	0
Snapshot 6	1	0
Snapshot 7	2	0
Snapshot 8	1	0
Snapshot 9	2	1
Snapshot 10	1	0
Average	1.2	0.1
Percentage users (VoIP)	4.8	0.4

Table 6-11.Results simulations 50 users in the cell (25 VoIP+ 25 WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	3	1
Snapshot 2	5	0
Snapshot 3	1	0
Snapshot 4	2	2
Snapshot 5	2	1
Snapshot 6	4	1
Snapshot 7	2	2
Snapshot 8	1	0
Snapshot 9	2	0
Snapshot 10	2	1
Average	2.4	0.8
Percentage users (VoIP)	8	2.67

Table 6-12. Results simulations 60 users in the cell (30 VoIP + 30 WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	3	0
Snapshot 2	6	1
Snapshot 3	0	0
Snapshot 4	4	1
Snapshot 5	1	1
Snapshot 6	3	0
Snapshot 7	4	0
Snapshot 8	2	1
Snapshot 9	4	1
Snapshot 10	2	1
Average	2.9	0.6
Percentage users (VoIP)	8.29	1.71

Table 6-13. Results simulations 70 users in the cell (35 VoIP + 35 WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	2	0
Snapshot 2	4	0
Snapshot 3	4	1
Snapshot 4	3	0
Snapshot 5	4	2
Snapshot 6	2	1
Snapshot 7	2	0
Snapshot 8	2	0
Snapshot 9	4	1
Snapshot 10	5	2
Average	3.2	0.7
Percentage users (VoIP)	8	1.75

Table 6-14. Results simulations 80 users in the cell (40 VoIP + 40 WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	3	0
Snapshot 2	4	1
Snapshot 3	3	2
Snapshot 4	2	1
Snapshot 5	2	1
Snapshot 6	2	0
Snapshot 7	5	2
Snapshot 8	5	3
Snapshot 9	4	0
Snapshot 10	3	0
Average	3.3	1
Percentage users (VoIP)	7.33	2.22

Table 6-15. Results simulations 90 users in the cell (45 VoIP + 45 WEB)

	Number of users with p≥2%	Number of users with p≥5%
Snapshot 1	2	1
Snapshot 2	3	2
Snapshot 3	2	1
Snapshot 4	4	0
Snapshot 5	5	2
Snapshot 6	1	0
Snapshot 7	3	1
Snapshot 8	2	1
Snapshot 9	5	1
Snapshot 10	2	1
Average	2.9	1
Percentage users (VoIP)	5.8	2

Table 6-16. Results simulations 100 users in the cell (50 VoIP + 50 WEB)

6.6.2 Web Users

All the websites for the cases of 10 to 50 users in the cell are received with a delay below the threshold; the results for the simulations with 60 to 100 users in the cell can be seen on the following tables:

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	0	0	0
Snapshot 2	0	0	0
Snapshot 3	0	0	0
Snapshot 4	0	0	0
Snapshot 5	0	0	0
Snapshot 6	1	0	0
Snapshot 7	1	1	1
Snapshot 8	0	0	0
Snapshot 9	1	0	0
Snapshot 10	2	0	0
Average	0.5	0.1	0.1
Percentage users (WEB)	1.67	0.33	0.33

Table 6-4. Results simulations 60 users in the cell (30 VoIP + 30 WEB)

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	1	0	0
Snapshot 2	8	4	0
Snapshot 3	4	3	0
Snapshot 4	3	0	0
Snapshot 5	0	0	0
Snapshot 6	5	1	1
Snapshot 7	0	0	0
Snapshot 8	3	2	1
Snapshot 9	2	0	0
Snapshot 10	9	2	0
Average	3.5	1.2	0.2
Percentage users (WEB)	10	3.43	0.57

Table 6-5. Results simulations 70 users in the cell (35 VoIP + 35 WEB)

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	6	1	0
Snapshot 2	17	10	2
Snapshot 3	12	8	1
Snapshot 4	12	4	0
Snapshot 5	16	11	1
Snapshot 6	21	20	5
Snapshot 7	16	8	1
Snapshot 8	12	9	2
Snapshot 9	14	2	0
Snapshot 10	11	6	1
Average	13.7	7.9	1.3
Percentage users (WEB)	34.25	19.75	3.25

Table 6-6. Results simulations 80 users in the cell (40 VoIP + 40 WEB)

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	15	8	1
Snapshot 2	18	12	1
Snapshot 3	27	21	5
Snapshot 4	26	18	7
Snapshot 5	24	16	1
Snapshot 6	28	19	6
Snapshot 7	18	9	3
Snapshot 8	17	12	2
Snapshot 9	25	15	4
Snapshot 10	23	16	4
Average	22.1	14.6	3.4
Percentage users	49.11	32.44	7.56

Table 6-7. Results simulations 90 users in the cell (45 VoIP + 45 WEB)

	Number of users with pw≥5%	Number of users with pw≥10%	Number of users with pw≥20%
Snapshot 1	25	17	5
Snapshot 2	28	18	9
Snapshot 3	32	20	2
Snapshot 4	32	24	6
Snapshot 5	30	24	5
Snapshot 6	31	28	9
Snapshot 7	31	24	5
Snapshot 8	27	17	6
Snapshot 9	36	27	9
Snapshot 10	35	32	13
Average	30.7	23.1	6.9
Percentage users	61.4	46.2	13.8

Table 6-8. Results simulations 100 users in the cell (50 VoIP + 50 WEB)

6.7 Bibliography

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