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WCDMA – Enhanced Uplink performance evaluation

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Thesis

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Abstract

This thesis focuses on the performance of the Enhanced Uplink for UTRAN, also called High Speed Uplink Packet Access (HSUPA), with special emphasis on the fast packet scheduling based at the NodeB, and how different scheduling mechanisms affect the performance of the system.

An extension to the Network Simulator (ns-2) environment has been developed to support the functionality of the Enhanced Uplink, including the HARQ and the Fast Packet Scheduling. Several scheduling mechanisms were implemented: a Rate Scheduling in which many users are scheduled at the same time with low rates; a Round Robin Scheduler which schedules few users with high rates; a UCQI Scheduler that gives priority to the users with best channel conditions; a Rate Estimation Scheduler that estimates the maximum supported rate by a user depending on its path loss, and a Priority Scheduler which gives priority to the user depending on the type of data they are transmitting. All these schedulers were used in different simulation scenarios to analyze and compare their impact in the performance of the system, and conclusions made.

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Abbreviations

3G	3^{rd} generation mobile communications
3GPP	3^{rd} generation partnership project
AMC	Adaptive Modulation and Coding
ARQ	Automatic repeat request
AVI	Actual value interface
BLER	Block error rate
CDMA	Code division multiple access
CN	Core network
CQI	Channel quality indicator
DCH	Dedicated channel
DPCCH	Dedicated physical control channel
DPDCH	Dedicated physical data channel
Eb/No	Data bit energy to interference ratio
E-DCH	Enhanced dedicated channel
FDD	Frequency division duplex
FTP	File transfer protocol
GGSN	Gateway GPRS support node
GPRS	General packet radio service
GSM	Global system for mobile communications
HARQ	Hybrid automatic repeat request
HSDPA	High speed downlink packet access
HSUPA	High speed uplink packet access
MAC	Medium access control
NR	Noise Rise
PDF	Probability density function
PDU	Packet data unit

PHY	Physical layer
PS	Packet scheduler
QoS	Quality of service
RLC	Radio link control
RNC	Radio network controller
RTT	Round trip time
TBS	Transport Block Size
TCP	Transmission control protocol
TDD	Time division duplex
TFC	Transport format combination
TTI	Transmission time interval
UCQI	Uplink channel quality indicator
UE	User Equipment
UMTS	Universal mobile telecommunications system
UTRAN	UMTS terrestrial radio access network
WCDMA	Wideband code division multiple access

1 Introduction

The evolution of wireless mobile communications had been encouraged by the new applications high bit rates and low delay demands. Video streaming, online gaming, Video conference, Voice over IP (VoIP), positioning applications, web browsing and FTP are some examples of these kind of applications that demand higher bit rates and low delay. Second generation systems, like GSM (Global System of Mobile communications), are concentrated in voice services which make them not the appropriate system to handle these type of applications; even with GPRS (General Packet Radio Service), an enhancement that supports packet data transfer, data bit rates are limited to 115kbs. Universal Mobile Telephone System (UMTS), a technology based on wideband code division multiple access (WCDMA) as air interface, has been developed to fulfill these high data rates and low delay demands and overcome the limited data handling of second generation systems.

Most of the applications considered for UMTS require a service where its traffic flow goes mainly in the downlink directions, thats why most of the work in this area is being focused on the optimization of this part of the system. As a result of that work an enhancement for the downlink called High Speed Downlink Packet Access (HSDPA) use several techniques to improve the efficiency of WCDMA and can provide peak data rates up to 10Mbps and considerably lower time delays. However, the uplink traffic direction of UMTS has not been forgotten and recently the 3GPP publish the specifications for an enhancement called High Speed Uplink Packet Access (HSUPA) or simply Enhanced Uplink.

This work will be focused in the enhanced uplink techniques used in HSUPA, mainly the NodeB packet scheduling. These techniques will be studied and modeled for using them in a system level simulator in order to obtain performance metrics for analysis, comparison and evaluation of the enhancement. In order to see the impact of different scheduling mechanisms in the uplink performance several algorithms were implemented in the simulator and their performance, in terms of delay, throughput and transmited power, was tested in different scenarios.

Chapter 2 presents an overview of UMTS describing its air interface, its system architecture and functionality as given in the 3GPP UMTS Release ‘99. Chapter 3 presents the new features of Release 6 describing the new techniques for the Enhanced Uplink. In Chapter 4 and 5an explanation of how these techniques were modeled for implementation in ns-2. Chapter 6 presents the simulations and results obtained using the implemented model of the Enhanced Uplink in different scenarios. Finally the conclusions of this work are presented.

2 **UMTS Overview**

UMTS is a 3rd Generation mobile communication system which provides high data rates and speech quality and is able to serve packet and circuit switched services. It has been chosen in almost all Europe and other countries as Japan and Australia as next generation of mobile communications systems, intended to replace GSM.

2.1 *WCDMA*

WCDMA (Wideband Code Division Multiple Access) is the technology behind UMTS used as its air interface. It is a complete set of specifications in which a detailed protocol defines how a mobile phone communicates with the base station or NodeB.

WCDMA can be divided into two modes, the Time Division Duplex mode and the Frequency Division Duplex mode. In TDD the uplink and the dowlink transmissions are time multiplexed into the same carrier while in FDD the uplink and the dowlink transmissions occur in different frequency bands (around 1900 Mhz range for the uplink and 2100 Mhz range for the dowlink) with a 5MHz bandwidth for each band. The FDD mode has been chosen as the mode of operation in Europe. In the rest of this document FDD will be the operation mode when it is referred to WCDMA.

In WCDMA air interface all users belonging to a cell are separated by codes, this allows the users to transmit in the same frequency and at the same time. Two kinds of codes are used in WCDMA operation, the channelization or spreading codes and the scrambling codes. The channelization codes separate channels from a single transmitter while the scrambling codes are used to separate transmitters. Channelization codes are also used to spread the bandwidth of the signal (thats why they are also called spreading codes).

2.1.1 **Channelization Codes**

Channelization codes (or spreading codes) are use to separate transmissions from a single source. This codes are based on the Orthogonal Variable Spreading Factor (OVFS) technique which give them a cross-correlation property and a auto-correlation property, that means that the inner product of the code with codes of other users or with a shifted version of itself is very small. When a full orthogonality is achieved (its an ideal state because orthogonality is affected by multipath propagation) there is no interference between codes.

WCDMA uses a fixed transmission chip rate of 3.84Mcps in order to approximately use the 5MHz frequency bandwidth of the channel. WCDMA can use the channelization codes to transmit information at different bit rates, transmitting every bit of information as a code at 3.84Mcps, the bit rate will depend on the length of the code. The shorter the code the higher the information bit rate. If every bit of information is multiplied by the spreading code with a chip rate of 3.84Mcps, it means that the bandwidth of the information signal is spread along the bandwidth used by the chip rate (approximately 5MHz).

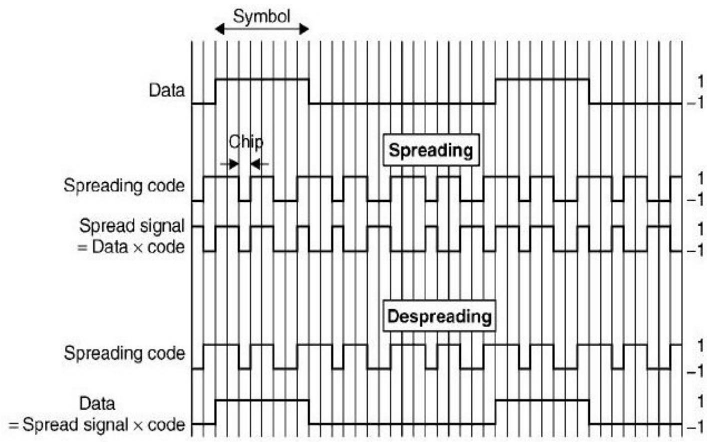

Figure 2.1: Spreading and Despreading [20].

2.1.2 Scrambling Codes

Scrambling codes are used to separate transmissions from different sources. In the uplink scrambling codes separate different mobiles and in the downlink separate different NodeB cells or sectors. This codes does not affect the transmission bandwidth and they are just use to separate transmitters. There are two families of scrambling codes (short and long codes). In the uplink short and long codes are used depending on NodeB receiver (long codes for Rake receiver and short codes for multiuser detectors). Each family of codes contains millions of scrambling codes so planning is not needed (in the uplink). In the downlink the number of codes is restricted to 512 to prevent that the cell search procedure takes too much time. Since the number of codes is limited network planning is needed to assign a scrambling code to a cell or sector, but 512 scrambling codes is a large number of codes and the planning is trivial.

2.1.3 **Transmission and Reception**

A WCDMA transmitter use a spreading code and scrambling code to spread and separate the signal by multiplying the information signal with these codes. The spectrum of the transmitted signal is spread in the 5MHz bandwidth of the channel so the information is propagated below the thermal noise level. This makes makes the detection of the signal very difficult without knowledge of the spreading sequence which introduce certain security to the transmitted information.

At the receiver side the same codes are used to be able to recover the information and separate it form signals received at same time (interference) and from the thermal noise in the channel. Figure 2.2 describes the transmission and reception process using the spreading and scrambling codes.

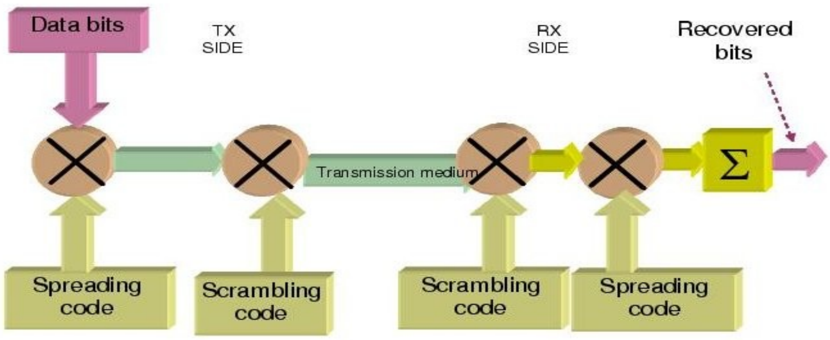

Figure 2.2: Simplified Transmission and Reception Process [16].

For being able to recover the information signal at the receiver, the power of the signal after despreading must be a few decibels above the interference and noise power. The Eb/No is defined as the energy, or power density, per bit (Eb) over the interference and noise power density (No). The higher the Eb/No the easier will be for the receiver to recover the information. The effect of raising the signal over the interference and noise power is called “processing gain”. Using correlation detection the signal is raised by the spreading factor form the interference present in the system.

$$Gp = \frac{TotalSpreadBandwidth}{InformationBitRate} = \frac{W}{R} \quad (1)$$

$$Eb/No = \frac{P}{(I - P)} \cdot \frac{W}{R} \quad (2)$$

where Gp is the processing gain, P is the received power of the information signal and I is the total interference (the total received powers plus the thermal noise).

2.2 UMTS Architecture

UMPS use WCDMA as air interface and an evolution of the GSM core network. Its architecture can be divided in three parts: the User Equipment (UE), the UMTS Terrestrial Radio Access Network (UTRAN) and the Core Network.

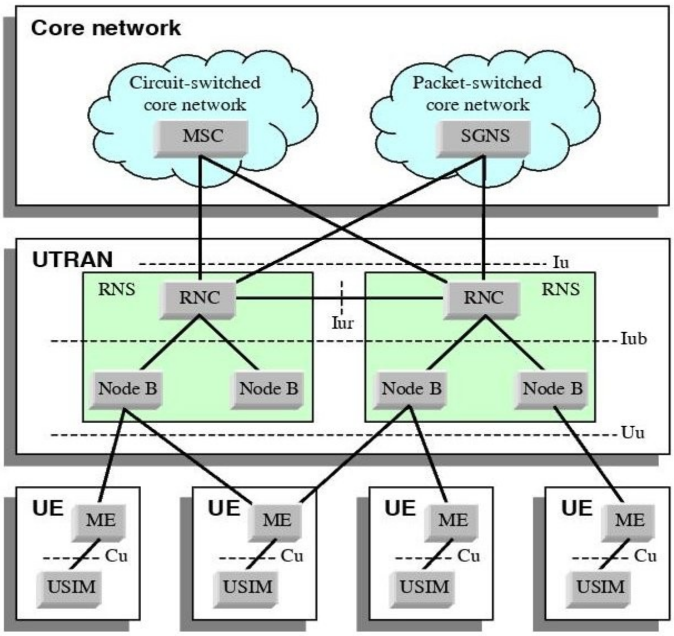

Figure 2.3 : UMTS architecture [15].

The UE is composed by the Mobile Equipment (ME) and the UMTS Subscriber Identity Module (USIM). The ME is the radio terminal used for radio communication with the UTRAN and the USIM is a smart card that holds the subscriber identity.

The UTRAN is composed by one or more Radio Network Sub-systems (RNS) and each of them consists of one Radio Network Controller (RNC) and one or several NodeBs. The main functionality of the NodeB is to provide the radio link between the UE and the UMTS network. It also performs Radio Resource Management operations as the inner loop power control. The RNC controls the radio resources of the NodeBs connected to it. It is responsible for the load and congestion control of the cell under its control, it also executes admission control and code allocation for new radio links to be established in those cells. Figure 2.3 illustrates the UMTS general architecture.

The main components of the Core Network are:

- HLR (Home Location Register) is a database located in the user's home system that stores the user's service profile.
- MSC (Mobile Service Switching Center) is the interface that serves the UE to access circuit switched data.
- VLR (Visitor Location Register) is the database that stores the user's service profiles and UE location within the system.
- GMSC (Gateway MSC) is the gateway between the UMTS and other external circuit switched networks like PSTN, ISDN, etc.
- SGSN (Serving GPRS (General Packet Radio Service) Support Node) is similar to the MSC but for packet switched services.
- GGSN (Gateway GPRS support Node) is the gateway that connects UMTS to other packet switched networks.

All the interfaces between the components of the UMTS network had been standardized:

- Cu is the electrical interface between the USIM and the ME.
- Uu is the WCDMA interface through which UE s access the fixed part of the system.
- Iu interface connects the UTRAN to the CN.
- Iur interface between RNCs which allows soft handover.
- Iub connects the NodeB with the RNC.

2.3 Transport Channels

In the UTRA the data generated at higher layer is carried over the air with transport channels. There are two types of transport channels, dedicated channels and common channels. A common channel is a shared resource between all the group of users in a cell and a dedicated channel is only used by a single user identifying it by a certain code.

DCH: is the Dedicated transport Channel used to carry all information form a given user coming from the higher layers.

BCH: is the Broadcast Channel, is a common channel used to transmit information needed within the cell (e.g available random access codes, access slots). It is decoded by the UE to be able to register to the cell.

FACH: Forward Access Channel, is a common channel that carries control information to the UEs in the cell.

PCH: Paging Channel, is a common channel that carries data for paging procedure (when the network wants to initiate communication with the UE).

RACH: Random Access Channel, is an uplink common channel that carries control information form the UE like a request to set up a connection

DSCH: Downlink Shared Channel, is a common transport channel that carries dedicated user data or control information. The channel can be decoded by all UEs but the information is intended for a specific UE.

2.4 Physical Channels

Transport channel are mapped into physical channels. A Physical channel can carry several transport channels or it can carry just information relevant to the physical layer.

Transport Channels	**Physical Channels**
DCH	Dedicated Physical Data Channel (DPDCH)
	Dedicated Physical Control Channel (DPCCH)
	Fractional Dedicated Physical Data Channel (F-DPCH)
RACH	Physical Random Access Channel (PRACH)
	Common Pilot Channel (CPICH)
BCH	Primary Common Control Physical Channel (P-CCPCH)
FACH	Secondary Common Control Physical Channel (S-CCPCH)
PCH	
	Sinchronisation Channel (SCH)
	Acquisition Indicator Channel (AICH)
	Paging Indicator Channel (PICH)
	MBMS Notification Indicator Channel (MICH)

Figure 2.3 Transport-channel to physical-channel mapping [4].

A DCH is mapped onto two physical channels. The Dedicated Physical Data Channel (DPDCH), which carries the user data, and the Dedicated Physical Control Channel (DPCCH) which carries the necessary physical layer control information. Since the DPDCH use a variable bit rate, the DPCCH carries the transport format combinations needed by the receiver to be able to decode the DPDCH in every frame.

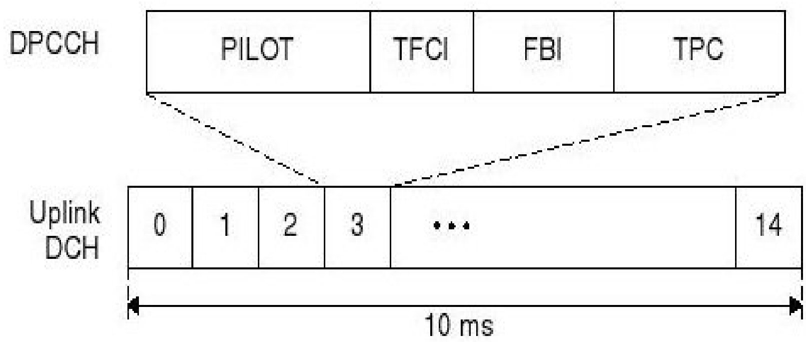

Figure 2.4: DPCCH frame structure [20].

Every Transmission Time Interval (TTI) the receiver will decode the TFCI information from the DPCCH frame to obtain the bit rate at which data (in the same TTI frame) was sent in the DPDCH. With this information the receiver can decode the data in the DPDCH.

2.5 Power Control

Since UEs that are located far away form the NodeB experience greater attenuation than the ones that are near the NodeB (the near-far effect) there is the need to control the transmission power of the UEs so they don't interfere with each other. If UEs transmit at the same power the received power at the NodeB for the near UE s will be much bigger that the power of the UE s that are far away, the interference cause by the near UEs can make the reception of the others not possible.

The main purpose of power control is to keep the interference level at a minimum level and try to keep the same received power levels form all user. In this way the UEs that are far away from the

NodeB will transmit at a higher power than the ones that are near the NodeB. In the same way the NodeB will transmit with less power to UEs located close to it and with more power to UEs located far away.

- **Open Loop power control:** it makes an estimate of the path loss by means of a dowlink beacon signal to set up the UE initial transmission power level. It is inaccurate because of the large frequency separation in the downlink and uplink bands of WCDMA FDD mode. It is mainly use to provide the initial power setting of the UE at the beginning of a connection.
-
- **Close Loop power control:** the NodeB measures the signal to interference ratio in the uplink and commands the UE to increase or decrease the transmission power. WCDMA updates the transmission power level at 1.5KHz (1500 times per second) and thats why it is called fast power control.
-
- **Outer Loop power control:** the NodeB tells the RNC that the quality indicator of the signal (in the uplink and in the downlink) is decreasing so the RNC will take a decision and commands the NodeB to change (increase) the signal to interference ratio level.

2.6 Admission and Congestion Control

Admission and congestion control are part of the radio resource management performed by the RNC. The admission control will make the decision of accepting the admission of a new UE in the cell if its admission doesn't exceed a certain limit in the cell load. Congestion control will take actions when the cell is becoming overloaded (mainly cause by movement of UEs from one cell to another). The RNC will make a decision when the cell is overloaded and will try to bring back the system to a normal condition reducing the non-real time applications bit rates or more drastically dropping low priority calls.

2.7 Packet Scheduling

Since non real time traffic doesn't require any strictly guaranteed data rate and delay they are not restricted in terms of the amount of minimum resources needed to meet a specific QoS. This traffic can be controlled or scheduled according to the available resources of the system. The Packet Scheduler (PS) performs this task taking into account the available resources, the amount of data in the RLC buffer and UE power capabilities.

In the 3GPP release 99 the scheduling operates at the network layer meaning that the scheduler is based on the RNC. The UE sends reports of its traffic volume measurements (TVM) to the PS in the RNC every time the buffer in the UE reach a certain threshold (1). The scheduling process in the RNC receives this reports and calculates the maximum Transport Format Combination (TFC) that a UE is allowed to use in the following scheduling period (2). The TFC allocated correspond to an index pointing to the maximum data rate allowed and its sent through Iub to the NodeB who forwards it to the UE (3). The UE can select any TFC bellow the maximum established by the PS (4). The maximum TFC allocated by the PS to a UE could require more power than the UE capabilities, in that case the UE will select a TFC bellow the allocated TFC which is suitable for its power capabilities for its data transmission (5). [15].

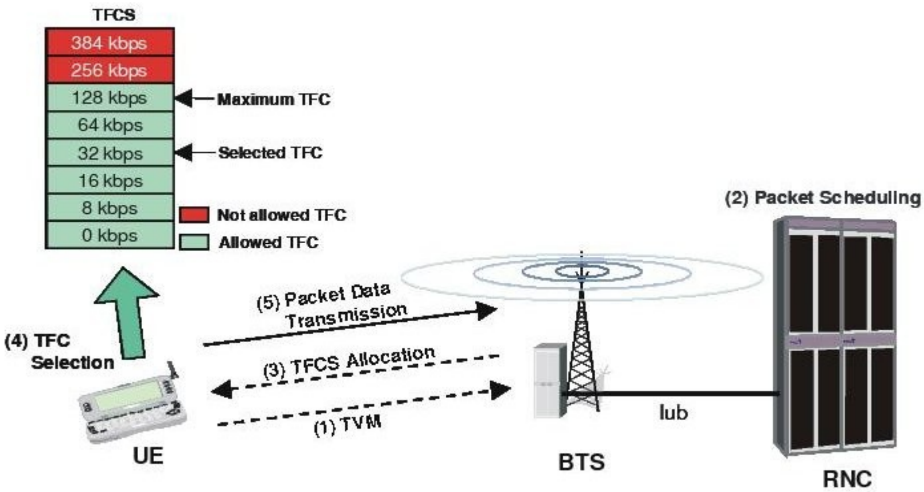

Figure2.5 Packet Access with PS based in the RNC [15].

3 High Speed Packet Access

3.1 HSDPA

High Speed Downlink Packet Access (HSDPA) introduce new techniques in order to optimize and improve the packet transmission over UTRAN. HSDPA introduce a new downlink common channel called HS-DSCH (High Speed DSCH) which support new features as high order modulation, fast link adaptation, fast channel dependent scheduling and fast hybrid automatic retransmission request (HARQ).

HSDPA can select a modulation method from QPSK to 16QAM in order to provide higher data rates. The link adaptation is based on AMC (Adaptive Modulation and Coding) where the most appropriate modulation scheme and coding rate are selected based on on fast channel quality feedback received from the UE. This ensures that the highest possible data rate is achieved for users with good signal quality and for users at the edge of the cell. Functions like the variable spreading factor (VSF) and fast power control are deactivated in HSDPA and their functionality is replaced by the new features like AMC.

The packet scheduling is moved from the RNC to the NodeB reducing the delay and allowing the scheduling algorithm to use parameters as the channel quality, terminal capabilities and QoS class.

The fast HARQ enables the UE to rapidly request physical layer retransmission of a data packet. This retransmission is done form the NodeB and not from the RNC as done before which make the retransmission faster. Besides the fast retransmission, incorrectly received packets are not discarded but stored and soft combined with the later retransmissions of the same packet to minimize the need for further repeat requests when multiple errors occur in transmitted signals.

3.2 HSUPA

As in the donwlink, the need to improve the uplink air interface lead to the introduction of HSUPA and its specification by the 3GPP in the Release 6.

The approach for a high speed uplink took some of the techniques used for HSDPA like the HARQ and fast scheduling. The technique of adaptive modulation and coding used in HSDPA will not be useful in HSUPA. One of the reasons is that the power resource capabilities of each individual user are limited. That means that a user may not have enough power capabilities for a high rate transmission (higher order modulation). Besides, higher order modulation has been found to cause loss in link performance compared to multi-code transmission with BPSK [6], so just BPSK and

QPSK will be used for the Enhanced Uplink, removing the advantage of high rates achieved with 16QAM in HSDPA. Another reason that doesn't allow AMC to be used in the uplink is that the received signal at the NodeB of UEs that transmit at the same time is not orthogonal and even if just one UE is scheduled at a time other cell interference will not allow a good quality estimation of the channel. The performance of the AMC depends on the quality estimation of the channel and since it will have to deal with inta-cell and inter-cell interference the channel quality will change too much from the time the measurement or estimation was made to the time of the transmission. All this reason lead to exclude AMC from the new features of HSUPA and use fast close loop power control mechanism as the link adaptation mechanism to be able to avoid the near-far problem.

For all this reasons we can expect that the results in the enhancements in the uplink will not be as good as in the downlink.

The basic improvements for HSUPA are the fast packet scheduling and fast Hybrid Automatic Retransmission Request (HARQ). The control of these two techniques will reside in the NodeB allowing faster response times and less round trip delay than residing in the RNC. This changes also allow the scheduling to react rapidly to changes in the traffic load giving or taking out resources from users depending on the load.

Another basic difference with HSDPA is the use of an enhanced dedicate channels (E-DCH) for each user in the uplink and not of a shared channel common to all users as in the downlink.

3.2.1 Enhanced Dedicate Channel (E-DCH) Protocol Architecture.

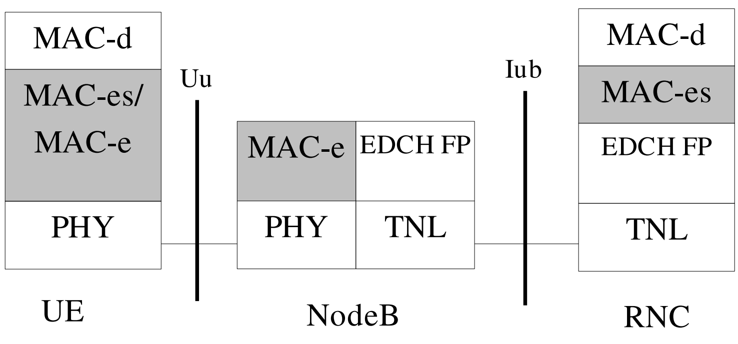

Figure 3.1: E-DCH protocol architecture

Basically, to support the E-DCH a new MAC entity (MAC-e/MAC-es) in the UE, NodeB and RNC is introduced. The MAC-e is added to the UE and NodeB in order to handle HARQ retransmission, scheduling, multiplexing (at the UE) and demultiplexing (at the NodeB). This entity will handle the primary enhancements of the high speed uplink which are the HARQ and the NodeB Packet Scheduling.

A re-ordering entity is introduce as a separate MAC sub-layer called MAC-es entity and it is added in the UE and in the RNC in order to provides in-sequence delivery. This entity is placed in the RNC because data packets could be transmitted through different NodeBs in case of soft handover so all the sequence of data packets will just be available in the RNC for their in-sequence delivery. The reordering is based on Transmission Sequence Numbers (TSN) include in the MAC-es at the UE side.

3.2.2 **MAC-es/MAC-e**

In HSUPA the overall MAC architecture includes the new MAC-es/MAC-e entity, this entity will be in controls of the E-DCH.

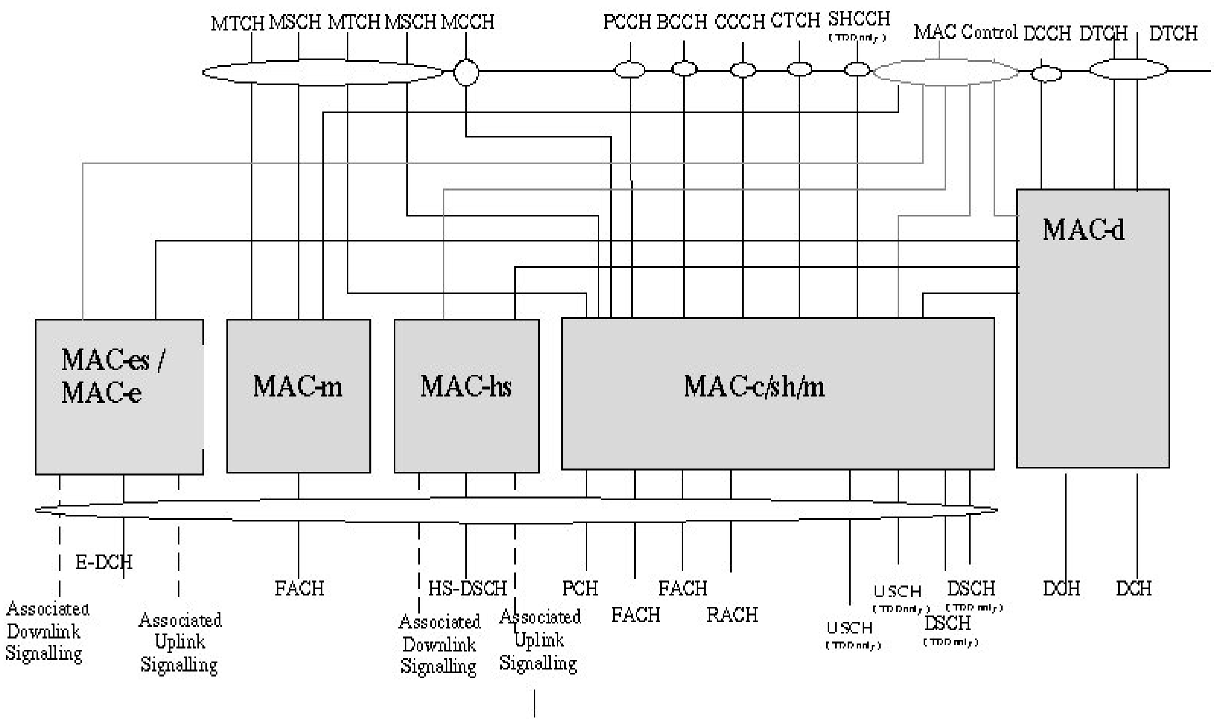

Figure 3.2: UE side MAC architecture [12].

In the UE the MAC-d entity will receive from the RLC the dedicated logical channels, the MAC-d is responsible for mapping these dedicated logical channels for the uplink onto dedicated transport channels. It can also multiplex several dedicated logical channels onto one transport channel. The

MAC-es entity will receive MAC-d PDUs and will attach to them a TSN (6 bits sequence number) to provide reordering at the receiver. MAC-e will multiplex the transport channels (MAC-es PDUs) into MAC-e PDUs with a DDI (6 bits Data Description Indicator) that identifies each transport channel. After multiplexing, the MAC-e must schedule the packet transmission according to the selection of a TFC from the serving set (taking into account its power capabilities) and transmit just for the time interval specified with the Absolute Grant.

MAC-e PDUs transmitted will be stored in the HARQ handler for retransmission if a negative acknowledgment is received. A fixed number of retransmissions will be established.

The MAC-e PDUs are mapped into the physical channel (E-DDCH).

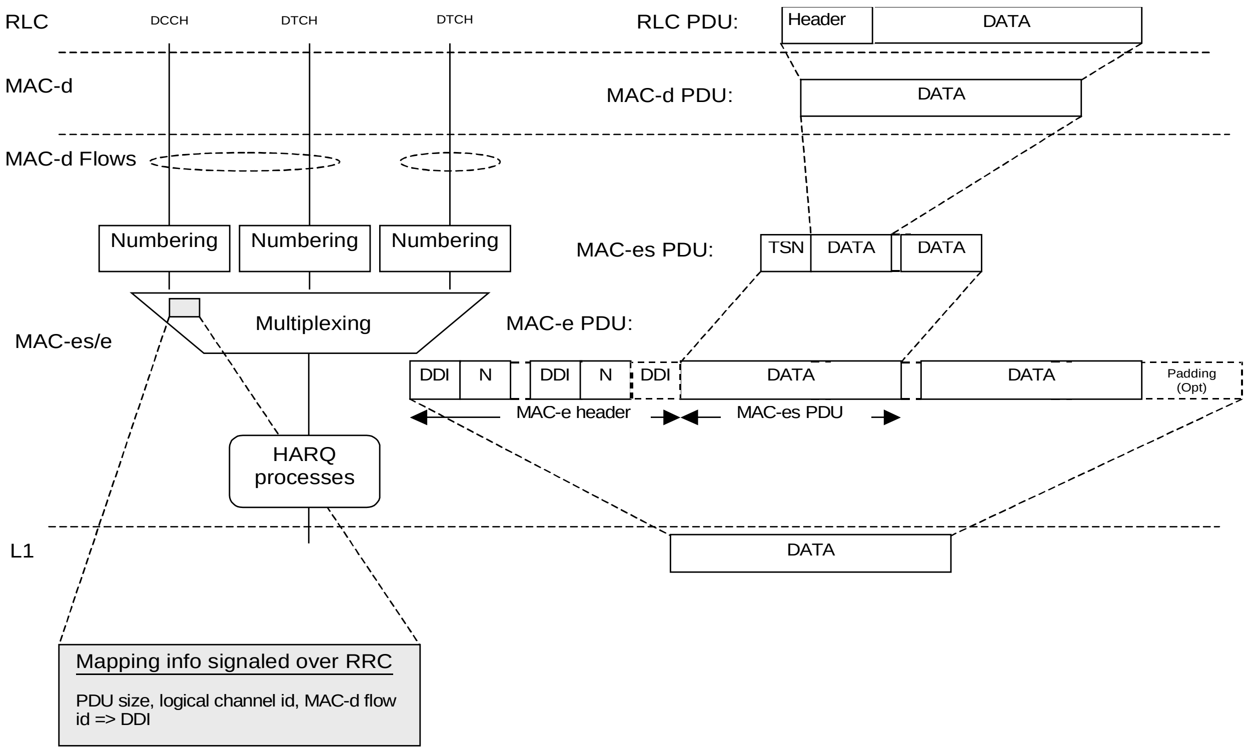

Figure 3.3: MAC inter working in UE [11].

For each UE that uses E-DCH, one MAC-e entity per Node-B and one MAC-es entity in the SRNC are configured. The frame received in the E-DDCH is decoded in the MAC-e entity. The HARQ handler will send an acknowledgment (through the E-HICH) if the packet was received successfully or a negative acknowledge if not. The MAC-e will demultiplex the frame into the respective transport channels based on the DDI in the frame header. In the RNC, the MAC-es will receive each transport channel packet and will reorder the packets based on the TSN attached to them. The MAC-d will separate the different logic channels coming in the same transport channel.

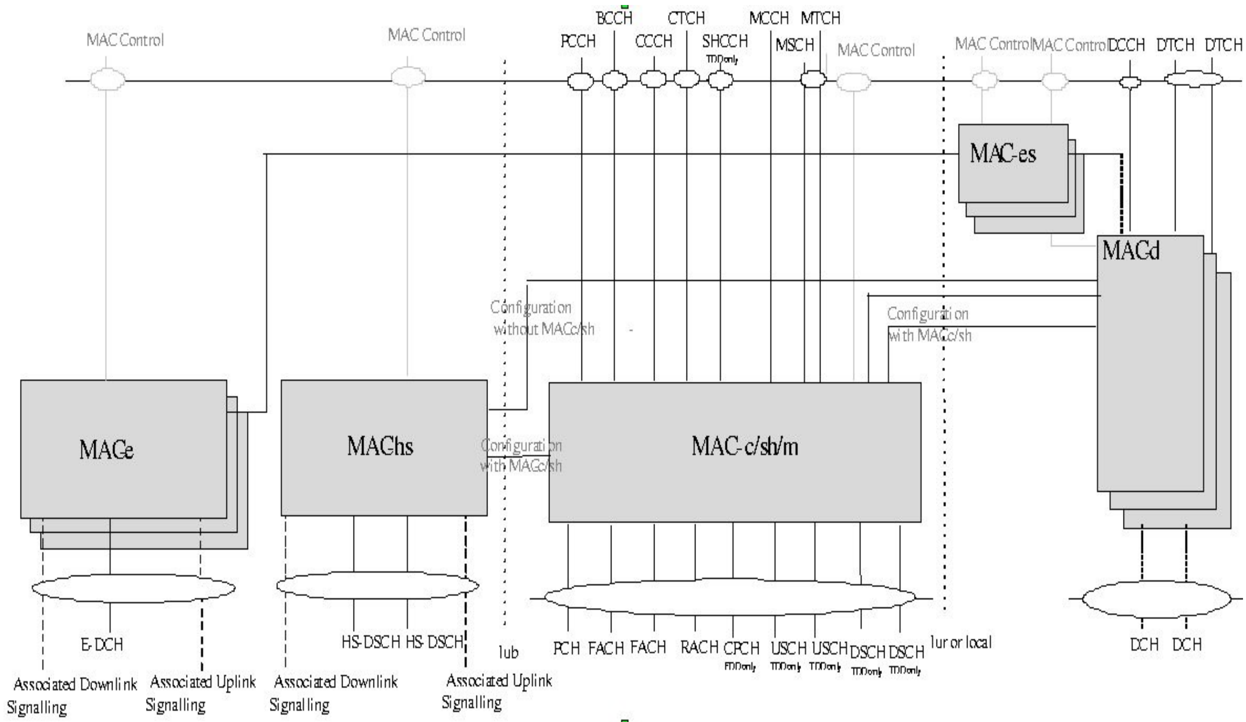

Figure 3.4: UTRAN side MAC architecture [12].

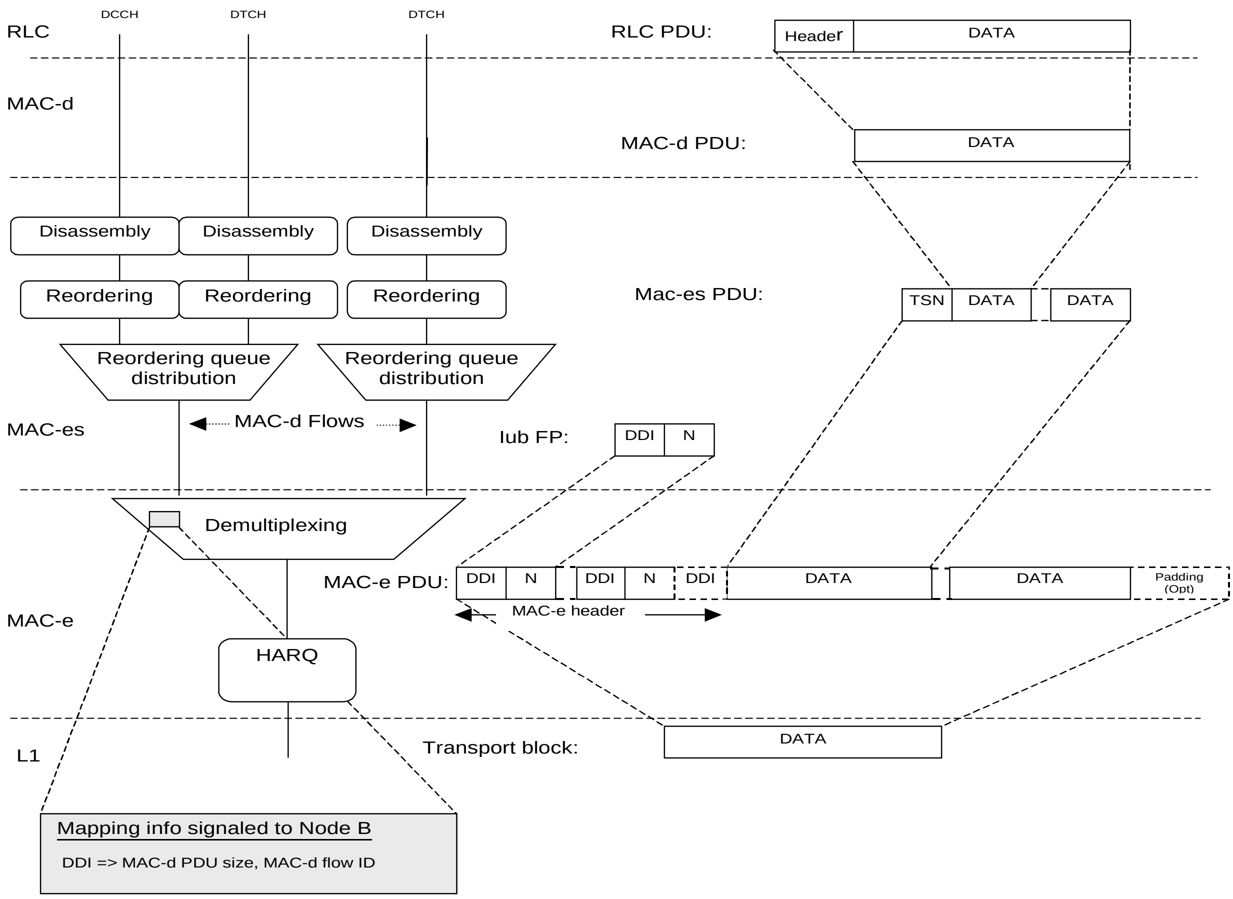

Figure 3.5: MAC inter working in UTRAN [11].

3.2.3 **Enhanced Uplink Physical Channels**

To support the new enhanced features of HSUPA new physical channels were introduced. Basically, the E-DCH is mapped into the E-DPDCH and the E-DPCCH. Other physical downlink channels are defined to carry the necessary physical layer signaling between the NodeB and the UE.

Uplink:

E-DPCCH: dedicated channel for uplink control signaling (just one per UE).

E-DPDCH: dedicated channel for uplink transport channel, it carries the user data and can have variable bit rates. The E-DPDCH is divided in 5 sub frames of 2ms.

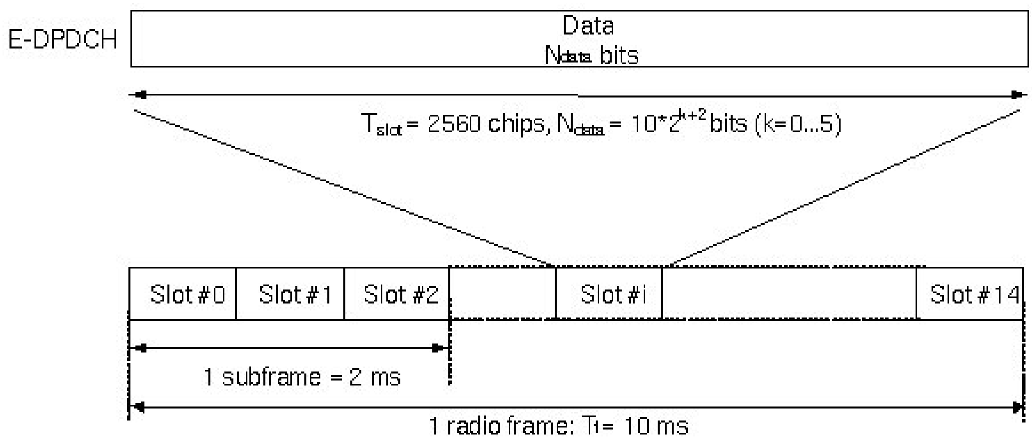

Figure 3.6 E-DDCH frame structure [13].

Downlink:

Fractional DPCH (F-DPCH): dedicated channel for layer 1 control information (e.g TPC).

Enhanced-Relative Grant Channel (E-RGCH): is a fixed rate (SF=128) downlink physical dedicated channel carrying the uplink E-DCH Relative Grants.

E-DCH Hybrid ARQ Indicator Channel (E-HICH): is a fixed rate (SF=128) downlink physical channel carrying the uplink E-DCH Hybrid-ARQ Acknowledgment (HARQ-ACK) indicator.

Enhanced Absolute Grant Channel (E-AGCH): is a fixed rate (30 kbps, SF=256) downlink common physical channel carrying the uplink E-DCH Absolute Grant.

Figure 3.7 shows the mapping of the transport-Channel to Pysical-channel introduced for the Enhanced uplink and for the High Speed Downlink.

Transport Channels	Physical Channels
E-DCH	E-DCH Dedicated Physical Data Channel (E-DPDCH)
	E-DCH Dedicated Physical Control Channel (E-DPCCH)
	E-DCH Absolute Grant Channel (E-AGCH)
	E-DCH Relative Grant Channel (E-RGCH)
	E-DCH Hybrid ARQ Indicator Channel (E-HICH)
HS-DSCH	High Speed Physical Downlink Shared Channel (HS-PDSCH)
	HS-DSCH-related Shared Control Channel (HS-SCCH)
	Dedicated Physical Control Channel (uplink) for HS-DSCH (HS-DPCCH)

Figure 3.7: Transport-Channel to Pysical-channel mapping [4]

3.2.4 Fast Hybrid Automatic Retransmission Request (HARQ)

In the release 1999 of WCDM the retransmissions are made in the Radio Link Control level so the retransmission procedures for a packet is located in the RNC. In the uplink, a packet must arrive to the RNC and a negative acknowledgment has to come back in order to perform a retransmission. With fast HARQ a retransmission can be controlled directly by the NodeB reducing the delay so the retransmissions are performed faster.

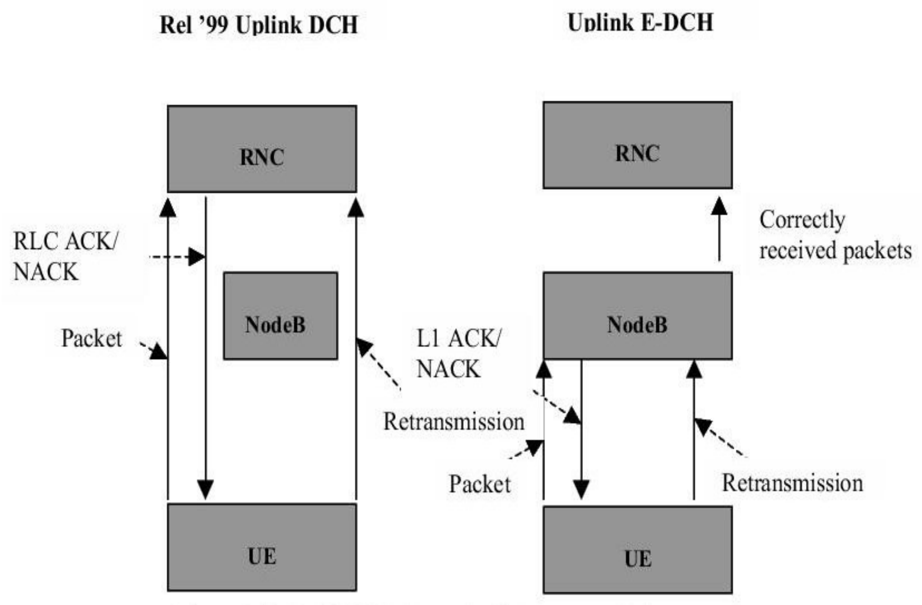

Figure 3.8: HARQ [27].

Figure 3.8 shows the difference between the retransmission mechanism used for the uplink DCH and the HARQ used in the E-DCH.

Fast HARQ is based on a stop-and-wait protocol which means that the transmission does not continue until a positive or negative acknowledgment is received. A limited number of HARQ processes is allowed to be active at the same time, that means the UE can keep sending packets continuously while waiting for the acknowledgments of each sent packet until it reaches a limit number. The number of processes depends on the transmission time interval (TTI) and have been specified 8 for 2ms TTI and 4 for 10ms TTI for the Enhanced Uplink [12]. Also a limited number of retransmission attempts is set so if this number is reached the packet is dropped.

HARQ enables rapidly request of physical layer retransmission of a data packet. The retransmission request is made in the NodeB decreasing the time than if it is made from the RNC.

Besides the fast retransmission, incorrectly received packets are not discarded but stored and soft combined with the later retransmissions of the same packet to minimize the need for further repeat requests when multiple errors occur in transmitted signals.

Soft combining: with this scheme if an erroneous packet is received, the packet is stored in the receivers memory. Retransmissions are exact copies of the first transmission and if again are received erroneously they are combined with the first transmission to improve the likelihood that the packet is correctly received.

Incremental Redundancy (not self-decodable): in this scheme the first transmission data is sent, and in the following transmissions only redundancy bits are sent. This means that the data in the retransmission can only be used together with the original packet.

Incremental Redundancy (self-decodable): In this scheme each retransmission is self-decodable, meaning that a retransmitted packet can be decode even if the original packet was lost.

It could be the case that the UE is in a soft-handover situation and two NodeB receive a packet and two acknowledgments are sent to the UE, if at least one positive acknowledgment is receive by the UE the packet is considered as correctly received. In a similar case a retransmission of a packet received by a second NodeB who didn’t receive the first transmission could only decode the packet if soft combining or self-decodable incremental redundancy are used.

3.2.5 **Fast Packet Scheduling**

The RNC based Packet Scheduling brings long delays because of the needed signaling between the UE and the RNC. RNC based Packet Scheduling of non real time traffic, which are characterized by a bursty traffic profile, is not efficient because it takes too much time to allocate and release resources; when a user is allocated with large amount of resources for a bursty transmission this resources take a long time to be released. To be able to rapidly allocate and release resources to the UEs depending on their load the Packet Scheduling is moved to the NodeB reducing the delay and allowing it to react faster than residing in the RNC. NodeB scheduling enables a better control over the uplink interference which leads to increase the capacity and improve the coverage of the cell.

Figure 3.9 illustrates the steps of packet transmission using NodeB PS. First the UEs will request the PS located in NodeB for resources using an uplink signaling(1) and the PS will determine the TFCS which they are allowed to use (2). The PS will grant each UE with a set of allowed TFC (3), form this allowed set the UE will select a TFC which is suitable according to its power capabilities (4) for transmission of the packet (5). [27].

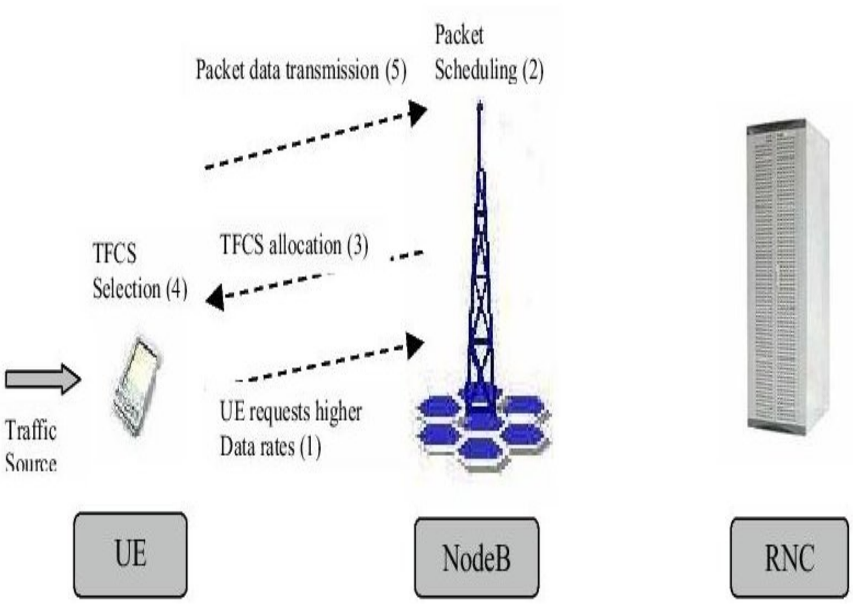

Figure 3.9: Uplink packet access scenario [27].

In HSDPA the PS was also moved to the NodeB but the differences between the downlink and the uplink transmission make the PS algorithm more complicated for the Enhanced uplink. In the downlink the transmission is made from a common entity, the NodeB, so it can control its own power. In the uplink, the transmission it is distributed among all the users in the cell which makes more complicated to control the received power form all the users.

The Packet scheduler needs to share the available air interface capacity between the users and there are two types of scheduling schemes that could be used for this purpose, Rate Scheduling (or Code Scheduling) and Time Scheduling.

Rate Scheduling is a type of scheduling in which users can perform uplink transmission in parallel with a low rate so that the noise rise level at the NodeB do not exceed the established threshold given by the RNC. This type of scheduling makes a restriction in the rate used by the UE but does not irestrict the time the UE used to transmit. Since several user can transmit at the same time their rates must be kept low when the cell is highly loaded so they don’t cause to much interference between each other.

In the Time Scheduling scheme only one or a set of UEs are allowed to transmit at a given time so that the noise rise level at the NodeB do not exceed the established threshold given by the RNC. With this scheme it is possible to allocate very high data rates during short periods.

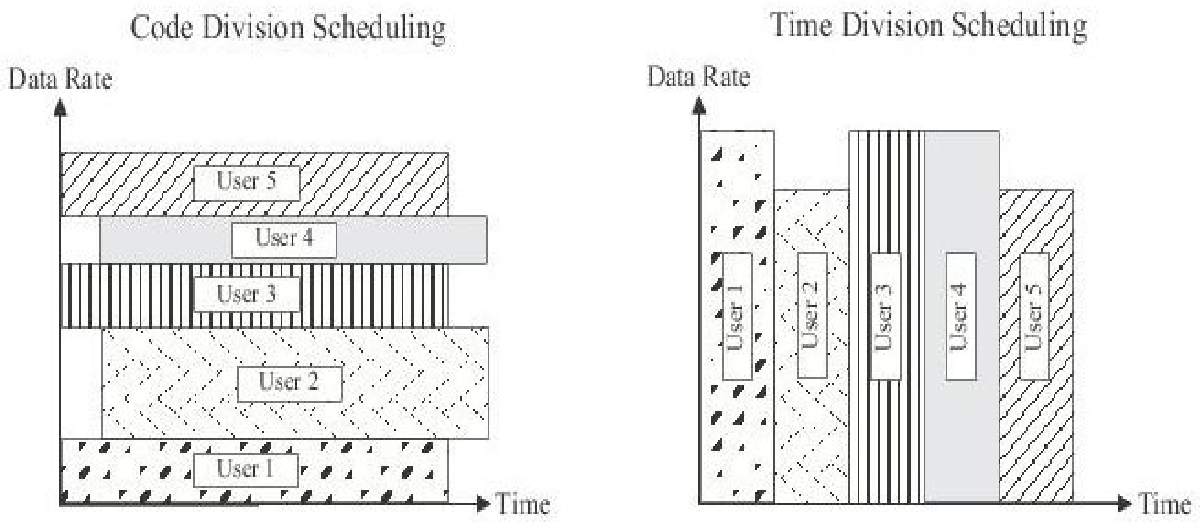

Figure 3.10 Code Division And Time Division Scheduling [20].

Moving the PS to the NodeB also implies that some information that could be used in the scheduling algorithm will not be available any more. This is the case for the information used for the RNC scheduling (the TVMs), the TVM is a network layer signaling which will not be available at the NodeB. This fact implies the new physical layer signaling must be introduce in the uplink and in the downlink direction.

3.2.5.1 *Uplink Scheduling information*

The scheduling signaling sent in the uplink direction is used as control information by the UEs to indicate the NodeB the amount of resources they requirer. This type of signaling can be seen as a request for resources from the UE to the NodeB. 3GPP release 6 specifies the signaling used by the UE to make a request. A single bit field is included in the E-DPCCH for every E-DCH transmission. This bit called “happy bit” takes two values, “not happy” and “happy” indicating respectively whether the UE could use more resources or not. Other needed information for the scheduling is sent in the uplink direction as part of the MAC header (specifically in the MAC-e header) and will be use to provide the NodeB with a better view of the amount of system resources needed by the UE and the amount of resources it can actually make use of. The information sent to the NodeB will include the total E-DCH buffer status (5 bits), the highest priority logical channel ID (4 bits) with its buffer status (4 bits) and the UE power headroom (5 bits) which is the ratio between the maximum allowed UE Tx power and the DPCCH.

The transmission of this uplink scheduling information can be configured by the Radio Resource Controller to be periodic (setting a timer) or triggered by an event (when the buffer status is larger than zero and there is no available grant).

Some schedulers will use the channel condition experienced by the users to make a scheduling decision. The Uplink Channel Quality Indicator (UCQI) is a parameter which indicate the quality of the channel. Basically this parameter is an indicator of the path loss experienced by each user and for the NodeB to calculated this parameter it will need the values of the received power and the transmitted power of the UEs.

To obtain the UE transmitted power the UTRAN must ask the UE for a measurement report. The UE will execute this measurement and it will signal the obtained value to the UTRAN in the Dedicated Control Channel. This is a network layer signaling which will not be available at the NodeB so the RNC will have to provide the NodeB with this information. This means that the measurement of the UE transmitted power will not be available in a TTI resolution in the NodeB to execute the scheduling process and an estimation of this value with a less frequent report will have to be made. Relaying on this method to get the value of the UE transmitted power will imply that all the gain obtain with a channel dependent scheduler will be depending on the estimation of this value.

The signaling of this measurement reports for the UE transmitted power is not considered in the simulator and it is assumed that the NodeB will have this information available every TTI for the scheduling process with a high accuracy. Even doe this assumption is used in the simulator, in reality it is possible to think on a different way for the NodeB to measure the UE transmitted power at a TTI resolution.

If the NodeB is able to obtain a value for the maximum UE transmission power at the beginning of the connection it is possible to calculate the UE transmitted power every TTI using the headroom signaling included in the MAC-e header and computing the path loss.

The headroom is defined as the ratio between the maximum transmission power (MaxPow) and the power of the DPCCH (TxDPCCH).

$$Headroom = \frac{MaxPow}{TxDPCCH} \tag{3}$$

At the NodeB using the known value of the UE maximum transmission power (MaxPow) and the received power of the DPCCH (RxDPCCH) it is possible to calculate the path loss (PL) experienced by that user.

$$Headroom = \frac{MaxPow}{RxDPCCH * PL} \tag{4}$$

$$PL = \frac{MaxPow}{RxDPCCH * Headroom} \tag{5}$$

There are different types of UE with different power capabilities so the NodeB must be able to get information about the maximum transmission power of each UE. Since this parameter will not change it can be signaled just once at the beginning of the connection.

*3.2.5.2 **Downlink Scheduling information***

Dowlink signaling sent from the NodeB to the UEs is needed in order to control the use of the E-DCH system resources. This type of signaling is called a Grant because the NodeB grants the UEs with a certain amount of resources. Two types of Grants are specified Relative Grants and Absolute Grants.

Relative Grants are transmitted on the E-RGCH and they can only take one of three values, “up”, “down” or “hold”, indicating the UE to step up, step down or hold the index of its allowed TFC. Neighboring NodeBs are allowed to use a non-serving Relative Grant that just takes the values “down” and “hold” to indicate UE that are not under their control to adjust their transmitted rate.

Absolute Grants messages are sent on the E-AGCH from the serving cell and allows the NodeB scheduler to directly adjust the resources of UEs under its control in one single command. With absolute grants there is no ramp up or ramp down when giving or taking away large amount or resources. The AG contains an Absolute Grant Value which indicates the maximum E-DCH traffic to pilot ratio (E-DPDCH/DPCCH) that the UE is allowed to use in the next transmission.

At the beginning of a connection a UE is configured with a E-TFCI table (Enhanced Transport Format Combination Indicator table). Different tables are specified by the 3GPP in [12] for 2ms TTI and 10ms TTI. The TFC indexes are mapped in this table to their corresponding transport block size (TBS). Specifying the number of bits (the TBS) that can be transmitted per TTI (2ms or 10ms) a TFC index will determine instantaneous bit rate at which a UE is allowed to transmit.

3.2.6 **Performance of HSUPA**

The performance of the HARQ and the Fast Packet scheduling introduced for HSUPA has been evaluated in terms of throughput in a system level simulator in [15]. The results of this work show that using HARQ at the physical layer and a packet scheduler at the NodeB can provide a gain of 31% compared to a RNC packet scheduler with HARQ at the network layer. A Proportional Fair Throughput Scheduler is introduced which allocate resources fairly to all the UEs keeping track of their individual channel conditions and scheduling them when they experience good channel conditions relative to their own average. The scheduler used in [15] shows an improvement in the average throughput with a gain of 45.6% compared to the RNC packet scheduler.

There are several characteristics of the fast packet scheduler that contribute to the throughput gain shown in [15]. In this work several schedulers will be implemented in order to focus on specific characteristics that contribute to the better performance of the system. Analyzing results in terms of total transmitted power, average delay and individual user throughput and delay measurements will provide us with a good view of the contribution given by the scheduler enhancements and policies.

4 Modeling of HSUPA

To be able to implement the different techniques used in HSUPA as an extension to the Network Simulator (ns-2) a model that simulates as close as possible to reality the behavior of this techniques must be specified.

To be able to model this behavior it is necessary to include in our model the performance of the physical layer which measurements must be taken with a very high resolution capturing the behavior of every bit transmitted. At the same time we need to capture the behavior of events that occurred in much longer periods of time (like network layer protocol mechanisms) which need the simulation of the system to gather information through several minutes.

Combining this two requirements will lead to very large computer simulation times. For this reason simulations are dividend into Link level simulations, that operate at very high resolution (bit or chip level resolution), and System level simulations, that operate at a smaller resolutions (like slots or TTIs).

Using a technique called Actual Value Interface the model for the system level simulator will incorporate the link layer performance results made in previous simulations.

4.1 Actual Value Interface (AVI)

In system level simulations where the simulations operate at the resolution of the most frequent event, most of the times per slot or TTI for WCDMA, it is too complex to include the resolution needed to evaluate the performance in the link level where a chip-level time resolution is needed (3.84Mcps for WCDMA). That is why simulations are divided in Link level simulations and System level simulations to obtain feasible computer simulation times with the required accuracy. Nevertheless, there is the need to incorporate the accuracy of link level simulations into the system level simulation and it is done using Actual Value Interface (AVI) tables.

The link level simulation output is provided as input to the system level simulator in the form of curves of Eb/No vs BLER. The AVI tables contain the information of this curves providing a BLER as a result of a specific Eb/No.

To obtain the required AVI tables for the WCDMA uplink transmission a method described in [15] to generate uplink AVI tables for different modulations and codings schemes (MCS) was used. This method describes how to get uplink estimated results for different MCS using one base curve with an specific MCS in the uplink and a previous link level study for the downlink. The basic assumption of this method is that the difference (in dB) between the AVI tables for different MCS in the dowlink is the same as in the uplink.

Estimated link level results were used because real results are very difficult to find while studies for the downlink are easily found since a lot of investigation has been done in this area.

The first thing to be considered is that the dowlink curves are given in energy per symbol over noise ratio (Es/No) which is a measurement taken before channel decoding. For the uplink AVI tables the energy per bit over noise ration (Eb/No) after channel decoding is the needed measurement. To calculate one using the other this formula is used:

$$(Eb/No)=(Es/No)\frac{Rs}{R}\frac{1}{Rcoding} \tag{6}$$

where Rs is the symbol rate, R the information rate and Rcoding is the coding rate.

The uplink reference curve used as base is obtain from a BPSK modulation and coding rate 1/3. With this reference and the previous formula a formula to obtain the offset between the reference curve and the required curve is deduced:

$$\Delta(Eb/No)=\frac{(Eb/No)_{req}}{(Eb/No)_{ref}}=\frac{(Es/No)_{req}\frac{Rs}{R}\frac{1}{Rcoding}}{(Es/No)_{ref}\frac{1}{1/3}}=\Delta(Es/No)\frac{N}{3\,Rcoding} \tag{7}$$

This equations expresses the offset between (Eb/No) of the required curve and the (Eb/No) of the reference curve in terms of the offset of the (Es/No) of those curves, where N is the number of bits per symbol of the modulation used and Rcoding is the coding rate.

The next step is to consider the use of a uplink control channel, the DPCCH. Depending on the rate of the data channel (DPDCH) a ratio between the control channel and data channel is specified and it must be considered in the acquisition of the (Eb/No) offset.

$$\Delta(Eb/No)=\frac{\Delta(Es/No)\frac{N}{3\,Rcoding}+p}{1+p} \tag{8}$$

In this equation p is the ratio between the control and the data channel for the a given rate.

Given the obtained offset considering the previous formula, the offset is added (in dB) to the reference uplink AVI table to obtain the required curve.

Figure 4.1 shows an scheme of the required information needed to obtain the uplink estimated AVI tables for different MCS.

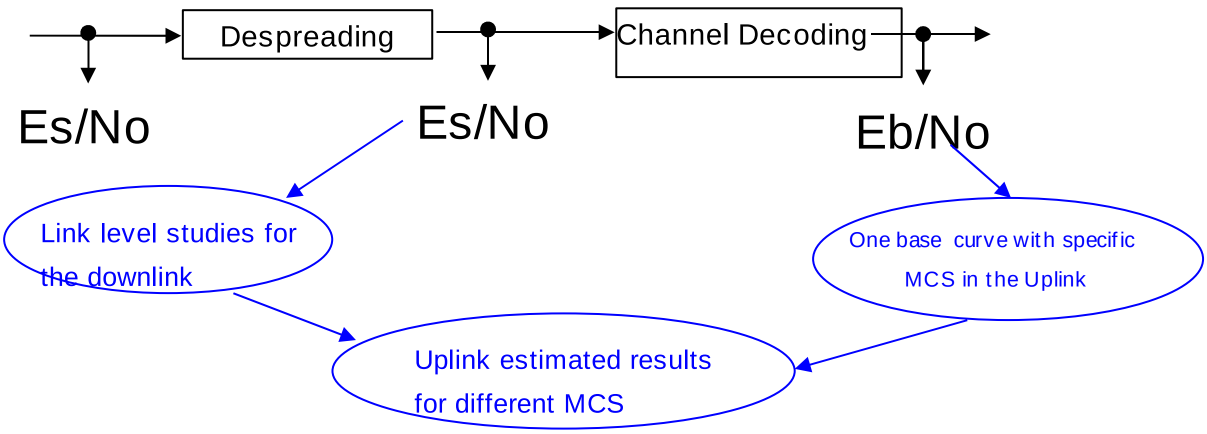

Figure: 4.1. Estimating link level results for different MCS in the uplink.

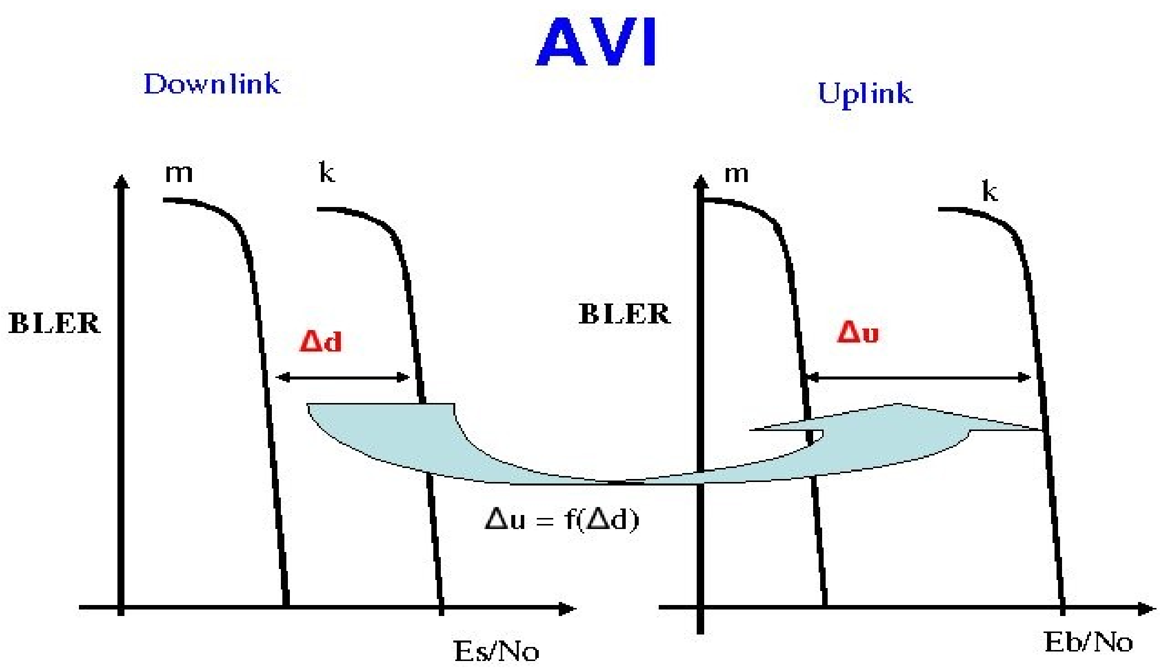

Figure: 4.2. Estimating link level results for different MCS in the uplink.

Figure 4.2 shows an overview of the method to obtain uplink AVI tables from using the AVI tables of a previous downlink study. The offset in the downlink AVI tables (Δd) is used to obtain the offset in the uplink AVI tables (Δu) and calculate the required curve (k) from the reference curve (m).

The AVI tables generated for this work are for the data rates for which the configuration of their radio access bearers is given as example for measurement channels in [2]. Table 4.1 show the calculated uplink offset Δ(Eb/No) for a given rate using the given parameters and modulation (BPSK or QPSK). The reference uplink AVI curve is for a 60 Kbps DPDCH bit rate, with DPCCH/DPDCH power ratio of -5.46 and coding rate 1/3 using BPSK.

DPDCH Rate (Kbps)	***DPCCH/DPDCH power ratio (dB)***	***Coding Rate***	***Offset (dB)***	***Modulation***
60	-5.46	1/3	0	BPSK
240	-9.54	1/3	-0.63	BPSK
480	-11.48	1/3	-0.79	BPSK
960	-11.48	1/3	-0.79	BPSK
2*960	-11.48	1/3	-0.79	QPSK

Table 4.1. Physical layer parameter for different data rates

The AVI tables are calculated using the obtained offsets of Table 4.1. and the result are shown in Table 4.2.

BLER	60Kbps	240Kbps	480Kbps	960Kbps	1920Kbps
1.000	1	0.37	0.21	0.21	0.21
0.900	1.7	1.07	0.91	0.91	0.91
0.800	2	1.37	1.21	1.21	1.21
0.300	2.5	1.87	1.71	1.71	1.71
0.100	2.6	1.97	1.81	1.81	1.81
0.025	3	2.37	2.21	2.21	2.21
0.010	3	2.37	2.21	2.21	2.21
0.002	3	2.37	2.21	2.21	2.21
0.001	3	2.37	2.21	2.21	2.21

Table 4.2. AVI tables BLER vs Eb/No (dB) for different data rates considering a DPCCH.

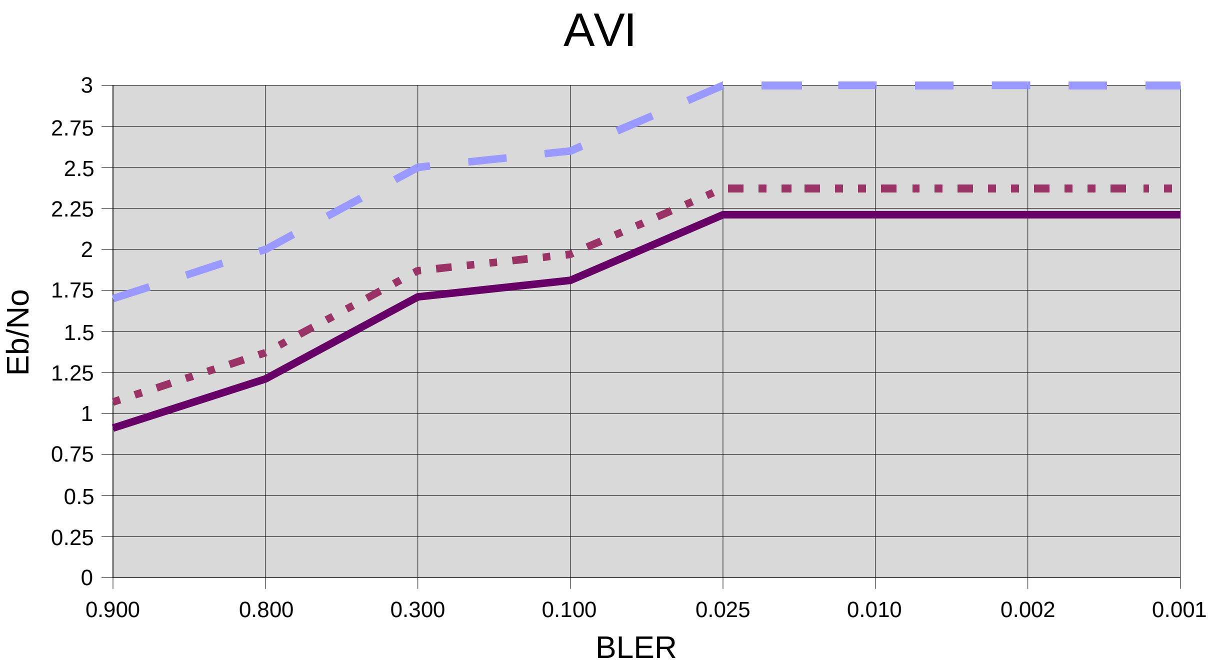

Figure 4.3. AVI tables (BLER vs Eb/No) for different data rates considering a DPCCH.

4.2 Modeling of HARQ

With HARQ different transmission of the same data frame received at the NodeB are combined to increase the probability of correctly decoding the packet. The modeling of HARQ and soft combining is done by a method described in [17] in which the received Eb/No is cumulated from all the received transmission of the same packet. This cumulate Eb/No is used as an input of the AVI tables to obtain the probability of error after combining. The accumulated Eb/No is just the summation of all the Eb/No received, the Eb/No of the first packet plus the Eb/No of all the retransmissions. The method also include a loss that is added to the cumulated Eb/No, this indicates that a perfect gain is not archived and a certain loss is produce depending on the combining technique (chase combining or incremental redundancy). Since it was shown in [17] that combining with incremental redundancy provides a small throughput gain compared to chase combining there will be no differentiation between the two techniques in the modeling of HARQ and the loss obtained by the combining technique will be kept constant. The values for that loss are given in [17] for a QPSK modulation at 3Km/H Pedestrian-A and Vehicular-A profiles with different coding rates. The value 0.93 (-0.32dB) was chosen for chase combining with no incremental redundancy gain, coding rate 1/3 and Pedestrian-A profile which is the worst case that gives the higher loss.

For this work the HARQ was modeled by adding the Eb/No of the retransmissions of the same packet considering the combining loss. The cumulated Eb/No is then used to obtain the probability of error form the AVI tables.

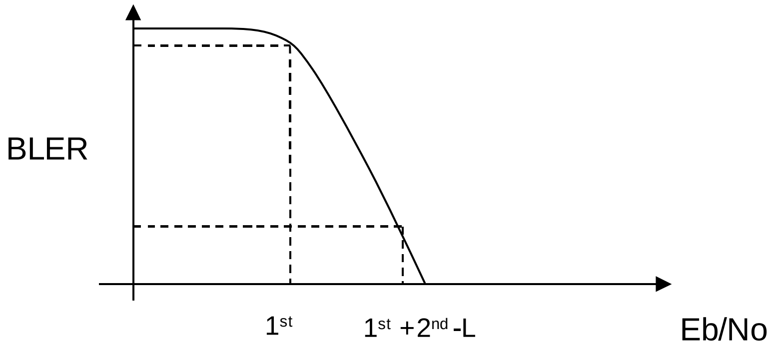

Figure 4.4. Modeling HARQ

In figure 4.4 is shown that the BLER will get lower when a retransmission is received and the Eb/No of the first transmission is added to the Eb/No of the retransmission. The ideal gain gain is not reach since the combining loss and the loss produce by the control channel are considered.

4.3 Modeling of NodeB Packet Scheduling Signaling

Packet Scheduler based in the NodeB must use certain signaling with the UE. In the uplink direction the needed information is included as part of the MAC header and in the downlink direction separate channel are used to transmit the grants and the HARQ acknowledgments. In this model the uplink signaling overhead of the information included in the MAC header is taken into account but the overhead of of the signal transmited in a separate dowlink channel is not taken into account. The delay of these signaling is simulated considering the transmission time and processing times in the NodeB and UE MAC layer.

The processing times were obtained from [23] which suggest 8 slots processing time in the NodeB from reception of E-DPCCH/E-DPCCH to the transmission of E-AGCH/E-HICH/E-RGCH. The UE processing time is of 4 slots form the reception of a E-AGCH/E-HICH/E-RGCH to the transmission the E-DPCCH/E-DPDCH.

The delay of signaling form the UE to the NodeB includes a transmission delay of one TTI (3 slots for 2ms TTI) plus the NodeB processing time of 8 slots. The delay of the signaling form the NodeB to the UE includes a transmission delay of one TTI (3 slots for 2ms TTI) plus the UE processing time of 4 slots. Note that transmission are made every TTI so for example a grant reception at the UE will be decoded in 4 slots but the start of the transmission will have to wait until the beginning of the next TTI (after 6 slots for 2ms TTI). In the same way a reception of E-DPCCH/E-DPDCH at the NodeB will be decoded after 8 slots and the data can be send up to the upper layers but the transmission of an acknowledgment and/or a grant will have ti wait until the beginning of the next TTI (after 9 slots for 2ms TTI). Figure 4.5 shows a diagram of the signaling delay including transmission delay and processing times.

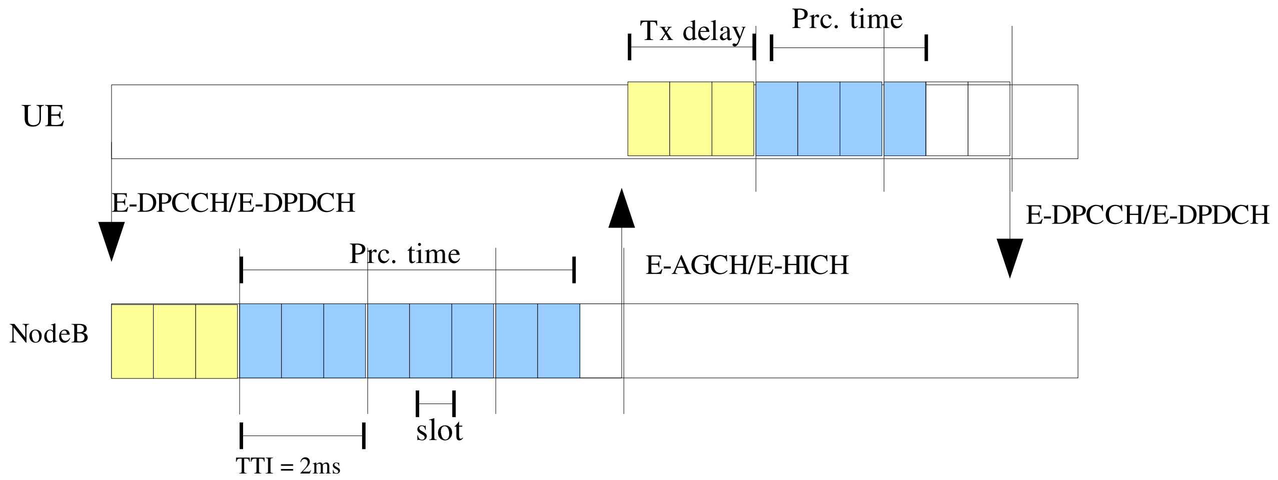

Figure 4.5 Signaling delay between UE and NodeB.

The buffer status information, as specified in [12], is transmitted as part of the MAC-e header so it will be transmited in the E-DPDCH, the happy bit is included in the E-DPCCH, the absolute grants are transmited in the E-AGCH and the HARQ acknowledgments are transmited in the E-HICH. Base on the transmission and processing times, the total signaling delay for each signaling message is calculated.

Having the necessary information form the UEs the NodeB schedules the UEs for transmission according to the policies of the scheduling algorithm.

4.4 Mobility

Mobility is included to be able to change the distance loss of the UEs from the base station during the simulation time. An area of coverage is established within a radius of 3Km in which the UE are trapped and they can not get out of it. At the beginning of simulation the UEs are assigned with a random positions within the cell. Every TTI the UEs calculate their new position and it will depend on their speed, which can be determined as a parameter of the simulation (all UEs will move at the same specified speed 3, 50 or 120 Km/H). UEs that reach the limits of the cell will bounce with the border going back and moving in the opposite direction staying always within the limits of the cell. The base station has a fixed position.

The number of meters traveled in one TTI are calculated form the UE speed and the angle of the movement direction is initialized with a random number at the beginning of the simulation and it only changes when the UE reaches the border of the cell.

4.5 Path Loss

The modeling of the path loss includes the distance attenuation, shadowing and fading. The distance attenuation or distance loss (dl) is calculated using the Okumura-Hata propagation model for a suburban macro cell with base station antenna height of 30m, mobile antenna height of 1.5m and carrier frequency of 1950MHz [20]:

$$dl=129.4+35.2\,log10(d) \tag{9}$$

where d is the distance form the NodeB in Kilometers.

The shadowing modeling of the path loss is based in the study and simulation done in [21]. It is calculated taking into account that its value is not independent form one location to another and there is a correlation with respect to the location.

$$S(x+\Delta x)=a\cdot S(x)+b\cdot\sigma\cdot N \tag{10}$$

$$a=\exp(-\Delta x/D) \tag{11}$$

$$b^2=1-a^2 \tag{12}$$

Here x is the distance to the NodeB and **Δx** is the difference between the actual distance and the distance measured at the previews calculation of the shadowing. The shadowing in the new position **S(x+Δx)** depends on the shadowing of the previous position S**(x).** In the formula (11), D is the correlation distance which is equal to 40m. In (10) is the standard deviation in suburban areas which its typical value is 8dB and N is a random variable that satisfies the standard normal distribution.

Multi-path fading is precalculated in a link level simulation for different environments and UE speeds. The multi-path fading model of this link level simulation is described in [29] and assumes the usage of an ideal Rake receiver. This simulations provide the values of the multi-path fading contribution to the loss experienced by a UE for a period of 2000 seconds. The results of these calculations are used as an input file for the system level simulator. Every TTI all the UEs will use the same file to extract its corresponding fading value but during initialization they begin reading the file at different random points. The implementation of the model allows that each UE can be specified to use a different file, this can be useful for simulations where each UE moves at a different speed.

4.6 E-DCH in EURANE

Enhanced UMTS Radio Access Network Extensions for ns-2 (EURANE) comprises the main functionality of the Release '99 version of UMTS. The new HSUPA was created as a new MAC layer which supports the new features of the Enhanced Uplink. This new MAC layer was include in the existing functional nodes of the simulator (the User Equipment and the Base Station) allowing them to support a new Enhanced uplink transport channel, the Enhanced Dedicated Channel (E-DCH).

4.6.1 Link Implementation Details

The new E-DCH is a dedicated uplink channel and its architecture in EURANE is equivalent to the one of a normal DCH described in [26]. The only differences between them relies in their MAC entity. A new MAC entity was created in order to support all the functionalities of the E-DCH.

As for a DCH, before a E-DCH is created, the *node-config* utility should be used to specify its uplink parameters and its RLC transmission mode. The E-DCH was specified to work with TTI values of 2ms and 10ms. Also new parameter like the scheduling mechanisms and the maximum noise raise level or Noise over Thermal (RoT) must be specified. This is an example command to create a E-DCH using a an Acknowledge Mode (AM) in the RLC with a uplink bandwith of 384kbs and TTI of 2ms, the Scheduling is setup to Rate Scheduling (see section 6.3.1), the value 31 is given as the random positioning seed (RP_Seed) and the maximum noise level is 6dB:

```
$ns node-config -llType UMTS/RLC/AM \
-dowlinkBW 384kbs \
-uplinkBW 384kbs \
 -downlinkTTI 2ms \
```

```
    -uplinkTTI 2ms \
    -Eul_Scheduling 1 \
   -RP_Seed 31 \
    -MaxRoT    6
set edch0 [$ns create-edch $ue1 $agent1]
```

This command creates the E-DCH in $ue1 and attach $agent1 to it.

After creating the channel the file which includes the fading values for the simulation must be specified. This file must contain the fading values obtained for a specific environment and UE speed.

```
$ue1 setErrorTrace-EUL 1 “Ped_A-3kmh-0m-2000s-Uenr1”
```

This command sets the file “Ped_A-3kmh-0m-2000s-Uenr1” for being used by the UE “ue1”, this command must be repeated for all the UEs in the simulation, even if all them use the same file.

Figure 4.6 shows an example TCL construction of a single UE connected to a NodeB. The FACH and RACH channel are created automatically and an agent is attaches to a E-DCH. A return channel in the downlink direction is created automatically using a normal DCH.

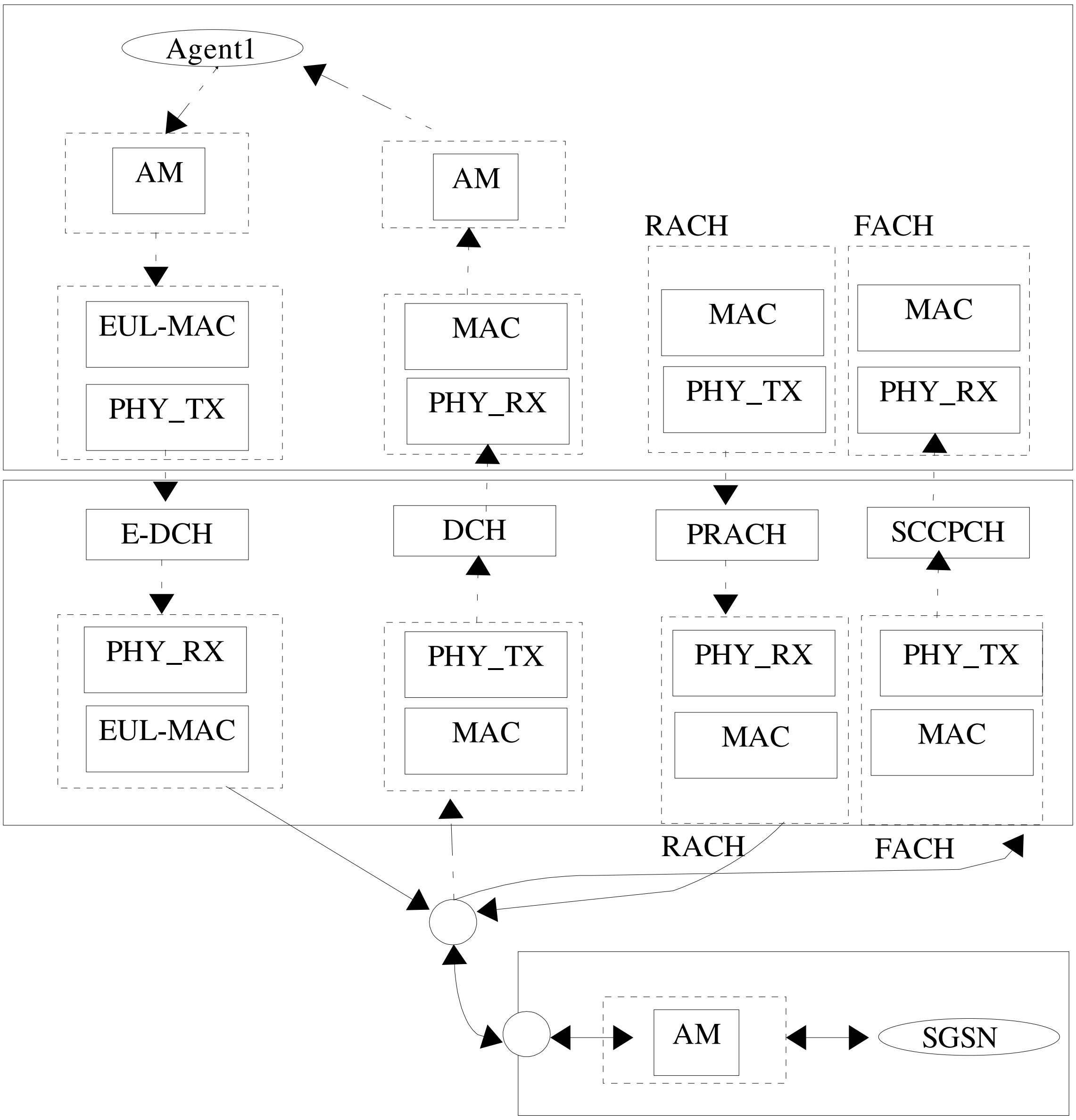

Figure 4.6 TCL construction of a single UE connected to a NodeB.

4.6.2 **Trace Support**

A new packet types where created in order to simulate the Enhanced uplink in ns-2. The "MacE" packet type encapsulates all the AM data packets into a big frame which is transmited in the E-DPDCH. A retransmission of this packet is defined as "MacE_HARQ" packet type. These packet types can be visualize in the output trace file of the simulation.

To be able to visualize the simulation results in the trace output file some extra commands must be executed. To visualize all the incoming events (reception of packets) by the UE and the NodeB the command "trace-inlink-Eul" will add this events in the trace output file. Similarly the command "trace-outlink-Eul" will add to the trace output file the arrival of packets into the MAC buffers and the transmission of the packets.

The RLC sequence number of shown in the output trace file of a "MacE" packet type specifies the smallest RLC sequence number of the packets encapsulated in that frame.

The next example will add to the trace file of the simulation all the arrivals of packets to the MAC buffers, the departure of the packets of the buffers (transmission of the packets) and the reception of those packets in all the UEs and NodeB.

#Create trace *for {set i 0} {$i < $numUsers} {incr i} { $ue_($i) trace-inlink-Eul $f 2 $ue_($i) trace-outlink-Eul $f 2 $bs trace-inlink-Eul $f [expr $i + 2] $bs trace-outlink-Eul $f [expr $i + 2]*

}

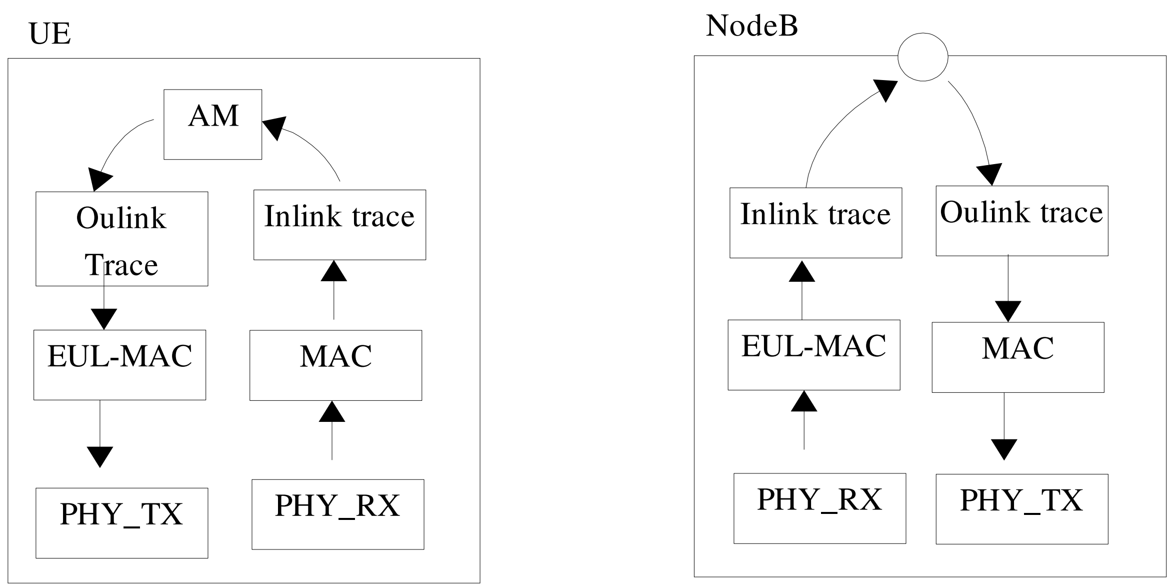

Figure 4.7 Trace support.

5 NodeB packet Scheduling

The packet scheduling main function is to share the available air interface resources between the users. To improve the performance of the uplink dedicated transport channel, the scheduling functionality was moved to the NodeB and the scheduling algorithms use the time division and code division approach (see section 4.3) to allocate bit rates every TTI to the active users. The NodeB scheduling algorithms are based on the uplink scheduling information which is a requests coming from the UEs giving the scheduler information about their current buffer status and power capabilities.

The main purpose of the scheduling is to allocate resources to the UEs in the most optimal way. In the uplink a pure time division scheduling where a single user transmits at a time will not be a very efficient way because UEs are power limited and a single UE will not be able to take advantage of the full uplink capacity. Because of this reason the uplink scheduling combines the time division scheduling with a code division scheduling so that more than one UE can transmit at the same time in order to maximize the utilization of the uplink channel.

Depending on the scheduling algorithms used the allocation of resources can maximize the system throughput at a price of being unfair or can allocate resources fairly to all the UEs at the cost of decreasing the overall system throughput. In order to maximize the system throughput the scheduling algorithm must take into account the channel conditions experienced by the UEs which are requesting for resources. Using this information the scheduling algorithm can make a decision of the UEs that must be scheduled first and can also predict the amount of resources that the UE is able to support so that no extra resources are allocated to UEs that are not able to take advantage of them.

In order to see the impact of different scheduling mechanisms in the uplink performance several mechanisms were implemented in the simulator. All the algorithms perform a basic functionality for the resource allocation process but depending on the prime mechanism of the scheduling some of the basic functionalities are performed in a different way.

5.1 Scheduling Algorithm Basic Functionality

The packet scheduling is executed every TTI. It's first function is to receive all the rate requests in the uplink scheduling information received from each user. Depending on the scheduling algorithm a bit rate will be assigned to each UE that requests resources for that TTI and that has packets waiting in its buffer. All the UEs are ordered in a list that gives priority to the request of the first UE in the list, the sorting of this list depends on the scheduling algorithms used. The scheduler will try to grant the assigned bit rates of all UEs depending on the amount of interference and the block error rate required for each transmission. If those requirements are not achived a reduction of the bit rate for the last UE in the list will take place. If the minimum bit rate for a UE is reached that UE is eliminated from that scheduling period and it will not be allowed to transmit during that TTI, a new request for that UE is automatically generated at the NodeB to be taken into account in the next TTI.

Upon the reception of a request for resources form the UEs the scheduler assigns each UE with a bit rate. The assignation of bit rates to each UE can be done using an estimation of the maximum bit rate that can be supported by each UE or can be just an assignation of the maximum allowed bit rate, this assignation will depend on the scheduling mechanism used.

To have a measure of the amount of allocated resources to each user the load factor of each UE is calculated:

$$\eta=\frac{1}{1+\frac{W}{(Eb/No)\cdot R}} \tag{13}$$

where W is the chip rate, R is the assigned bit rate and Eb/No is the required energy per bit over noise ratio for a specific block error rate (this value is taken from the AVI tables).

The total uplink load factor then is calculated as the sum of the load factors of the active UEs:

$$\eta UL=(1+i)\cdot\sum\frac{1}{1+\frac{W}{(Eb/No)\cdot R}} \tag{14}$$

The interference from other cells is included with the parameter i which is the other-to-own interference ration. Since the simulator use a single-cell model, this parameters is kept constant during the simulation.

Using the total uplink load factor the total received wideband interference power is calculated:

$$Itotal=\frac{PN}{1-\eta UL} \tag{15}$$

where PN is the thermal noise.

Using the the total uplink interference power (Itotal) and the load factor of each UE the power that each UE needs to transmit at the assigned bit rate is calculated:

$$Pj = Itotal \cdot \eta UL \tag{16}$$

After the bit rate assignation the UEs are sorted in a list where its order will depend on the scheduling mechanism policies. Since not all the UEs can be scheduled to transmit at the same TTI, the order of the list will determine the priority of each UE to be scheduled for that particular TTI.

Two conditions must be satisfied during the allocation of resources by the scheduler. First, considering the estimation of the interference that will be generated by all the UEs transmitting at the assigned bit rates, the scheduler will calculate if the Eb/No requirements will be fulfilled by all the UEs.

The scheduler can verify that each user meet the required Eb/No using its power (Pj) and the total interference (Itotal):

$$Eb/No = \frac{W}{R} \cdot \frac{Pj}{Itotal - Pj} \tag{17}$$

If one of the UEs does not meet this requirement the process of decreasing the bit rate of the UE at the end of the scheduling list will take place. This process will continue and if the UE at the end of the list is eliminated the process will continue with the next UE on the list. If at least one UE doesn't meet the Eb/No requirements this condition will fail.

The second condition is to keep the noise ratio below the established limit setup by the RNC. The noise ratio is defined as the amount of interference generated over the thermal noise.

$$NR = \frac{Itotal}{PN} \tag{18}$$

The NR (noise ratio) is set as a fixed value of 6 dB. If the calculated NR is bigger than the specified value the process of decreasing the bit rates of the last UE on the list will continue.

If the two conditions are satisfied all the UEs in the sorted list will be granted with the assigned bit rates. If one of these conditions fails the scheduler will start the process of rate decreasing and UE elimination.

The process of rate decreasing and UE elimination will be executed over the UE at the bottom of the sorted list. This process will start decreasing the bit rate of this user trying to get to the point where the two previous conditions are attained. When a UE reaches the minimum allowed bit rate assignation this UE will be eliminated and will not be granted with resources for transmission during that TTI and its request will be postponed for being considered in the next TTI. Depending on the scheduling mechanism policies an eliminated UE could get a higher priority for being scheduled in the next TTI.

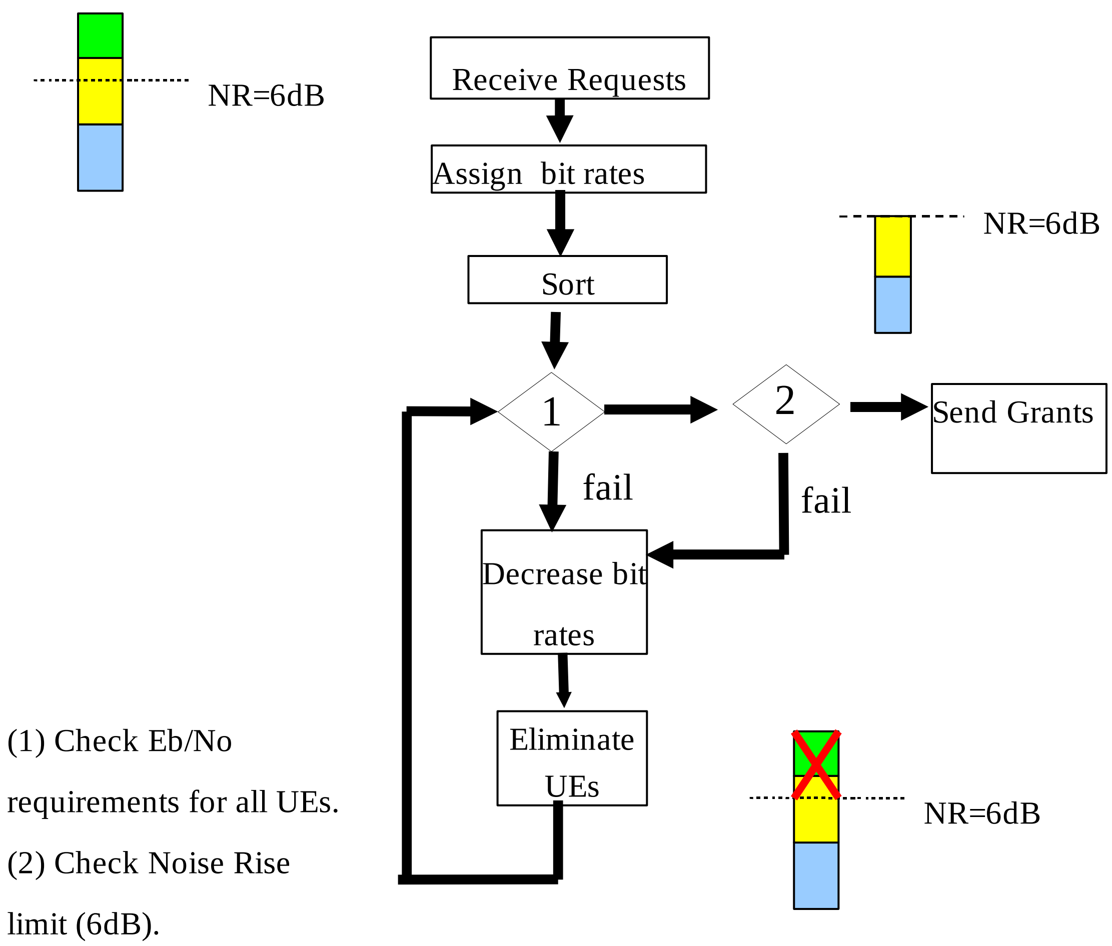

Figure 5.1: Scheduling Algorithm Basic Functionality.

5.2 Code/Rate Scheduling

The basic principal of the code or rate scheduling is to serve all users at the same time. As the amount of users increase each user will be granted with a lower rate to be able to share the total amount of resources between all the users.

The basic scheduling functionality is performed with some changes in the procedure of decreasing of bit rates and user elimination. All the users that request for resources are assigned with the maximum allowed data bit rate. If the required Eb/No of all the that have to be scheduled is not satisfied or if the noise ratio exceeds the established limit the bit rate of the user at the bottom of the scheduling list will be decreased. Immediately the unfortunate user will be positioned at the top of the list so if the two scheduling conditions are still not satisfied another user will be the selected for bit rate reduction. When many users request for transmission for the same TTI it is possible that all of them reach the minimum allowed bit rate and still the scheduling conditions are not satisfied; in that case elimination of users will take place in the same order as the bit rate reduction was being executed.

5.3 Round Robin Scheduling

The basic principal of the Round Robin scheduling is to give priority to a different user in a round robin fashion every TTI. A pure time division scheduling giving all the resources to a single user will be inefficient since UEs are not able to take advantage of the full channel capacity so more than one user will be scheduled every TTI but one of them will have the priority to transmit at the maximum allowed bit rate.

All the users that request for transmission will be assigned with the maximum allowed bit rate at the beginning of the scheduling process. If the scheduling conditions are not satisfied the process of rate reduction will be executed over the user at the end of the scheduling list, the priority of this user will not change during that TTI and if the scheduling conditions are not yet satisfied the bit reduction will continue affecting this user until it reaches the minimum allowed bit rate and its eliminated for transmission during that TTI. A user affected by the bit reduction or elimination process will have priority for the next TTI and will be positioned at the top of the scheduling list. With this mechanism every TTI a different user will be granted to transmit at the maximum allowed bit rate.

5.4 UCQI based Scheduling

The basic principal of this scheduling mechanism is to give priority to the users with best channel conditions. The Uplink Channel Quality Indicator (UCQI) will be relative to the amount of bits that a user can transmit during that TTI and the power it will use for transmitting them. The highest UCQI will determine the user which can transmit highest amount bits with the lowest power. The amount of bits to be transmitted in one TTI is determine by the highest allowed Transport Format Combination (TFC) which specifies the maximum transport block size and which is assign to all the users that request for transmission. The power use by each UE will be determine mainly by its path loss, the calculation of this parameter can be done at the NodeB using the received power and a measurement of the transmitted power of each UE (measurement available at the NodeB every TTI [8]).

Using the UCQI as a sorting parameter the users are organized in the scheduling list where the top of the list is occupied by the user with best channel conditions and the bottom of the list will be occupied by users with worst channel conditions.

If the scheduling conditions are not satisfied the process of rate reduction will be executed over the user at the end of the scheduling list and will continue affecting it until it reaches the minimum allowed bit rate and eliminated for transmission during that TTI. A user affected by the bit reduction or elimination process will not have priority for the next TTI and the scheduling list will be sorted again depending on the new values of the UCQI. With this mechanism every TTI the user with best channel conditions will have the priority to use the highest allowed bit rate and the channel with worst channel conditions will be the most likely to be eliminated and not allowed to transmit. This is an unfair scheduling mechanism and a user in a bad channel conditions will be affected and will just be allowed to transmit when the users with good channel conditions don’t request for resources.

5.5 Rate Estimation Scheduler

The basic principal of this scheduling mechanism is to predict the maximum bit rate that UEs are allowed to support and keep the fairness using a round robin fashion. When scheduling users with bad channel conditions the assignation of the maximum allowed bit rate can be a waste of resources since those UEs will need higher power that could be out of their capabilities.

In this model all the UEs where assume to have the same power capabilities. The prediction of the maximum supported bit rate is base on simulations made with one single UE experiencing different path loss through the cell area. The results of these simulations are used to classify each UE in a range according to the path loss that is experiencing and assign it with the corresponding bit rate.

Rate (Kbps)	Max Path loss (dB)
1600	118
1200	120
680	122
480	124
300	126
160	128

Table 5.1: Maximum path loss for bit rate support.

In this scheduling mechanism a user that has the highest priority according to the round robin fashion can be allocated with a lower bit rate than other users with lower priority because the bit rate assignation will depend on the experienced path loss of each user. The reduction and elimination process will behave as in the implementation of the Round Robin scheduling.

6 Simulations and Results

Using the model implemented in the simulator several simulations were executed to study and analyze the behavior and the impact of the NodeB packet scheduling mechanisms on the enhanced uplink performance.

The simulation time was established to 400 seconds and each simulation was repeated six times with different seeds to initialize the random position in the cell and the traffic generation. All the mean values are obtained using the data gathered in the six simulations and the 95% confidence interval (95% CI) is given for the mean measurements.

BLER	*10%*
Noise Rise Target	6dB
TTI duration	2ms
Max UE transmission power	26dBm (-4dB)
UE speed	3Kmph
Cell radius	1.5Km
Other-to-own Interference ratio	0.6

Table 6.1: Simulation Parameters.

6.1 Scenario 1

During the simulations using this scenario 24 UEs are initialized in a random position within the cell area. Each user transmits data according to the Pareto traffic model at 250kbps during burst periods. The mean burst time and mean idle time of each UE are set to 5 seconds.

The Pareto On/Off Traffic Generator is a traffic generator application that generates traffic according to a Pareto On/Off distribution. Packets are sent at a fixed rate during on periods, and no packets are sent during off periods. It uses a constant packet size, and the on and off periods are taken from a Pareto distribution with mean values entered as parameters for mean burst time and mean idle time.

Under the specified characteristics of this scenario different simulations were made using the different implemented scheduling algorithms.

6.1.1 **Rate Scheduling**

In these simulations the rate scheduling algorithm was used for packet scheduling in the NodeB. This algorithms try to schedule all the UEs to transmit at the same time using small rates. Since there is a large number of users in the cell the NodeB receives every TTI several requests for resources from different UEs. The rate scheduling algorithm allocates small bit rates to all the users that request for resource so that all of them are able to transmit a small amount of information during the same TTI.

This way of allocating resources result in very low rates when there is a high load of users. In the simulation this behavior can be seen showing the utilization Transport Block Size (TBS) used by the UEs. The TBS is the number of bytes that a UE transmits during one TTI. As expected for the rate scheduling algorithm the minimum TBS of 40 bytes is used all the time for all the UEs. In this case UEs with a lot of data in their buffers and good channel conditions are restricted by the scheduler to just transmit 40 bytes during one TTI since the resources were shared between all the UEs that requested for transmission during that TTI. Figure 6.1.1 in the appendix A shows the simulation results for the TBS utilization using the rate scheduling algorithm.

Figure 6.1.2 shows two instances during the simulation period where in every TTI it is shown the UEs that were scheduled for transmission. With two different instances of the simulation it can be shown that the apparent deterministic behavior of allocation of resources per TTI is just for small periods of time but the randomness of the Pareto traffic generator will make that the UEs request the NodeB for resources at random times.

The NodeB packet scheduler will try to keep the noise rise level as close as possible to the target of 6 dB specified for the simulation and in this case, using the rate scheduling algorithm, on average nine UEs are scheduled for transmission during the same TTI. There is no preference for any particular UE regardless of its channel condition or priority level and all uses receive on average the same amount of resources.

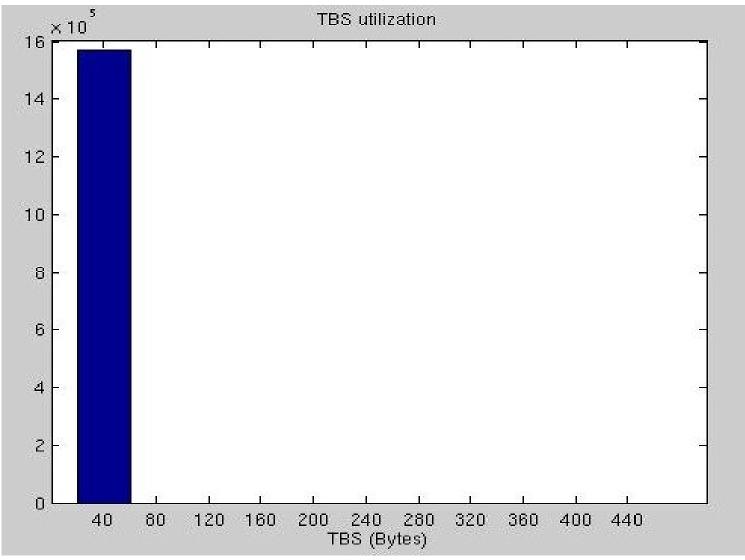

Figure 6.1.1: TBS utilization using the rate scheduling algorithm.

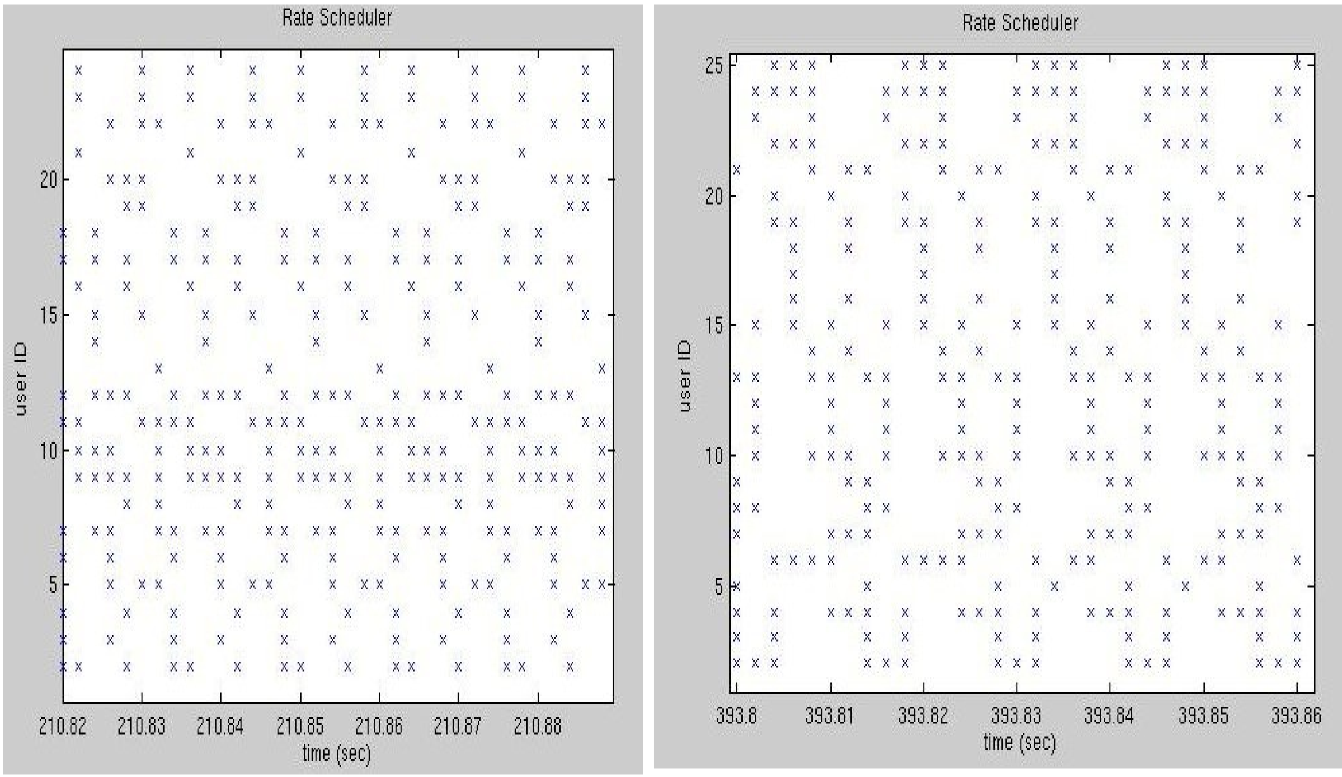

Figure 6.1.2: UEs allocated per TTI with Rate Scheduling

6.1.2 **Round Robin**

In these simulations the Round Robin scheduling algorithm was used for packet scheduling in the NodeB. This algorithm gives priority to a UE allowing it to transmit at maximum rate in one TTI. The scheduler will choose the UE with highest priority every TTI in a round robin fashion. As explained in the last chapter a strict time division multiplexing which schedules one UE to transmit at a time is not a good option since one single UE will not be able to take advantage of all the capacity of the channel. Because of this reason more than one UE will be able to transmit in the same TTI and after allocating the maximum rate to the UE with highest priority (according to the round robin fashion) the remaining resources will be used to schedule more UEs.

Figure 6.2 shows an instant during the simulation where it is shown for every TTI the UEs that were scheduled for transmission. It is easy to see the Round Robing mechanisms behavior where on average two UEs are scheduled to transmit every TTI. Since on average two users will share the whole channel capacity during one TTI each UE is allocated with high data rates, it will only depend on the UE power limitations if it is able to use the allocated resources or not.

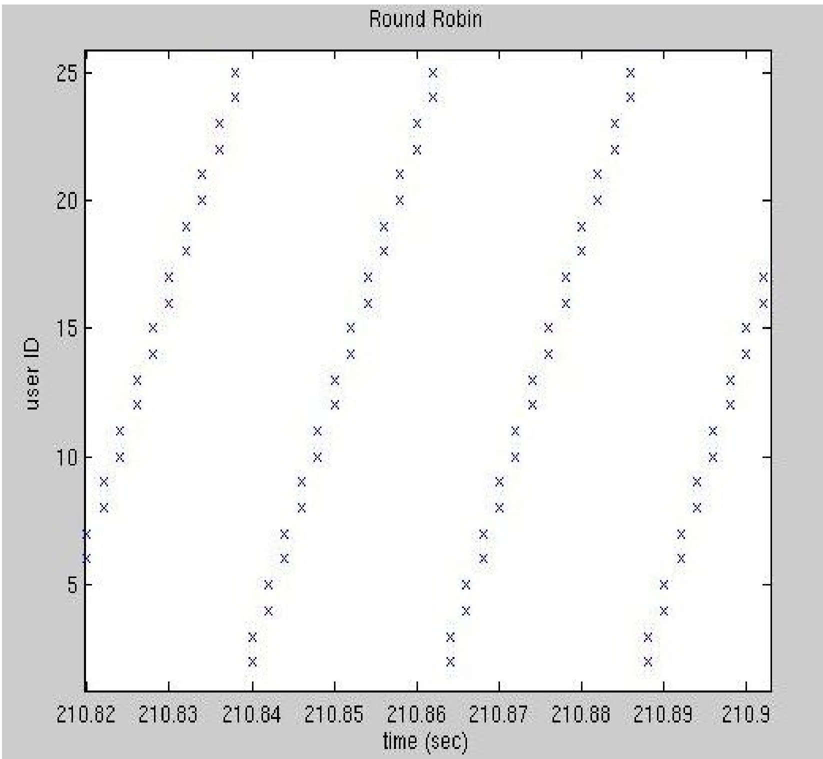

Figure 6.2: UEs allocated per TTI with Round Robin Scheduling.

In general high data rates will be allocate to each UE. Figure 6.3 shows the utilization of TBS during the simulation and as expected the maximum allowed TBS of 400 bytes is the most highly used. The TBS of 280 bytes has also a high utilization because it is the TBS that the UEs allocated with the remaining resources will use. This TBS value of 280 bytes depends not only in the target noise rise level but also in the inter-cell interference and in the packet size of the RLC packets arriving to the Enhanced MAC (for these simulations the RLC packet size was setup to 40 bytes). The minimum TBS was setup to allow the transmission of at least one RLC packet of 40 bytes plus its corresponding header. TBS of 40 bytes is also highly used during the simulation because UEs that fall into bad channel conditions caused by fading are not able to transmit at high rates any more and they are just able to transmit a single RLC packet.

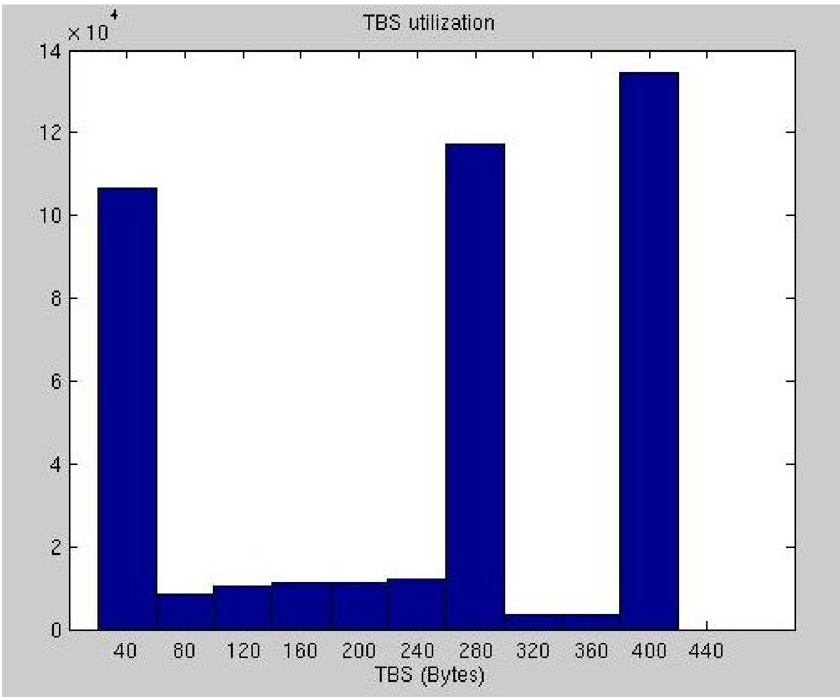

Figure 6.3: TBS utilization using the Round Robin scheduling algorithm.

6.1.3 **Scheduler based on UCQI:**

In these simulations the a scheduler based on a uplink Channel Quality Indicator was used for packet scheduling in the NodeB. This algorithm gives priority to the UEs with the best channel condition during that TTI.

This is an unfair scheduling mechanism since the users with bad channel conditions will not get the same amount of resources than users with good channel conditions. A bad channel condition user will only be scheduled for transmission in a TTI where no user with better channel conditions request for resources. This particular behavior will trigger a repeating request mechanism in the bad channel conditions UEs. When a UE sends a request for resources the information about its buffer status, headroom and UCQI are attached into the Enhanced MAC header. If after a period of time the UE has not receive any grant from the NodeB the UE will repeat its request for resources sending a packet without payload and just attaching the necessary information to the header. This repetition of request is not likely to happen when using fair schedulers since the grant for a request will usually arrive to the UE before the timer expires, but using unfair scheduler like the one base on UCQI a bad channel condition user will have to wait until there are no more UEs asking for resources or until its channel conditions improve. With a noise rise level of 6 dB this type of scheduler will fulfill all the available resources with just two users at maximum rate but figure 6.4 shows that in some cases three or four UEs are scheduled for transmission during one TTI. The reason of this behavior is that in this case what we are seeing in figure 6.4 is the repetition of requests mechanism of the bad users. On average two users are being scheduled with TBS of 400 bytes and TBS of 280 bytes filling the available resources but also the timer of the bad users expires since they are not receiving any grants and they transmit an empty packet. This empty packet, without payload, is assumed to have no significant impact interfering other transmissions but it is still printed in the output trace file of the simulations as a transmission of zero bytes.

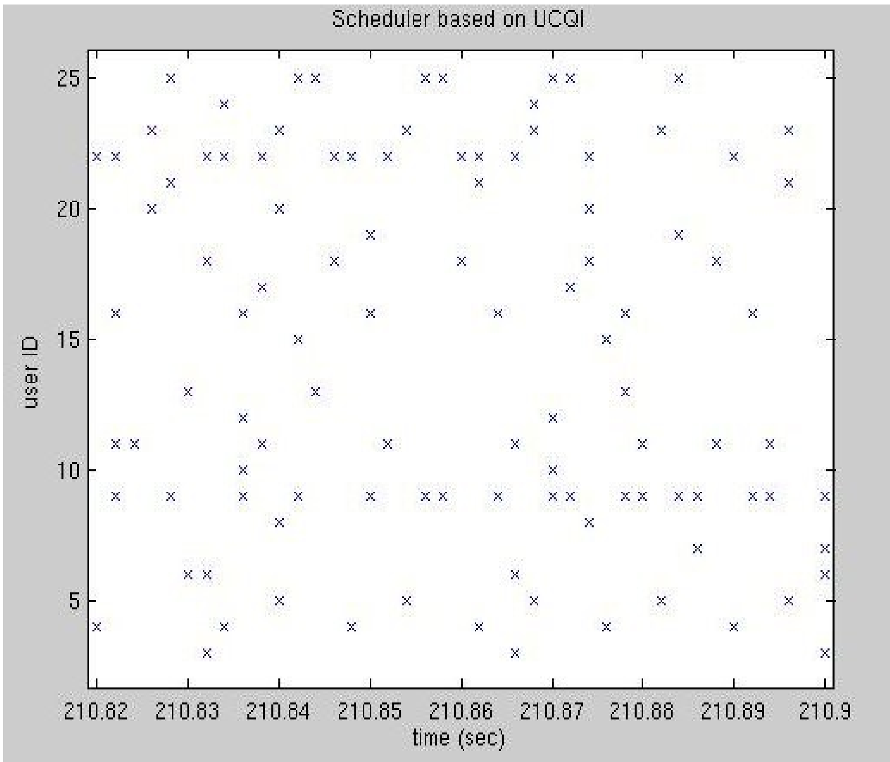

Figure 6.4: UEs allocated per TTI with UCQI based scheduling.

Most of the scheduled UEs in this simulation have good channel conditions when they receive a grant and most of the times they are able to use all the resources they were allocated. Figure 6.5 show that the big TBS are the most highly used which means that most of the times large amount of bytes are being transmitted in one TTI. The smallest TBS are used when no good channel conditions users are active and bad channel conditions users are scheduled. It is also possible that the channel conditions of a user change drastically in a short period of time, shorter than a round trip time (8ms for 2ms TTI), so when the information about the channel conditions was sent from the UE to the NodeB it reports good channel conditions but at the time of reception of the grant this conditions have changed. In the simulator the only factor that changes significantly in a round trip period is the fading but in reality other sources of interference must be taken into account basically the interference caused by UEs in another cells. This is one of the main factors why Adaptive Modulation and Coding (AMD) was not considered for being implemented in the uplink since the constant variation of the channel conditions doesn't allow a good channel condition estimation and ruin the performance of AMD.

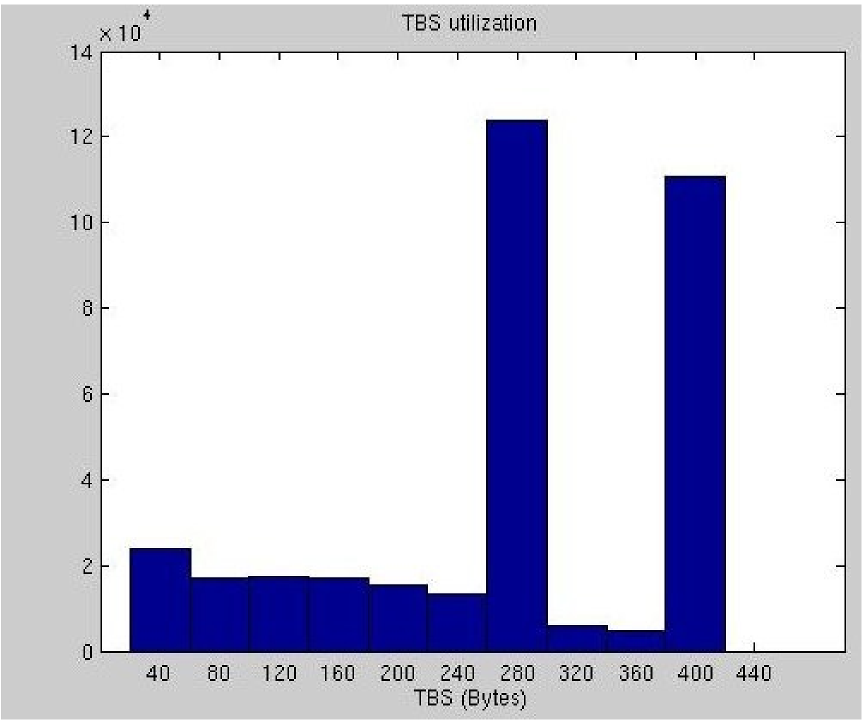

Figure 6.5: TBS utilization using the UCQI based scheduling algorithm.

6.1.4 Scheduler with Rate Estimation:

In these simulations the a scheduler use a rate estimation mechanism using the path loss experienced by each user to calculate or estimate the maximum rate that it will be able to support. This algorithm doesn't give priorities to the UEs and they are scheduled in a Round Robin fashion but their allocated rates will depend on the estimation of its maximum supported rate. With this algorithm UEs will not be assign with more resources than what they are able to support in an ideal condition with no interference of other users and no inter-cell interference.

In Figure 6.6 the Round Robin fashion can be noted. The difference between figure 6.6 and figure 6.2, which show a simple Round Robin scheduling with no rate estimation, is that more than two UEs are scheduled for transmission during the same TTI. In this case the users are not sending an empty packet repeating a request for resources, all users are being scheduled with at least the minimum TBS but users that experience a higher path loss are allocated with smaller rates. The allocation of smaller rates for some users allows the scheduler to allocate the remaining resources to other users.

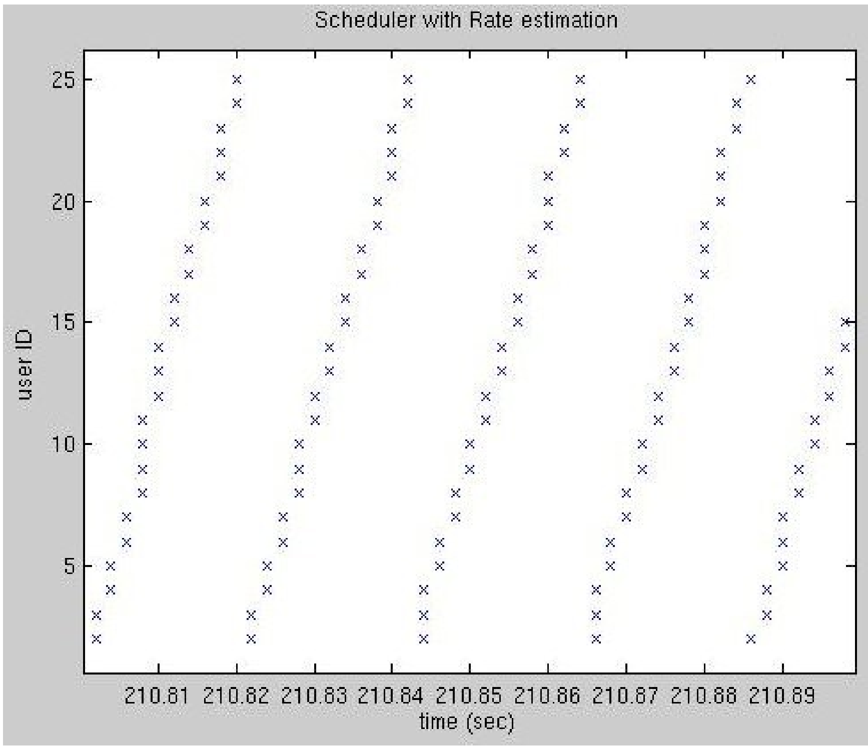

Figure 6.6: UEs allocated per TTI with Rate Estimation scheduling.

The more accurate allocation of resources estimating the maximum rate that a UE is able to support lead to a better utilization of the channel wasting less resources allocated to UEs that are not able to use them. In figure 6.7 the TBS utilization shows an increment of the maximum TBS of 400 bytes and a reduction of the minimum TBS of 40 bytes. Also an increment in the utilization of TBS of 80 bytes and 160 bytes is shown. These TBS were not likely to be used with the other schedulers because the allocation of high rates to the users with more priority will not leave resources to allocate small rates to other UEs. These TBS were only used when bad channel conditions users were scheduled with high rates but their power limitations make them reduce the rate. With the rate estimation algorithm users with bad channel condition or high path loss are allocated with small rates by the scheduler.

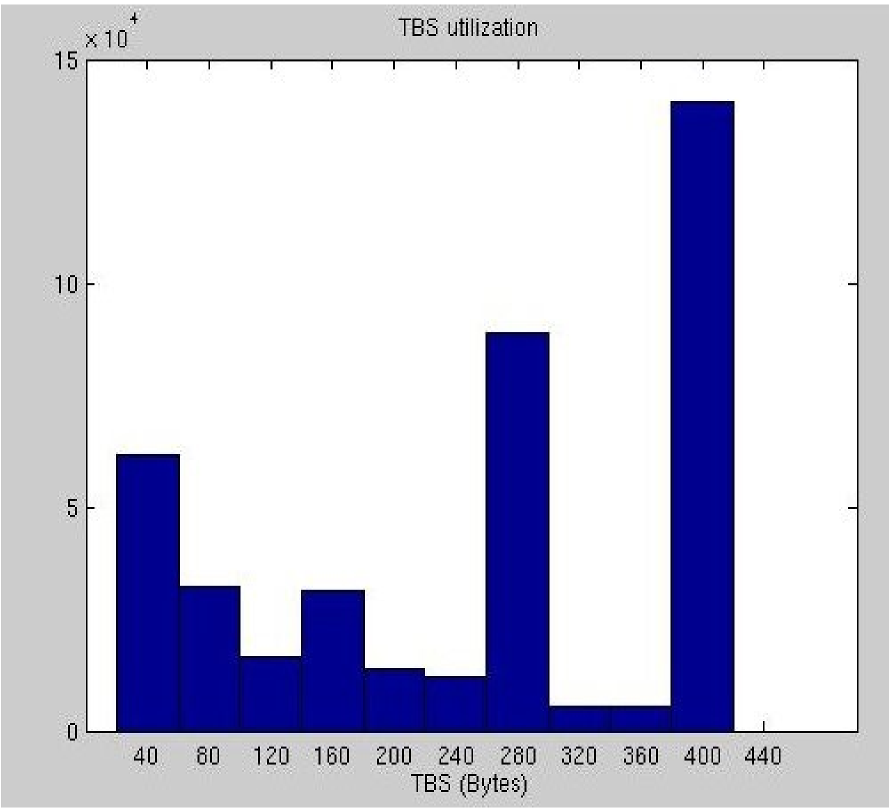

Figure 6.7: TBS utilization using the Rate Estimation scheduling algorithm.

6.1.5 **Time Delay:**

Depending on the type of traffic it is very important to consider the delay experienced during the data transmission. In some applications the destination is not expecting the data within a certain time and basically the delay is not so important. Other application are sensitive to the delay and they impose a limit of acceptable delay to achieve acceptable quality.

The behavior of the different scheduling mechanism affect the time delay experienced by the users. The delay measurements are taken from the time at which a packet arrives from the RLC to the MAC of the UE until that packet is delivered to the RLC by the MAC at the NodeB side.

Table 6.2 gives the mean and standard deviation for all the implemented schedulers during the simulation time. Figure 6.8 show the delay experienced by all the received RLC PDUs during the whole simulation for the different schedulers. The highest average delay is given by the Rate Scheduling which allocating small rates to all the users doesn't give enough resources to the UEs to transmit large amounts of data. The data in their buffers is transmitted in small chunks every time they receive a grant from the NodeB having to wait until the next grant to transmit again. The data that arrives to MAC from the RLC also comes in bursts of packets, the first packets in this burst are transmitted very quickly, using the first low rate grant given by the NodeB, but the last packets of the burst will have to wait a long time in the buffer before they are transmitted increasing the average delay measurement and the standard deviation.

The UCQI based scheduler shows the smallest average delay but with the highest standard deviation. The average delay is small because most of the times good channel conditions users are scheduled with high rates which they use to transmit large amount of the data in their buffers reducing the time they have to wait to be transmitted. The fact that the bad channel conditions users are just scheduled when the good channel condition users are inactive makes that these bad users experience large delays. In figure 6.8 large delay measurements are shown for the UCQI scheduler when the bad users finally had the chance to transmit their data.

On the Round Robin and the Rate Estimation schedulers the UEs will have to wait until they get their turn to transmit in the round robin fashion but when they get a grant they will be able to transmit large amounts of data from their buffers, that is why the average delay of these schedulers in higher than for the UCQI based scheduler but lower than for the Rate Scheduler. The improvement in the more accurate allocation of resources of the Rate Estimation scheduler is shown as a smaller average delay and standard deviation compared to the Round Robin scheduler.

	Mean	Std	Mean 95% CI
Rate scheduling	0.356	0.22	0.354 to 0.357
RoudRobin	0.222	0.16	0.218 to 0.225
Rate Estim Sch	0.207	0.14	0.201 to 0.214
UCQI	0.157	1.72	0.153 to 0.160

Table 6.2. delay measurements of different scheduling schemes.

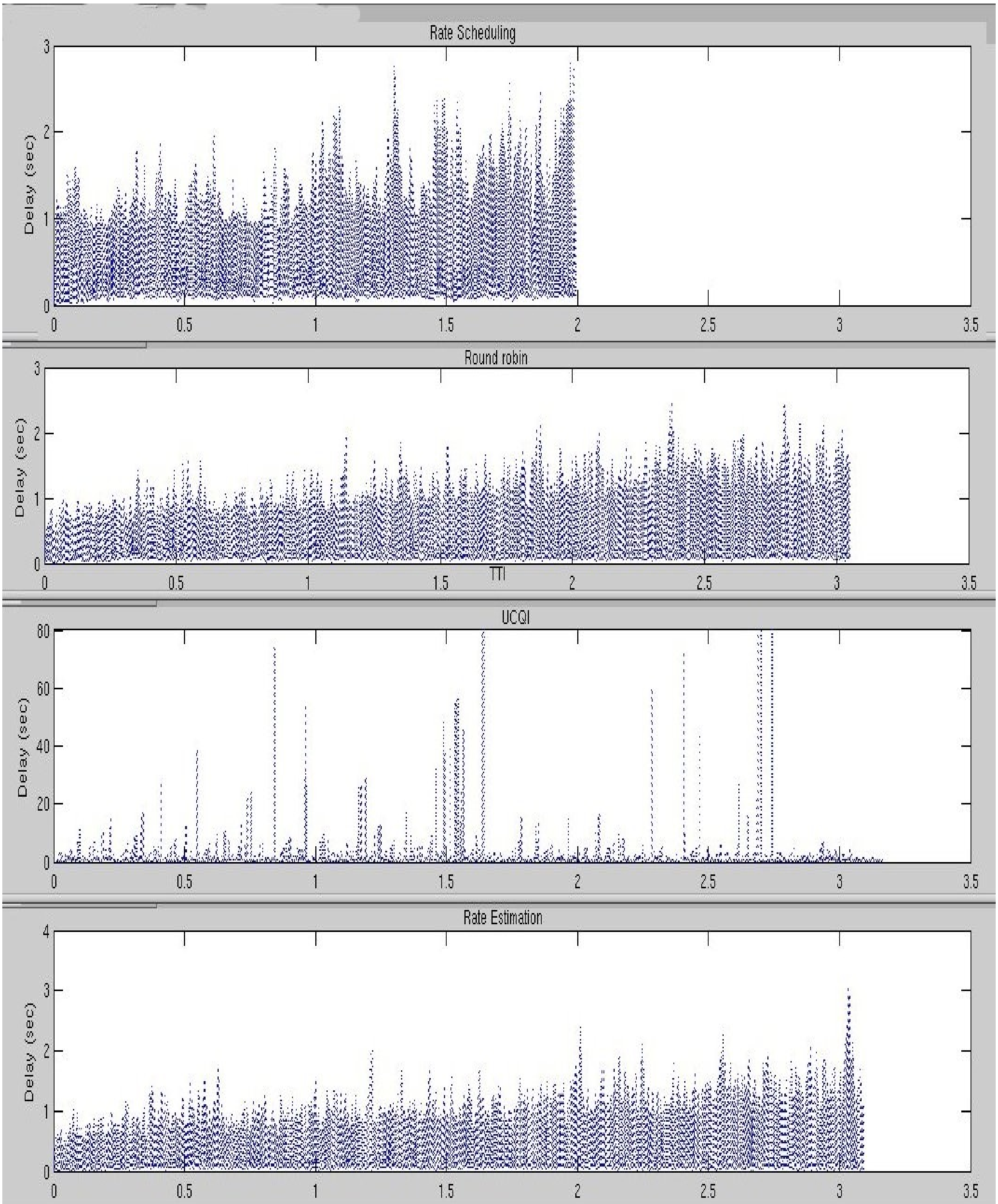

Figure 6.8: delay measurement of different scheduling schemes. X axis indicates the received RLC PDUs (x10^6).

6.1.6 **Total Transmitted Power:**

The scheduling policies that rule the behavior of the different scheduler will have a big impact in the performance regarding power consumption and interference generated to other cells. Measuring the transmitted power of the UEs it is possible to get an overview of the impact of the different schedulers in the power consumption and inter-cell interference.

Every TTI the power used by the UEs to transmit their data is measured. The total power (the addition of the powers of all the UEs transmitting in the same TTI) is shown in figure 6.9. Table 6.3 show the average total power and standard deviation for the different schedulers.

The measurements are expressed in decibels so the most negative value implies a lower power usage. As expected the UCQI based scheduler uses the lowest amount of power during the simulation time since the users with best channel conditions will require less power to transmit lager amounts of data. The other schedulers don't take the channel condition into account so bad channel condition UEs will use more power to transmit their data. The Rate scheduling use the most power since many users transmit their data at the same time causing more interference between each other which means that they will need to transmit at higher power to reach their target Eb/No. Also, the lower bit rates allocated in the Rate scheduler require a higher Eb/No and therefore a higher power.

The Round Robin scheduler and the Rate estimation scheduler perform very similarly because in both cases all the UEs will transmit in their corresponding turn in the round robin fashion regardless if they require a high or a low power for their transmission. Since the Rate estimation scheduler allocates resources to more than two users per TTI, depending if their rate estimation gives small rates so that the remaining resources can be used by other users, the power used is a little higher than the Round Robin scheduler. Nevertheless, the Rate Estimation scheduler have a gain in terms of througput and delay compared to the Round Robin scheduler since the Rate Estimation scheduler will not waste resource in UEs that can not use them.

In terms of power consumption the UEs will get benefit out of the UCQI scheduler. Basically, the UEs that fall into bad condition because of the fading will not be scheduled for transmission conserving their power during the bad conditions period and being scheduled for transmission when their conditions get better. Also in the opposite way, when the fading contributes to the correct reception of the data, the UCQI scheduler will schedule these UEs to transmit taking advantage of the positive fading peak.

The low power usage of the UCQI scheduler also implies that on average the amount of interference generated to other cell will be much lower than with the other schedulers. The whole system will have a big gain in this aspect because receiving lower inter-cell interference will make the UEs reach their Eb/No requirements with less power allowing them to transmit at higher rates or save power.

	Mean	Std	Mean 95% CI
Rate scheduling	-3.59	2.5	-3.65 to -3.52
RoudRobin	-5.46	3.05	-5.536 to -5.378
UCQI	-9.58	7.1	-9.810 to -9.350
Rate Estim Sch	-5.36	3.04	-5.472 to -5.255

Table 6.3: Total transmitted power (dB) measurements of different scheduling schemes.

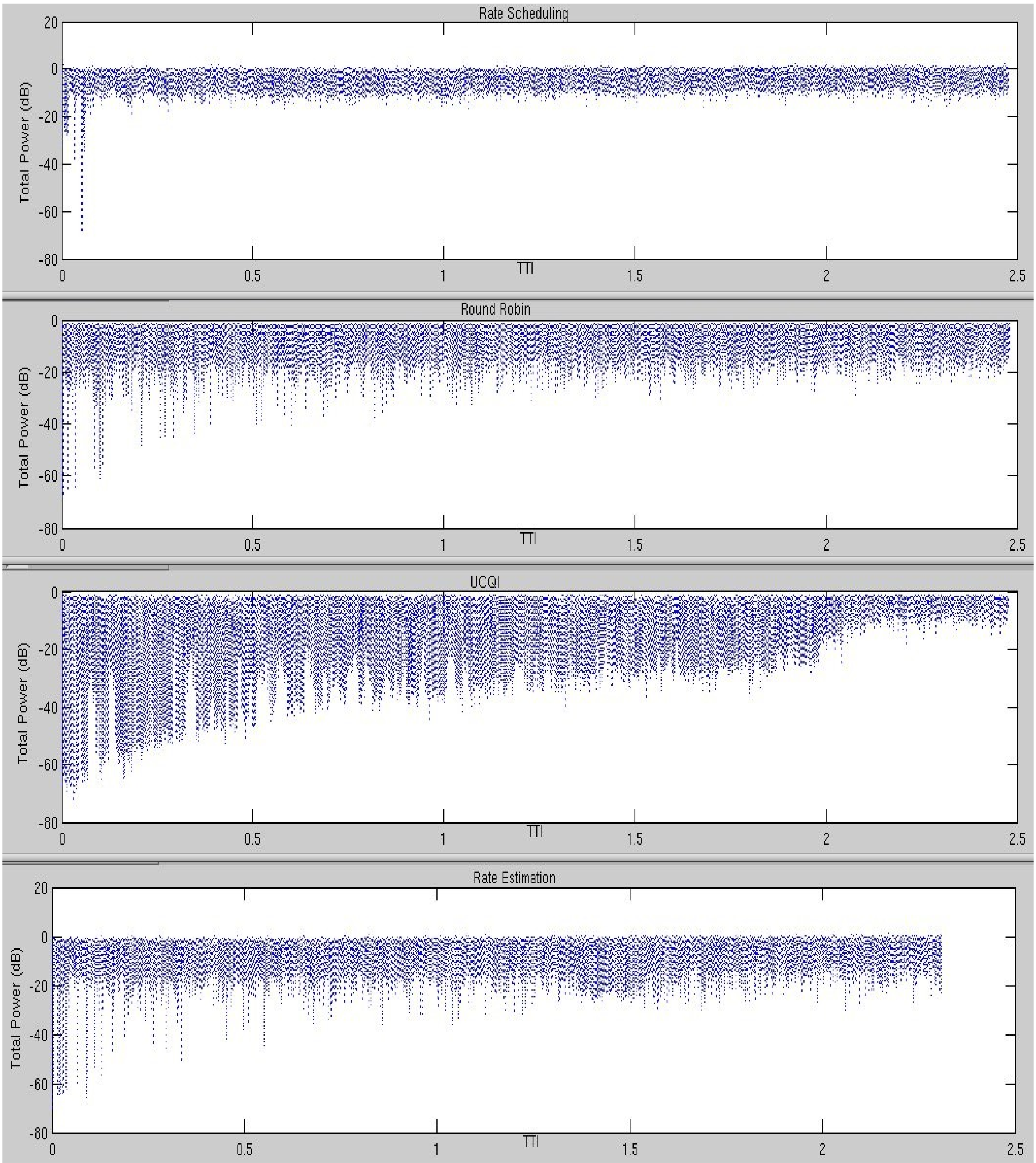

Figure 6.9: Total power measurement of different scheduling schemes. X axis x10^6.

6.1.7 **Throughput:**

The average cell throughput measured for the different schedulers is shown in figure 6.10.

Scheduling many UEs with low rates as it is done by the Rate scheduler shows to have the worst performance in terms of cell throughput. The main reason of this behavior is that as more users are scheduled for transmission during the same TTI they will generate more interference between each other, that means that every user will experience more interference from the other users in its own cell and they will need to use more power to reach their Eb/No requirement. The contribution to the noise given by each UE will be higher and the scheduler will reach the target noise rise (6 dB) sooner.

The other schedulers allocate users with high instantaneous bit rates giving all the available resources to just a few users. These schedulers allow the users to experience less interference so they can use their power to transmit at high rates. Also the fact that higher bit rates require lower Eb/No allows the users to use less power to reach the same performance. These facts make that the amount of bits transmitted per TTI becomes larger when scheduling a few users with high instantaneous bit rates than many users with low instantaneous bit rates.

The UCQI based scheduler gives priority to the users that have the best channel conditions and users with bad channel conditions have to wait until the conditions improve or until the other users become inactive. This way of scheduling presents a gain in the average cell throughput compared to a simple Round Robin scheduler where the conditions of the channel are not taken into account. With the model implemented in the simulator it is possible to see the gain given by the UCQI scheduler caused by the better utilization of the channel resources since good channel conditions users are schedueled more often and with high instantaneous bit rates. The Round Robin scheduler will allocate high instantaneous bit rates to all the users in a round robin fashion where it is possible that this allocation is given to a user with a bad channel condition. This bad channel condition user will not be able to transmit at such a high instantaneous bit rate and the resources will be wasted.

The advantage of the UCQI scheduler in terms of total transmitted power can not be translated into the cell throughput gain with the single-cell model of the simulator. As shown before, the UCQI scheduler will produce lower inter-cell interference compare to the Round Robin scheduler but the simulator use a single-cell model where the inter-cell interference doesn't depend on the scheduler behavior since there are no other cells modeled and the inter-cell interference is kept constant. If all the neighboring cell use the same type of scheduler the inter-cell interference using UCQI scheduler will be lower (results given in [Rosa] show an average of 0.2 other-to-own interference ratio for a scheduler with the same principle in a multiple cell simulator) than the one assumed for this study (0.6 other-to-own interference ratio) and we could expect a significant increase of the average cell throughput.

The small difference between the UCQI scheduler and the Rate Estimation scheduler shown in figure 6.10 is cause by the better utilization of the channel resources when the good channel condition users are not active. In those cases the UCQI will allocate bad channel users with high instantaneous bit rates which they will not be able to use. On the other hand the Rate Estimation scheduler will predict the maximum rate that the bad users are able to support allocating them with smaller rates leaving resources to share with other bad users. The lack of multiple cells and inter-cell interference model give us this result but we can expect when neighboring cell use the same scheduler the UCQI scheduler will have better average cell throughput than the Rate Estimation scheduler.

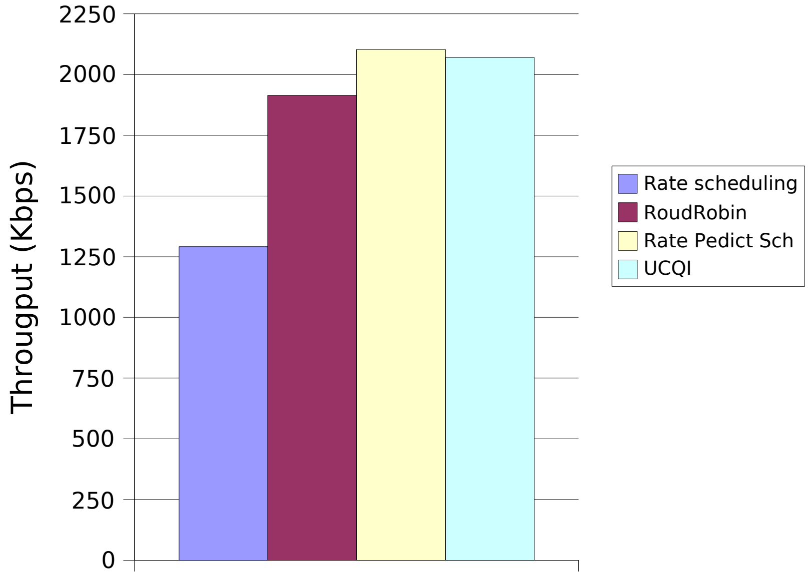

Figure 6.10: Average cell throughput for different schedulers.

	Mean	Mean 95% CI
Rate scheduling	1291	1281 to 1301
RoudRobin	1914	1894 to 1934
Rate Estim Sch	2069	2086 to 2119
UCQI	2102	2055 to 2084

Table 6.4: Mean cell throughput and 95% CI

To analyze individual UE throughput measurements the results are obtained from simulation were the seed for random positioning in the cell is kept constant so that it is possible to distinguish good and bad channel condition users. Their condition will change gradually while they move around the cell or while they experience the different fading and shadowing contributions but they will not jump form very bad to very good conditions (which could happen when changing the positioning seed and users are positioned close to the NodeB instead of at the border of the cell). Table 6.5 show the throughput measurements for all the users with the different schedulers.

The throughput of all the users are affected by their channel conditions but also by the scheduler being used. Lets take user 14 as an example of a bad channel condition user and user 2 as a good channel condition user. In the rate scheduling user 14 and user 2 don't have a significant difference compared to the difference in the UCQI based scheduler. Regardless of their channel conditions the users in a Rate scheduling get similar small throughputs during the simulation. On the other hand, users in a UCQI based scheduler get very high throughput measurements if they are in good channel conditions but very low throughput measurements if they have a bad channel condition.

In the Round Robin scheduler bad users are also affected by the fact that they have to wait for their turn in the round robin fashion to transmit again while in the Rate scheduler they transmit more often since they do it at the same time with many other users. But good users got benefit in the Round Robin scheduler because when they receive a grant to transmit they can do it at a very high instantaneous bit rate.

The Rate estimation schedule behaves in the same way as the Round Robin with a small gain given by the better utilization of the resources caused by the prediction of the maximum rate that each user is able to support.

User ID	Rate scheduling	RoudRobin	Rate Estim Sch	UCQI
1	59041	53763	56760	43275
2	61479	116008	131381	165902
3	45341	37268	42125	8088
4	66036	105354	111937	94192
5	37937	26774	29101	9813
6	52814	92495	101220	151386
7	51008	103479	114312	124003
8	45683	73064	85043	87104
9	62768	88203	78116	60558
10	43417	56549	63866	47043
11	51219	106645	127605	123923
12	43621	41730	44512	8269
13	53019	105155	124884	129968
14	35861	28414	29578	5423
15	59288	91330	104012	155480
16	55140	113120	112811	129546
17	59106	71265	87108	106639
18	54487	115016	129827	110705
19	46248	55299	65285	55336
20	59488	116109	123896	116523
21	55436	41435	46920	24679
22	59838	114594	126198	119668
23	35647	29665	32388	9697
24	83882	104559	111750	155242

Table 6.5. Throughput measurements per UE in bits per second.

6.2 Scenario 2

During the simulations using this scenario 12 UEs are initialized in a random position within the cell area. Each user transmits data according to the Pareto traffic model at 760kbps during burst periods. The burst time and idle time of each UE is setup to 5 seconds.

Under the specified characteristics of this scenario different simulations were made using the different implemented scheduling algorithms.

The simulation results using this scenario show concordance with the results with scenario 1. The simulation results with lower amount of users but with a higher traffic load help us to ensure the analysis of average cell throughput made for scenario 1. Figure 6.11 shows the results for the average cell throughput for the different schedulers. A similar behavior compared to scenario 1 is perceptible but with a small difference regarding the Rate estimation scheduler and the UCQI scheduler. As explained for scenario 1 the difference between these two schedulers rises when the good channel condition users are not active and the turn comes to the bad channel conditions to be scheduled. In scenario 2 the high traffic implies that good channel conditions users will not have so much idle periods during the simulation as in scenario 1 and the gain of the Rate estimation scheduler will disappear in this case.

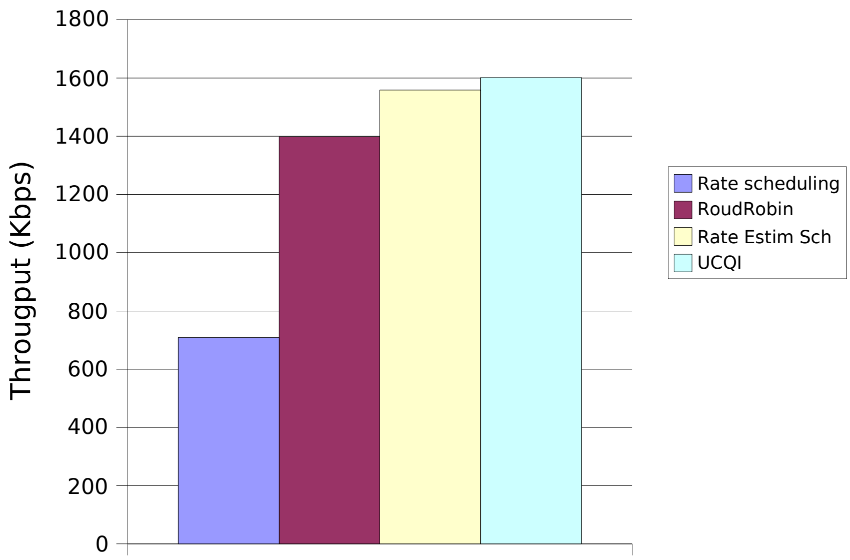

Figure 6.11: Average cell throughput for different schedulers. Scenario 2.

	Mean	Mean 95% CI
Rate scheduling	914	890 to 938
RoudRobin	1803	1734 to 1871
Rate Estim Sch	2010	1971 to 2049
UCQI	2066	2034 to 2099

Table 6.6: Mean cell throughput and 95% CI

6.3 Scenario 3

During the simulations using this scenario 24 UEs are initialized in a random position within the cell area. Each user transmits data in a constant bit rate at 64kbps. With a packet size of 2080 bytes the MAC layer will receive bursts of 40 bytes every 26msec form the RLC layer.

The results of these simulations are governed by the particular and not very realistic parameters of this scenario.

Using this scenario with low traffic rates per UE the simulation results show a disadvantage of using the UCQI scheduler. One of the reason for this behavior has to be with the fact that in the cell area a bad condition user can still transmit a single packet of 40 bytes during one TTI. When the traffic rate is very low there is not much difference, in terms of the amount of transmitted bits, when scheduling a good or a bad user. When the good channel condition user is scheduled with high instantaneous bit rate it will not use the resources since it just have a small amount of data in its buffer. The bad channel condition user will not be able to transmit at a high rate because of its power limitations but transmitting at a low rate it will be enough to transmit the small amount of data in its buffer. Because of this reason the UCQI and Round Robin scheduler will tend to perform in a similar way.

Nevertheless, the results still show a small gain in the cell throughput of the Round Robin compared to the UCQI scheduler. That behavior can also be attributed to the particular traffic source. Good channel condition users in the Round Robin scheduler get some benefit because of the peculiarity of the traffic source because while they wait for their scheduling turn more data arrives to their buffer and they can make a single transmission at a high instantaneous bit rate rather than many small transmission. The UCQI scheduler will keep scheduling good channel condition users before their buffer gets bigger and they will only use the resources to transmit small amount of data while the bad user accumulate data in their buffers but will still need many transmission since they don’t have enough power capabilities to transmit lager amounts of data.

	Mean
Rate scheduling	-3.41
RoudRobin	-5.8
Rate Estim Sch	-6.58
UCQI	-7.89

Table 6.7. Total transmitted power (dB) measurements for different scheduling schemes.

The gain of the UCQI regarding total transmitted power is still considerable in this scenario as shown Table 6.7. This lower transmitted power implies lower interference generated to other cell. With the model of the simulator is not possible to see that gain in terms of average cell throughput but we should keep in mind that if the neighboring cell use also a UCQI scheduler the average cell throughput of the UCQI scheduler will have a significant gain. Even if the neighboring cells use another scheduler mechanism, the fact that they will receive lower inter-cell interference will allow them to use less power which will also generate lower interference to other cells.

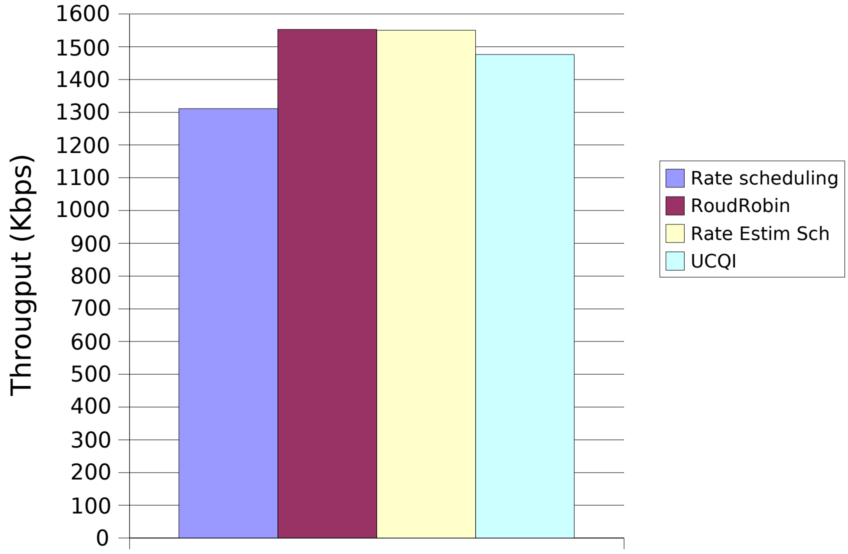

Figure 6.12: Average cell throughput for different schedulers. Scenario 3.

	Mean	Mean 95% CI
Rate scheduling	1311	1300 to 1321
RoudRobin	1553	1537 to 1568
Rate Estim Sch	1551	1537 to 1563
UCQI	1477	1474 to 1478

Table 6.8: Mean cell throughput and 95% CI

6.4 Scenario 4

During the simulations using this scenario 24 UEs were initialized in a random position within the cell area. Different traffic models are used and attached to the UEs. Eight UEs use the FTP traffic model, eight UEs use the HTTP traffic model and eight UEs use a video traffic model.

Table 6.9 and figure 6.13 show some of the results. We can see that the Rate scheduling uses higher power to transmit lower amount of data during the simulation time. On the other hand, the UCQI uses lower power to transmit more data during the simulation. Without taking into account the decrease of inter-cell interference, the UCQI scheduler shows an average cell-throughput similar to the Round Robin scheduler but using less power.

In this scenario the video traffic source pumps bursts of data at regular intervals in a similar way that the CBR traffic model. Since eight of the UEs use the video traffic source the results of the different schedulers are determined mainly by this type of traffic. The average cell throughput results in figure 6.13 are very similar to the ones obtain in scenario 3 using the CBR traffic model.

	Mean	95% CI
Rate scheduling	-7.2	-7.98 to -6.42
RoudRobin	-10.49	-10.98 to -10
Rate Estim Sch	-10.6	-11.44 to -9.77
UCQI	-12.24	-12.82 to -11.66
Priority	-11.05	-11.94 to -10.16

Table 6.9. Total transmitted power (dB) for different scheduling schemes.

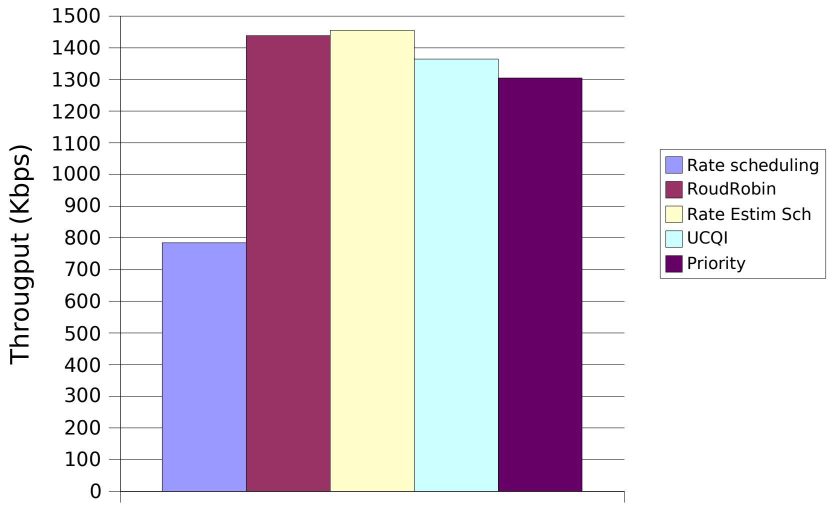

Figure 6.13: Average cell throughput for different schedulers. Scenario 4.

	Mean	95% CI
Rate scheduling	784	707 to 861
RoudRobin	1378	1378 to 1499
Rate Estim Sch	1366	1366 to 1545
UCQI	1294	1294 to 1434
Priority	1253	1253 to 1355

Table 6.10: Mean cell throughput and 95% CI

6.5 Priority Scheduler:

Having this combination of different flows is a more realistic scenario. It should be taken into account that the different types of flows have different requirements. The FTP, for example, is not a delay sensitive application so data being transmitted can wait longer stored in the buffers before its transmission. More strict requirements have to be achieved while transmitting other types of flows like video applications that are delay sensitive and the receiver is expecting the arrival of data within a determine period of time. The scheduler used in the NodeB can affect the requirements of those type of flows. For example the UCQI based scheduler will not allocate resources to a UE using a video application and will give the resources to a FTP user which has a better channel condition. Because of this reason it should be important to consider giving priorities to the different flows.

Even doe it is not supported by the standards, an extra field was included in the header of the Enhanced MAC in the simulator. This field carries information about the priority of the flow being transmitted. With this information the scheduler can make a better decision to allocate resources to the users with higher priority flows. A new scheduler was implemented modifying the UCQI scheduler in a way that the sorting of the users depends on the new value for the priority included in the header and not in the UCQI value.

Video application flows were ranked with the highest priority, the HTTP flows were ranked with a lower priority and the lowest priority was given to the FTP flows. With this assignation of priorities and the new scheduler, the video users will have the priority to be scheduled first and with the highest allowed bit rate while FTP users will not be scheduled unless there are no high priority users active.

Since this priority scheduler doesn't consider the channel conditions, users with bad channel conditions but with high priority will have the priority to be selected for transmission. Those bad users are not able to transmit at a high rate and will use a high power to transmit small amounts of data. This fact decrease the performance in terms of average cell throughput (see figure 6.13) and total transmitted power (see table 6.9) but it will try to keep the quality of the service of the high priority users regardless of their channel conditions.

The average delay per user for the simulations done without changing the positioning seed are given in table 6.11. The results show how the scheduler affects the delay experienced by each user. As an example lets analyze user number 7 which on average experience good channel condition's and as a result it suffers a delay of 0.08sec when using the UCQI scheduler. The traffic flow of this UE is of a FTP application which is not delay sensible and the resources used accomplishing such a low delay could be used on more sensible traffic flows. User number 21 which on average experience bad channel conditions suffers a high delay which will make the quality of its video application unacceptable. If we compare the results of these two users in the UCQI scheduler and in the Priority scheduler we can see that the FTP, beside having good channel conditions, experience a higher delay since the FTP priority was established as the lowest priority and its UE will receive lower amount of resources. The video application user, besides having a bad channel condition, experience a lower delay in the Priority scheduler than in the UCQI since the video flow was established as the highest priority. Figure 6.14 shows the comparison between this two users using the UCQI and Priority scheduler. We can see the improvement in the PDF when a video flow with bad channel condition use the priority scheduler instead of the UCQI scheduler and the the decline of the PDF when a FTP flow with good channel conditions use these schedulers.

	UE ID	Rate scheduling Mean Delay	RoudRobin Mean Delay	Rate Pedict Sch Mean Delay	UCQI Mean Delay	Priorty Mean Delay
	1	3.12	0.46	0.16	0.42	0.54
	2	2.59	0.2	0.18	0.17	0.3
F	3	3.17	0.9	0.47	1	1.9
T	4	0.85	0.13	0.12	0.1	0.23
P	5	3.03	0.14	0.27	0.1	0.48
	6	0.99	0.15	0.21	0.32	0.13
	7	0.83	0.17	0.09	0.08	0.38
	8	3.19	0.17	0.23	0.38	0.34
	9	1.57	0.12	0.1	0.11	0.46
H	10	0.97	0.25	0.16	0.22	0.2
T	11	2.43	0.14	0.08	0.11	0.16
T	12	1.87	0.2	0.51	0.27	0.57
P	13	2.75	0.15	0.1	0.1	0.45
	14	1.37	0.08	0.08	0.05	0.09
	15	1.8	0.09	0.09	0.1	0.19
	16	1.5	0.22	0.08	0.06	0.12
	17	2.9	0.46	0.13	0.52	0.15
	18	2.6	0.14	0.12	0.1	0.08
V	19	3	0.73	0.21	0.84	0.26
I	20	3.1	0.12	0.08	0.08	0.07
D	21	3.8	0.78	0.32	1.25	0.37
E	22	2.6	0.1	0.1	0.13	0.08
O	23	3.01	0.1	0.1	0.05	0.06
	24	2.9	0.09	0.09	0.15	0.08

Table 6.11: Average delay (sec) experienced per user with different schedulers.

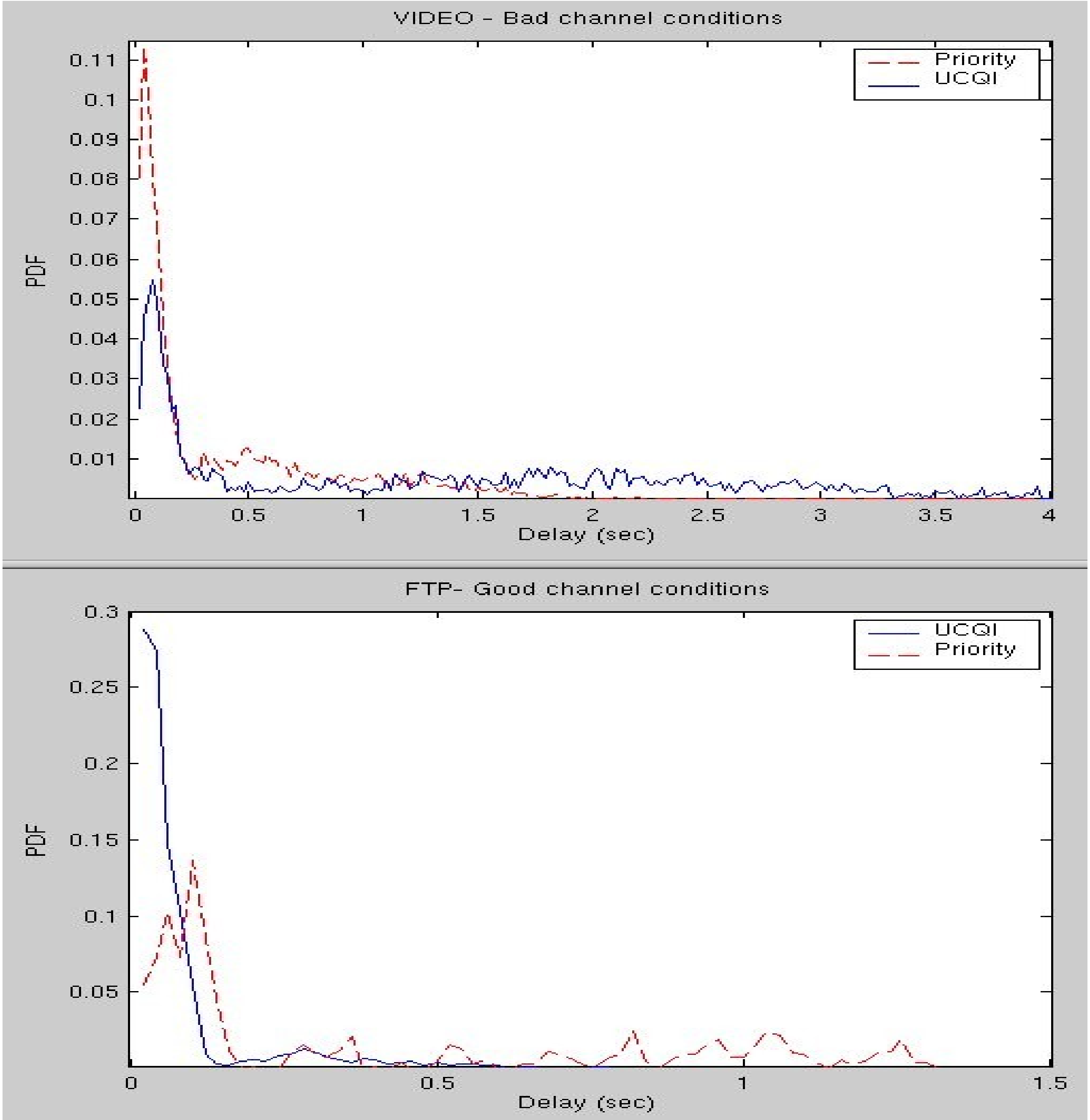

Figure 6.14: PDF for different flow types experiencing different channel conditions

6.6 E-DCH and DCH comparison

During this simulation study the conditions of Scenario 1 were repeated. 24 UEs were initialized in a random position within the cell area. Each user transmitting data according to the Pareto traffic model at 250kbps during burst periods. The mean burst time and mean idle time of each UE were set to 5 seconds.

The aim of this simulation study is to compare and quantify the gain given by the enhancements introduce by HSUPA. To be able to directly compare the two channels the TTI of 2ms of the E-DCH was modified to 10ms since the DCH, as specified in Release 99, doesn't support 2ms TTI. Also, the same scheduler mechanisms (Rate Scheduler) was used in the RNC for the DCH and in the NodeB for the E-DCH. This scheduler doesn't take channel conditions into account so the gain archived by the E-DCH in this simulation will be just given by the reduction in the scheduling delay and by the rapid physical layer retransmission obtained with the HARQ technique.

The average cell throughput of the E-DCH and the DCH are shown in figure 6.15 where we can see a significant gain of 17%. As shown in the simulation results using different types of schedulers at the NodeB, extra gain can be obtained when including channel conditions into account in the scheduling mechanism and when using this information to estimate the amount of resources (maximum rate) that a UE is able to support.

	Mean	95% CI
DCH	682	646 to 717
E-DCH	798	791 to 805

Table 6.12: Mean cell throughput (Kbps) and 95% CI

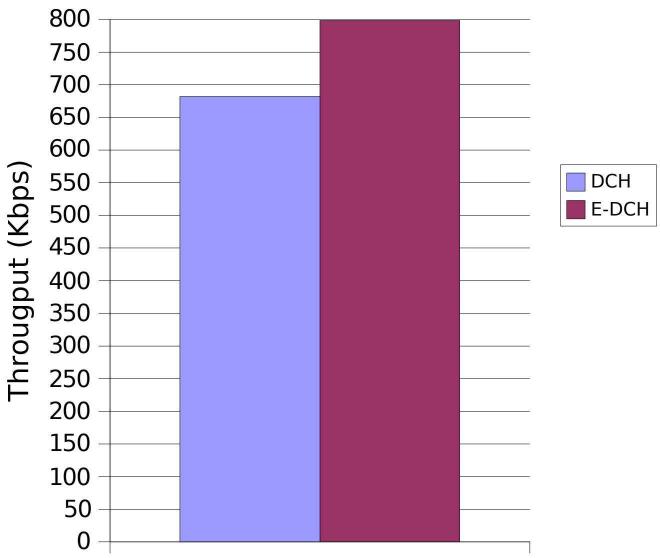

Figure 6.15: DCH and E-DCH average cell throughput.

7 Conclusions

The HARQ and Fast Packet Scheduling techniques introduced for the Enhanced Uplink show to provide significant improvements in performance. As mention in Chapter 3.2.5 studies made in a multiple cells simulator gave a 37% gain compared to normal Dedicated channel with RNC packet scheduler. Also using our single-cell model the results obtained show 17% gain when using a no channel condition dependent schedulers in the RNC and the NodeB.

The simulations and results obtained using different scheduling mechanisms in the enhanced uplink show how the policies that rule the scheduling affect the performance of the system. The results also show that the scheduling mechanisms could affect the required QoS needed for a certain application.

In general the results show us that scheduling many users at the same time with small bit rates, like it is done in the Rate scheduler, is not an efficient method. The performance metrics obtained during the simulation period show that this type of scheduler uses the highest power to transmit the smallest amount of data during the simulation compared to the other schedulers. There are two main reason for this wicked performance of the Rate scheduler. The first one is that for smaller bit rates the required Eb/No tends to be higher which means that more power is need to achieve the same performance. The second reason is that as more users transmit at the same time they will create more interference between each other and also increasing the required Eb/No that they must achieve.

Scheduling a few number of users to transmit at the same time at a high bit rate is shown to be a better option in the simulation results. The schedulers that follow this principal (the Round Robin, Rate Estimation and UCQI schedulers) show a big gain in performance compared to the Rate scheduler, around 50% gain when there are large amount of users in the cell.

The difference between the schedulers that grant a few users to transmit at the same time at high bit rates is given by the methods they use to select the user with highest priority for transmission. While in the Round Robin scheduler there is no preference for any user and all of them are scheduled in round robin fashion, the UCQI scheduler always gives priority to the user who has the best channel conditions at the moment of the request.

One of the most relevant metrics measured during the simulations was the total transmitted power. As shown in the results the UCQI scheduler gave a gain compared to the Rate Scheduling and Round Robin scheduling transmitting with less power (6dB less than the Rate Scheduler and 4dB less than the Round Robin scheduler). Since the UCQI scheduler gives priority for transmission to the UEs that experience best channel conditions, these users can transmit their data at a low power. Since the scheduler is based on the NodeB it can react very fast to the changes in channel variations of the user, in that way the variations caused by fading can be taking into account preventing transmissions during fading drops and taking advantage of fading gains. The benefits of this behavior are not just regarding power consumption in the UE but benefits the whole system since lower amount of interference is generated to other cells.

The Rate Estimation scheduler focus our attention in another aspect of the scheduler mechanisms. This scheduler is not concern in giving any priorities to the users so it also grants all the users in a round robin fashion but this time the assignation of bit rates is done considering the channel conditions of each user. Each user is assigned with a corresponding bit rate related to its channel condition so that the resources are not wasted in users that are not able to use them. The better utilization of the channel of the Rate Estimation scheduler is reflected in the simulation results achieving higher average cell throughput, using less power, achieving higher throughput per user and lower delays. Regardless of the policies use to select which users should have the priority to transmit, the rate estimation should be implemented so no resources are wasted in allocations to users that are not able to support them.

The results also show that the different scheduling mechanisms can affect the quality of the service experienced by applications. This lead to include priority assignation to the different types of flows so that the scheduler could take this information into consideration when allocating resources. Even doe the transmission of this type of priority information from the UE to the NodeB is not supported by the standards it was included in the simulation for the purpose of the study. The results of the simulations show that including priority to the different types of flows can improve the service provided to high priority flows even when they experience a bad channel condition. Low priority flows (which are assign to applications which doesn't have strict requirements) are affected by the scheduler mechanism but the quality will not be drastically affected since these types of applications can afford a decrease in their rate or an increase of their transmission delay without major impact in the quality experienced by the user. Adding signaling for priority differentiation of the flows give the scheduler more control for managing the quality and throughput balance.

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