Improving HSDPA network performance

Building a HSDPA system simulation with OPNET

by

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Acknowledgments

This has been an intense year, as I would not have thought possible. It has been an experience that I will remember dearly only because I was in the company of great people.

I dedicate this thesis to Anca, without which my journey here would not have happened. I would also like to thank my supervisor, Meriel Huggard, for guiding me along the way and pointing me in the right direction to start with.

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Supervisor: Meriel Huggard

This thesis provides a background on the High Speed Downlink Packet Access (HSDPA) technology; a feature which has been implemented since 2002 as part of the Release 5 specifications of the 3GPP WCDMA/UTRA-FDD standards. One of the network elements that have a direct influence on performance is the scheduling algorithm used in the base station (Node B). Most of the new scheduling algorithms proposed by the research community have not been tested in a real-world environment or on an industry standard network simulator, therefore unable to gain more practical utility. The aim of the thesis is to design a new HSDPA model library using the OPNET system simulator environment. The model should be capable of supporting implementation of current scheduling algorithms for thorough testing. To support an evolution toward more sophisticated network and multimedia services, the main target of HSDPA is to increase user peak data rates, quality of service, and to generally improve spectral efficiency for downlink asymmetrical and bursty packet data services. This is accomplished by introducing a fast and complex channel control mechanism based on a short and fixed packet transmission time interval (TTI), adaptive modulation and coding (AMC), and fast physical layer (L1) hybrid ARQ. The former part of the thesis provides background knowledge of WCDMA systems, more specifically their evolution to UMTS and HSDPA. It continues with a discussion on the influence of packet scheduling and QoS requirements on the design of HSDPA systems. The later part focuses on the simulation environment using OPNET Modeler, with an analysis of the UMTS built in model and the development of a new HSDPA system model.

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Chapter 1

Introduction

This chapter provides an introduction of the dissertation, its objective and the topics that will be discussed in the dissertation. At the end of this chapter, the major issues that shall be discussed in each chapter shall be outlined

1.1 Motivation and Goals

HSDPA networks provide us with mobile broadband at speeds up to a theoretical maximum of approximately 10 Mb/s. However, due to network congestion it is often extremely difficult to achieve speeds anywhere near this maximum value. One of the challenges for these networks is to meet the stringent quality of service requirements of traffic such as voice over IP (VoIP) and video streaming, whilst trying to optimise network throughput. One of the network elements that have a direct influence on performance is the scheduling algorithm used in the base station (Node B).

Current scheduling algorithms are very basic in design and there have been new, more sophisticated algorithms proposed by the academic community. Most of these new algorithms have not been tested using live test-bed scenarios or even with industrial strength simulators; therefore it is very difficult to comment on their performance in a meaningful and critical way.

This project aims to design a new HSDPA model library using the OPNET system simulator environment. This model is designed to be able to accomodate various scheduling algorithms and test their impact on the network performance. Some of the main points of research that have been followed while writing this paper are:

1. Analyze the UMTS Release 5 specifications in order to distinguish key features that need to be integrated in a valid HSDPA simulation environment

2. Analyze in detail the operation of the layers 1 and 2 of the radio interface protocol stack in UTRAN and their implementation in the OPNET UMTS model
3. Identify and describe the key components that require modifications to match the HSDPA 3GPP standard
4. Propose and implement the changes required in a new model library
5. Perform simulation experiments to analyze the behaviour of the proposed changes

1.2 Research Scope

The following research areas are investigated to meet the dissertation's goals:

1. Architecture and functionality of release 99 of UMTS and more focus on Release 5
2. Classification and requirements of HSDPA operation with focus on packet scheduling and QoS
3. Computer simulation tools to evaluate the efficiency of existing packet scheduling algorithms
4. Operation of OPNET Modeler's programming model and the UMTS model library

1.3 Dissertation outline

This document is organized as follows:

Chapter 2 briefly describes the evolution of WCDMA networks towards 3G systems, gives an overview of UMTS Release 99 and some focus on HSDPA key technologies

Chapter 3 provides the state of the art in the field of packet scheduling and quality of service in HSDPA

Chapter 4 contains an analsys by the author of the UMTS model implemented in opnet and the suggested changes required to upgrade it to HSDPA

Chapter 5 describes the implementation part of the project

Chapter 6 summarizes and concludes the dissertation

Chapter 2

Evolution of WCDMA to HSPA

This chapter presents an overview of Wideband Code Division Multiple Access (WCDMA), including the evolution steps to the creation of current 3.5G systems (i.e. HSDPA - High Speed Data Packet Access) and future technologies that are currently close to implementation status (i.e. HSPA+ R8 - High Speed Packet Access release 8).

2.1 Introduction

As part of the natural evolution of data communications and telecommunications, the development of cellular networks is having an exponential increase (Fig. 2.1) that started in 2002 with the launch of the Wideband Code Division Multiple Access (WCDMA) networks. The Third generation Partnership Project (3GPP) WCDMA networks are currently deployed in 120 countries around the world, with 379.3 million subscribers as of August 2009 [8]. Mobile communication forecasts suggest that the average revenue per user for voice traffic is set to decline in cellular networks (e.g. in Western Europe as stated in [9]), which leads operators to introduce new data services to sustain their business growth in the regions where the speech service is already deployed. This pushes the WCDMA networks to deliver an ever increasing share of voice and data traffic, driven by the rapid increase in the demand for data services, primarily IP.

The provision of IP-based services in mobile communication networks has raised expectations of a major traffic emergence analogous to the one already experienced over the Internet. In [10], the Universal Mobile Telecommunication System (UMTS) Forum predicts a considerable increase of the worldwide demand for wireless data services over the years to come of the present decade. Besides the Internet-like services, the addition of mobility to data communication enables new services not meaningful in a fixed network [11], e.g. location-based services.

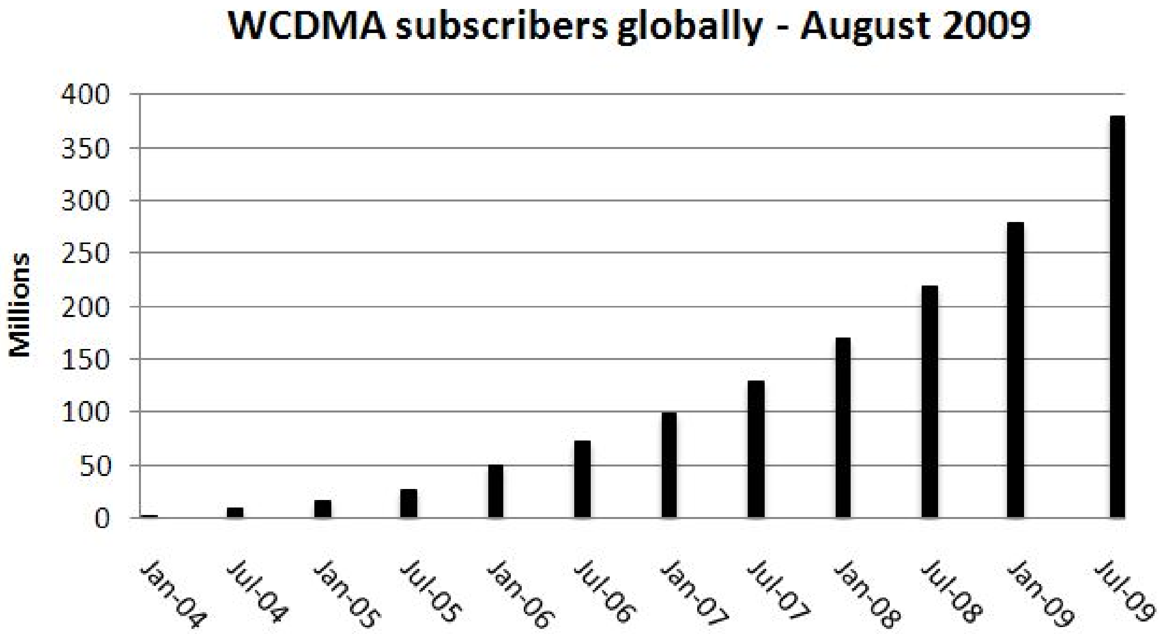


Figure 2.1: 3G WCDMA subscriber growth

The goal of Third Generation (3G) mobile communication systems is to provide users not only with the traditional circuit switched services, but also with new multimedia services with high quality images and video for person-to-person communication, and with access to services and information in private and public networks

2.2 Third Generation Systems Overview

3G systems are defined by a collection of international standards and technologies, defined in the ITU [12] specification International Mobile Telecommunications-2000 (IMT-2000), with the goal of increasing efficiency and improving the performance of mobile wireless networks. IMT-2000 is a radio and network access specification, defining several recommended methods and technology platforms that meet the overall goals of the specification. It provides a framework for worldwide wireless access by connecting the various systems of terrestrial and/or satellite based networks and intends to take advantage of the potential synergy between digital mobile telecommunications technologies and systems for fixed and mobile wireless access systems.

Some of the main ITU-2000 requirements for the third generation of mobile cellular systems are listed bellow:

- Compatibility of service within the IMT-2000 family of standards (GSM, ATM, IP)

- High quality of service
- Worldwide or at least regional common frequency bands
- Small terminals for worldwide use and global roaming capability
- Support for multimedia services, applications and terminals
- Flexibility for evolution to the next generation of wireless networks
- High speed packet data rates, at least:
  - 2 Mbps for fixed environment
  - 384 kbps for pedestrian speeds
  - 144 kbps for vehicular speeds

The advantage to the end users comes from greater data speeds, increased capacity for voice and data connections and access to more diverse services on the world wide web. Just as with common cable networks, data services are expected to have a significant rate of growth over the coming years and will likely become the dominating source of traffic load. A broader range of applications is expected to supplement speech services, such as: multi-player games, instant messaging, on-line shopping, face-to-face video conferences, video and music streaming, as well as personal/public database access.

As more complex services evolve, a major challenge of cellular systems design is to achieve a high system capacity and simultaneously facilitate a mixture of diverse services with very different quality of service (QoS) requirements. Various traffic classes exhibit very different traffic symmetry and bandwidth requirements. For example, two-way speech services (conversational class) require strict adherence to channel symmetry and very tight latency, while Internet download services (background class) are often asymmetrical and are tolerant to latency. The streaming class, on the other hand, typically exhibits tight latency requirements with most of the traffic carried in the downlink direction.

In the standardisation forums, WCDMA technology has emerged as the most widely adopted third-generation air interface. Its specification has been created in the 3rd Generation Partnership Project (3GPP), which is the joint standardisation project of the standardisation bodies from Europe, Japan, Korea, the USA and China. Within 3GPP, WCDMA is called Universal Terrestrial Radio Access (UTRA) Frequency Division Duplex (FDD) and Time Division Duplex (TDD), the term WCDMA being used to cover both FDD and TDD operations. UTRA is the radio access part of the UMTS network. A comprehensive and up to date study of the 3GPP UMTS release 5 can be found in the 2006 edition of [2].

CDMA is the dominant technology used in 3G systems also due to the following design issues [13]:

Bandwidth: An important design goal for all 3G systems is to limit channel usage to 5MHz. There are several reasons for this goal. On one hand, a larger bandwidth improves the receivers ability to resolve multipath when compared to narrower bandwidths. On the other hand, available spectrum is limited by competing needs, and 5 MHz is a reasonable upper limit on what can be allocated for 3G. Finally, 5 MHz is adequate for supporting data rates of 144 and 384 kHz, the main targets for 3G services.

Chip rate: Given the bandwidth, the chip rate depends on desired data rate, the need for error control and bandwidth limitations. A chip rate of 3 Mcps (mega-chips per second) or more is reasonable given these design parameters.

Multirate: The term multirate refers to the provision of multiple fixed-data-rate logical channels to a given user, in which different data rates are provided on different logical channels. The advantage of multirate is that the system can flexibly support multiple simultaneous applications from a given user and can efficiently use available capacity by only providing the capacity required for each service. Further the traffic on each logical channel can be switched independently through the wireless and fixed networks to different destinations. Multirate can be achieved with multiple CDMA codes, with separate coding and interleaving, and map them to separate CDMA channels.

2.2.1 UMTS Release 99 Overall System Architecture

The UMTS utilises the same well-known architecture that has been used by all main second generation systems and even by some first-generation systems. It consists of a number of logical network elements with different defined functionalities. A comprehensive study of the architecture can be found in [14] and I will only showcase some of it's main features. The network elements can be grouped as follows (Fig. 2.2):

- Radio Access Network (RAN; UMTS Terrestrial RAN (UTRAN)) - handles all radio-related functionality;
- Core Network (CN) - responsible for switching and routing calls and data connections to external networks;
- User Equipment (UE) - the interfaces between the user and the radio interface.

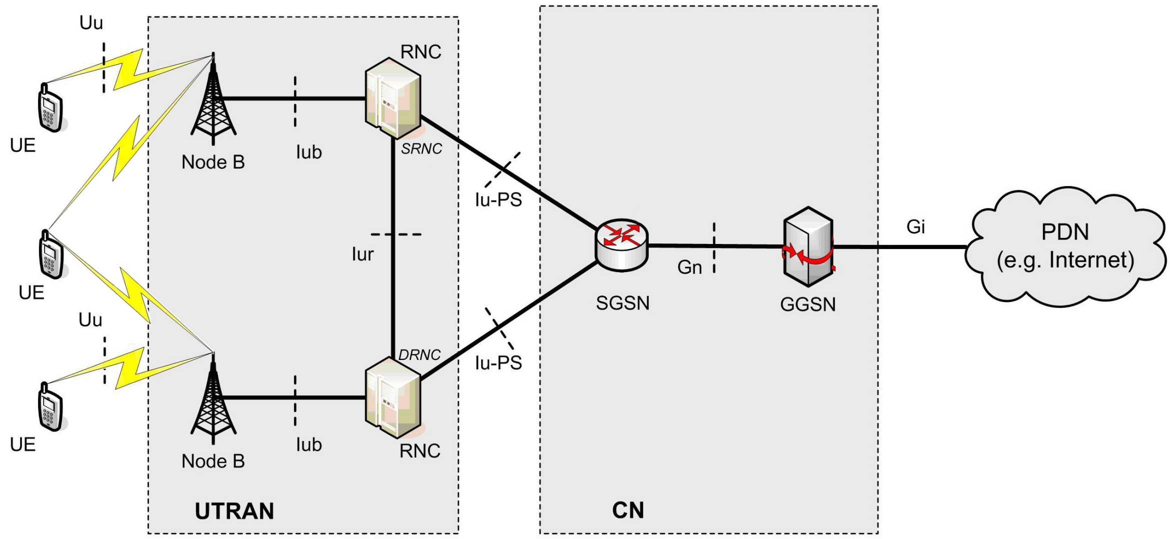


Figure 2.2: UMTS Architecture

2.2.2 UMTS Radio Access Network Architecture

The main task of UTRAN is to create and maintain Radio Access Bearers (RAB) for communication between UE and the CN. The RAB emulate a fixed communication path to the UE so that the CN elements are released from the need to take care of radio communication aspects. From the bearer architecture point of view, the main task of the UTRAN is to provide bearer service over these interfaces; in this respect the UTRAN controls Uu interface and in Iu interface the bearer service provision is done in co-operation with the CN [15].

UTRAN consists of Radio Network Subsystems (RNS) and each RNS contains various amount of Base Stations (BS, or officially Node B) deploying the Uu interface, and one Radio Network Controller (RNC). The RNSs are separated from each other by UMTS Interface between RNCs (Iur) interface forming connections between two RNCs. The Iur, which has been specified as an open interface, carries both signalling and traffic information.

Node B Its main tasks are to establish the physical implementation of the Uu interface (channel coding and interleaving, rate adaptation and spreading, etc.) and, toward the network, the implementation of the Iub interface by utilising the protocol stacks specified for these interfaces. It also the inner loop power control function to minimise the effects of the near-far phenomenon, as well as measuring the connection metrics and signal strength, thus supporting soft handover between different antennas connected at the same node B.

RNC The RNC is the switching and controlling element of the UTRAN. The RNC is located between the Iub and Iu interface. It also has the third interface called Iur for inter-RNS connections. Referring to the bearers, the RNC is a switching point between the Iu bearer and RB(s). One radio connection between the UE and the RNC, carrying user data is a RB. The RB in turn is related to the UE Context. UE Context is a set of definitions required in Iub in order to arrange both common and dedicated connections between the UE and the RNC.

The RNC holding the Iu bearer for certain UE is called *Serving RNC* (SRNC). Another logical role the RNC may have is Drifting RNC (DRNC). When in this mode, the RNC allocates UE Context through itself, the request to perform this activity comes from the SRNC through the Iur interface. Both SRNC and DRNC roles are functionalities, which may change their physical location. When the UE moves in the network performing soft handovers, the radio connection of the UE will be accessed entirely through a different RNC than the SRNC, which originally performed the first RB set-up for this UE. In this case the SRNC functionality will be transferred to the RNC, which practically handles the radio connection of the UE. This procedure is called SRNC or SRNS Relocation.

Also part of the RNC, the Radio Resource Management Algorithms have the following functions:

- Handover Control - guarantees the user mobility in a mobile communications network, where the subscriber can move around;
- Power Control - essential feature of any CDMA based cellular system, due to the near-far problem, interference dependent capacity of the WCDMA and the limited power source of the UE;
- Admission Control (AC) - estimates whether a new call can have access to the system without sacrificing the bearer requirements of existing calls, based on Signal to Interference Ratio measured per UE;
- Packet Scheduling - the AC is responsible for the QoS management of the accepted RABs and their influence to the overall performance of the UTRAN.
- Code Management - Both channelisation and scrambling codes used in the Uu interface connections are managed by the RNC

2.2.3 UMTS Core Network Architecture

The core network remains mostly unchanged from the second generation GSM network with the addition of a new interface called the Iu interface (denoted IuPS to indicate packet-switched mode). This interface is used to connect the core network to the new UTRAN element. The CN is responsible for switching and routing calls and data connections to external networks. It is the platform for all communication services provided to the UMTS subscribers and contains functions for inter-system handover, gateways to other networks, mapping the end-to-end Quality of Service (QoS) requirements to the UMTS bearer service and performs location management.

The SGSN (Supporting GPRS Support Node) and GGSN (Gateway Support Node) provide functionality for packet-switched services. The GGSN is the interworking unit between the UMTS system and the packet data networks. This node contains routing information for mobile users, performs address conversion and tunnels data to a user via encapsulation. The SGSN requests user addresses, keeps track of UE locations, is responsible for collecting billing information and performing several security functions, such as access control.

Packet data is transmitted from a packet data network via the GGSN and SGSN directly to the RNCs. From there it is transmitted to the Node Bs and finally to the UE.

2.2.4 User Equipment Architecture

The UE has mostly been regarded as providing the application interface and services to the end user and consists of two parts:

- The Mobile Equipment (ME) is the radio terminal used for radio communication over the Uu interface;

- The UMTS Subscriber Identity Module (USIM) is a smartcard that holds the subscriber identity, performs authentication algorithms, and stores authentication and encryption keys and some subscription information that is needed at the terminal;

From a network standpoint, the overall protocol architecture and functions of the terminal are subject to standardisation and their implementation and additional internal capabilities are strictly implementation dependent.

2.3 HSDPA Overview

In order to improve support for high data-rate packet-switched services, 3GPP deployed an evolution of UMTS based on WCDMA known as High Speed Downlink Packet Access (HSDPA) which is included in the Release 5 specifications. HSDPA is targeting increased capacity, reduced round trip delay, and higher peak data rates up to 10 Mbps. To achieve these goals, a new shared downlink channel, called the High Speed Downlink

Shared Channel (HS-DSCH) is being introduced to replace the DSCH transport channel used in previous releases. In addition, three fundamental technologies are utilised, which are tightly coupled and rely on rapid adaptation of the transmission parameters to the instantaneous radio conditions:

Fast link adaptation techniques based on multiple Modulation and Coding Schemes (MCS) enable the use of spectrally efficient higher order Quaternary Amplitude Modulation with 16 states (16 QAM) when channel conditions permit. Alternatively, these revert to conventional and more robust Quaternary Phase Shift Keying (QPSK) modulation for less favourable channel conditions.

Fast Hybrid Automatic Repeat Request (HARQ) algorithms rapidly request the retransmission of missing data entities and combine the soft information from the original transmission and any subsequent retransmissions before another attempt is made to decode a data packet.

Fast scheduling shares the HS-DSCH among the users. This technique, which exploits multi-user diversity, strives to transmit to users with favourable radio conditions. Moreover, the time interval considered for scheduling is no longer based on radio frames of 10 ms but shortened to a 2 ms interval in FDD-mode and 5 ms interval in TDD-mode.

2.3.1 Contrasting features of UMTS Release 99 and Release 5

HSDPA is essentially modifying the existing protocol architecture, which affects different protocol layers as illustrated in Fig. 2.3 according to the Medium Access Control (MAC) layer specification in [16].

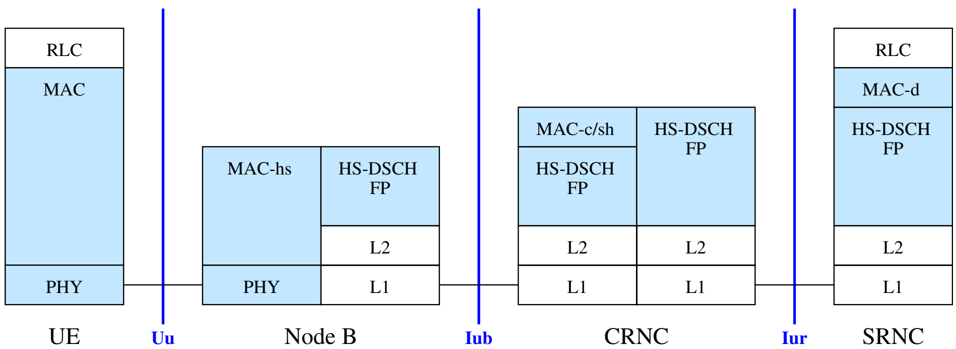


Figure 2.3: HSDPA protocol architecture, modified parts highlighted, [1]

The fundamental characteristics of the HS-DSCH and the DSCH are compared in Table 2.1 [17]. While being more complicated, the replacement of fast power control with fast AMC yields a power efficiency gain due to an elimination of the inherent power control overhead. Specifically, the spreading factor of the assigned channelisation code is fixed to 16 and up to 15 out of 16 orthogonal codes can be allocated for the HS-DSCH. The HS-DSCH uses AMC for fast link adaptation. A terminal experiencing good link conditions will be served with a higher data rate than a terminal in a less favourable situation. To support the different data rates, a wide range of channel coding rates and different modulation formats, namely QPSK and 16QAM, are supported.

UMTS Release 99	UMTS Release 5(HSDPA)
• TTI=10, 20, 40, 80 ms • Variable SF=1 256 • More transport block per TTI • Convolutional code or turbo codes • QPSK only • Configurable CRC • Scheduling in RNC • Retransmission in AM RLC • Power control • Soft hand off	• TTI=2ms • Fixed SF=16 • One transport block per TTI • Turbo codes only • QPSK and 16QAM according to UE capability • CRC of 24 bits • Scheduling in Node B • Physical layer retransmissions • Adaptive modulation and coding • Hard hand off

Table 2.1: Comparisons between Rel-99 and HSDPA

In order to increase the link adaptation rate and efficiency of the AMC, the packet duration has been reduced from normally 10 or 20 ms down to a fixed 2 ms. To achieve low delays in the link control, the MAC functionality for the HS-DSCH has been moved from RNC to the Node-B. It will enhance the packet data characteristics by reducing the round-trip delay.

Currently, the retransmission functionality in WCDMA R99 DSCH is implemented as conventional ARQ scheme. In the conventional ARQ scheme when a received packet is detected as erroneous, it is discarded and a negative acknowledgement is sent to the transmitter requesting for retransmission. The retransmitted packets are identical with the first transmission. The HSDPA supports both the incremental redundancy (IR) and chase combining (CC) retransmission strategies [17]. By combining soft information from multiple transmission attempts, the number of retransmission needed, and thus the delay, will be reduced. HARQ with soft combining also adds robustness against link adaptation errors and is closely related to the link adaptation mechanism.

In HSDPA, the use of a single CRC for all transport blocks in the transmission time interval (TTI) reduces the overhead compared to using a CRC per transport block that employing in R99 DSCH. Furthermore, the advantages by performing retransmission individually on transport blocks are limited since, in most cases, either most of the transport blocks transmitted within a TTI are erroneous or all of them are correctly decoded.

2.3.2 HSDPA channel structure

There are different Physical channels in WCDMA, which are mapped to Transport and Logical Channels as indicated in Fig. 2.4. Some exist only in Uplink (UL) direction, some in Downlink (DL) direction and some in both directions. Over the air interface, each HSDPA user receives the physical channels HS-PDSCH and HS-SCCH, and one associated dedicated physical channel (DPCH), as illustrated in Fig. 2.5. The HS-SCCH is used for layer-1 control signalling, while the associated DPCH is used for layer-3 control signalling such as radio resource control (RRC). Transmission on the HS-SCCH starts two slots before the actual data transmission on the HS-PDSCH while on the uplink, the HSDPA user is transmitting HSDPA-specific layer-1 signalling on the high-speed dedicated physical control channel (HS-DPCCH).

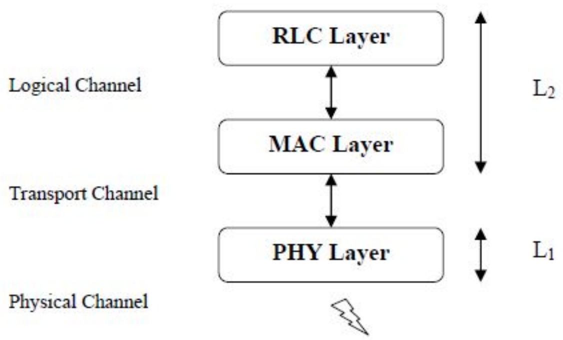


Figure 2.4: Channel Levels, [2]

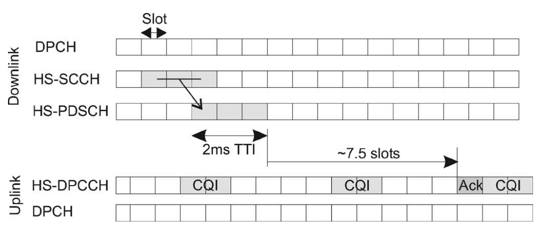


Figure 2.5: Downlink&Uplink channel structure

The list of all Transport channels and their corresponding Physical channels is given in Table 2.2. Every channel has its own functionality and role in the operation of system [18].

Transport Channel	Physical Channel
(UL/DL) Dedicated channel DCH	Dedicated physical data channel DPDCH Dedicated physical control channel DPCCH
(UL) Random access channel RACH	Physical random access channel PRACH
(DL) Broadcast channel BCH	Primary common control physical channel P-CCPCH
(DL) Forward access channel FACH (DL) Paging channel PCH	Secondary common control physical channel S-CCPCH
(DL) Downlink shared channel DSCH	Physical downlink shared channel PDSCH
(UL) Common packet channel CPCH	Physical common packet channel PCPCH Synchronisation Channel (SCH) Common Pilot Channel (CPICH) Acquisition Indication Channel (AICH) Paging Indication Channel (PICH) CPCH Status Indication Channel (CSICH) Collision Detection/Channel Assignment Indicator Channel (CD/CA-ICH)

Table 2.2: Transport and Physical Channels in WCDMA

Various methods for packet data transmission in WCDMA downlink already exist in Release 99 (R99). The standard provides two different channels for high data rate packet services in the downlink dedicated channel (DCH) and downlink shared channel (DSCH). Both can provide variable bit rate. The DCH is the basic radio bearer in WCDMA and supports all traffic classes due to high parameter flexibility. The data rate is updated by means of variable spreading factor while the block error rate (BLER) is controlled by inner and outer loop power control mechanisms. However, in the dedicated channel, spreading factor and spreading code are reserved from the OVSF (orthogonal variable spreading factor) code tree based on the highest data rate. This easily leads to shortage of downlink codes. Therefore, the DSCH is more appropriate for high data rate packet services. The benefit of the DSCH over the DCH is a fast channel reconfiguration time and packet scheduling procedure [19].

The HSDPA concept can be seen as the latest evolution of the DSCH and a new transport channel targeting packet data transmissions, the high speed DSCH (HS-DSCH), is introduced. The HS-DSCH supports three basic principles: fast link adaptation, fast Hybrid ARQ (HARQ), and fast scheduling. These three principles rely on rapid adaptation to changing radio conditions and the corresponding functionality is therefore placed in the Node B instead of the RNC. Similar to the R99 DSCH, each UE to which data can be transmitted on the HS-DSCH has an associated DCH. This DCH is used to carry power control commands and the necessary control information in the uplink namely ARQ acknowledgement (ACK/NACK) and channel quality indicator (CQI). To implement the

HSDPA feature, a new channel called High Speed Shared Control Channel (HS-SCCH) is introduced in the physical layer specifications. The HS-SCCH carries the control information that is only relevant for the UE for which there is data on the HS-DSCH [19].

2.3.3 Adaptive modulation and Coding

As discussed in [2], the principle of AMC is to change the modulation and coding format (transport format) in accordance with instantaneous variations in the channel conditions, subject to system restrictions. AMC extends the systems ability to adapt to good channel conditions. Channel conditions should be estimated by feedback from the receiver. For a system with AMC, users closer to the cell site are typically assigned higher order modulation with higher code rate (e.g. 16 QAM with r = 3/4 turbo codes). In [20] the coding rate to be used on HS-DSCH is defined as 1/3 turbo code. In the following rate matching uses, puncturing or repetition allow coding rates other than 1/3 become available. The effective coding typically is varied between 1/4 and 3/4. More redundancy up to 1/6 (only in combination with QPSK modulation) is possible under worst case conditions to prevent call drops ([21]), respectively less (nearly no) redundancy is possible (in combination with 16QAM) for high end equipment under extremely good radio conditions (Fig. 2.6). The decision about selecting the appropriate MCS is performed at the receiver side according to the observed channel condition with the information fed back to the transmitter in each frame [22].

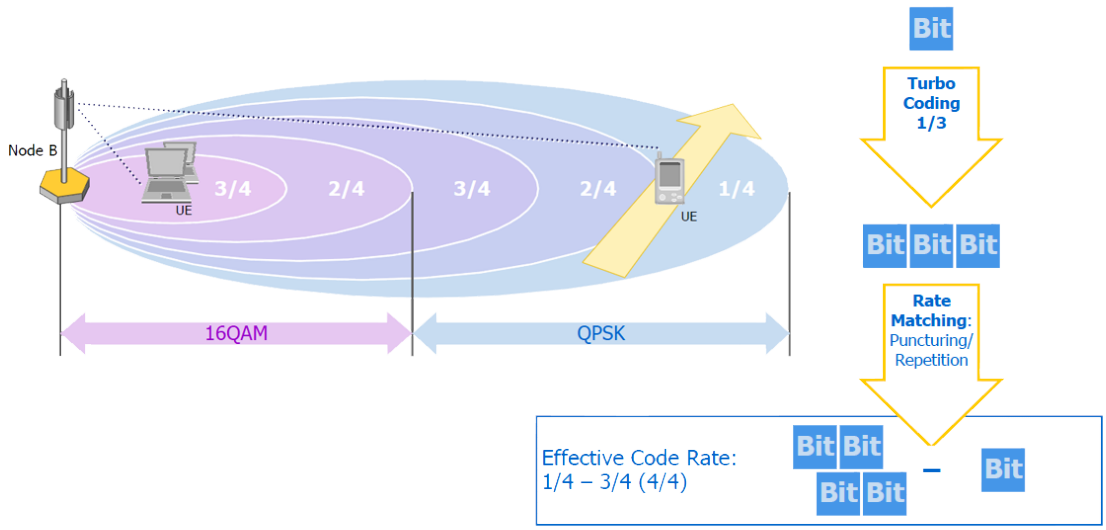


Figure 2.6: Adaptive Modulation and Adaptive Coding principles

A key factor determining the performance of an AMC scheme is the method used at the receiver to estimate the channel condition and thereby deciding for the appropriate

MCS to be used in the next frame [22]. Different control channels are used to communicate channel quality information between user equipment (UE) and Node-B and an appropriate AMCS is selected for the specific UE based on this information. As mentioned in [2], the sequence of Link Quality Feedback could be:

- UE measures channel quality by evaluation of CPICH (Common Pilot Channel)
- UE reports channel quality to BS by choosing a transport format (modulation, code rate, and transmit power offset) such that a 10
- BS determines the transport format based on the recommended transport format and possibly on power control commands of associated dedicated physical channel (DPCH).
- Transport format update rate: every TTI = 3 slots
- Measurement report rate: every TTI = 3 slots

Different to UMTS Rel. 99, in HSDPA the Spreading Factor SF is fixed to SF = 16 (Fig. 2.7). SF = 16 allows data rates of 240 ksymbol/s while higher data rates are possible via Multi-Code operation. Depending on the User Equipment capabilities up to 5, 10 or 15 Codes can be bundled to one connection, resulting in 1.2 Msymbol/s, 2.4 Msymbol/s or 3.6 Msymbol/s. Table 2.3 shows the maximum throughput for a combination of 5, 10 or 15 codes, under a variation of coding and modulation and using the following formula:

$$Throughput = \frac{\frac{T_{slot} \cdot NF}{TTI} \cdot ECR \cdot NAS}{SF} \tag{2.1}$$

T_{slot} = Time Slot Size = 2560 chips ≈ 0.67ms;
NF = Number of Frames = 3 (15 for UMTS);
TTI = (Transmission Time Interval) = 2ms;
BMS = Number of Bits per Modulation Symbol = 4;
ECR = Effective Code Rate = {1/4; 2/4; 3/4};
N = Number of Allocated Slots = {1; 5; 10; 15};
SF = Spreading Factor = 16.

3GPP also defined 12 different categories (Table 2.4) of HSDPA terminals according to their Physical Layer capabilities([23]). They are separated according to the following parameters:

- Support of 16QAM
- Minimum total RLC & MAC-hs buffer size (50/100/150 kBytes)

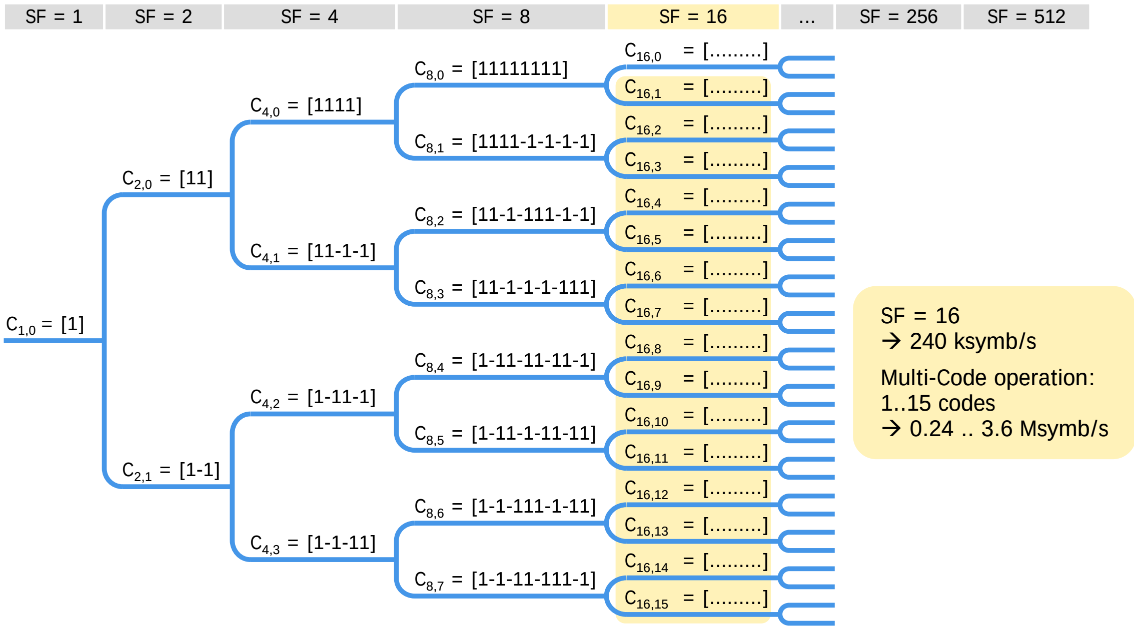


Figure 2.7: Spreading Factor - code tree

- Maximum number of SF = 16 HSDPA codes (5/10/15)
- Minimal Inter Transmission Time Interval TTI (1/2/3 Sub-Frames =2/4/6 ms)
- Maximum number of bits of an HS-DSCH transport block received within an HSD-SCH TTI
- The maximum number of Soft Channel Bits over all the HARQ processes

The system model illustrated in Fig. 2.8 ([3]) illustrates the AMC impact on the rate matcher, the modulator, as well as the channel degmentation block in the transmission chain.

MCS selection

The use of MCS leads to the increase of system throughput and spectral efficiency, as well as avoiding to predict turbo codes. Although there are no standard methods for MCS selection used in HSDPA, three methods have been tested and confirmed their efficiency: Threshold Decision Model, Markov's Model and Actual Value Interface.

The Markov Model, which takes statistical decision making approach for selecting the appropriate Modulation and Coding Scheme (MCS) is better, when considering the issue caused by the sensitivity of turbo code to the errors in predicting the channel SNR. Numerical results presented in [22] showing that Markov method substantially outperforms

Modulation	Coding Rate	Throughput with 5 codes	Throughput with 10 codes	Throughput with 15 codes
QPSK	1/4	600 kbps	1.2 Mbps	1.8 Mbps
	2/4	1.2 Mbps	2.4 Mbps	3.6 Mbps
	3/4	1.8 Mbps	3.6 Mbps	5.4 Mbps
16QAM	2/4	2.4 Mbps	4.8 Mbps	7.2 Mbps
	3/4	3.6 Mbps	7.2 Mbps	10.7 Mbps
	4/4*	4.8 Mbps	9.6 Mbps	14.4 Mbps

* Effective code rate: 1/4 - 3/4

Table 2.3: Multi-Code Operation & Data Throughput

HS-DSCH category	Buffer Size*1 [kbyte]	max. No. of HS-DSCH Codes	min.*2 Inter-TTI interval	max. No. of Bits/TTI*3	total No. of Soft Channel Bits*4	Reference combination Class
1	50	5	3(6ms)	7298	19200	1.2Mbps
2	50	5	3	7298	28800	1.2Mbps
3	50	5	2(4ms)	7298	28800	1.8Mbps
3	50	5	2(4ms)	7298	28800	1.8Mbps
4	50	5	2	7298	38400	1.8Mbps
5	50	5	1(2ms)	7298	57600	3.6Mbps
6	50	5	1	7298	67200	3.6Mbps
7	100	10	1	14411	115200	7.2Mbps
8	100	10	1	14411	134400	7.2Mbps
9	150	15	1	20251	172800	10.8Mbps
10	150	15	1	20252	172800	14.4Mbps
11*5	50	5	2	3620	14400	1Mbps
12*5	50	5	1	3620	28800	1.8Mbps

*1 on RLC & MAC-hs layer *3 Maximum No. of bits/HS-DSCH transport block
*2 TTI: Transmission Time Interval *4 for H-ARQ process *5 QPSK only

Table 2.4: Multi-Code Operation & Data Throughput

the conventional techniques that use a "threshold-based" decision making approach. Simplified Model proposed in [22] has fewer parameters, suitable to systems where changes in the fading characteristics need to be accounted for in an adaptive manner. It is shown in [22] that the Simplified Model only results in negligible loss in the expected throughput.

The third method proposed in [24] aims to incorporate the accuracy of link level simulations into the system level simulation. 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 chiplevel time resolution is needed (3.84Mcps for WCDMA).

As all data packets entering the HS-DSCH go through a rate 1/3 rate turbo encoder after having the 24 bit CRC appended, the rate matcher has to either puncture or repeat

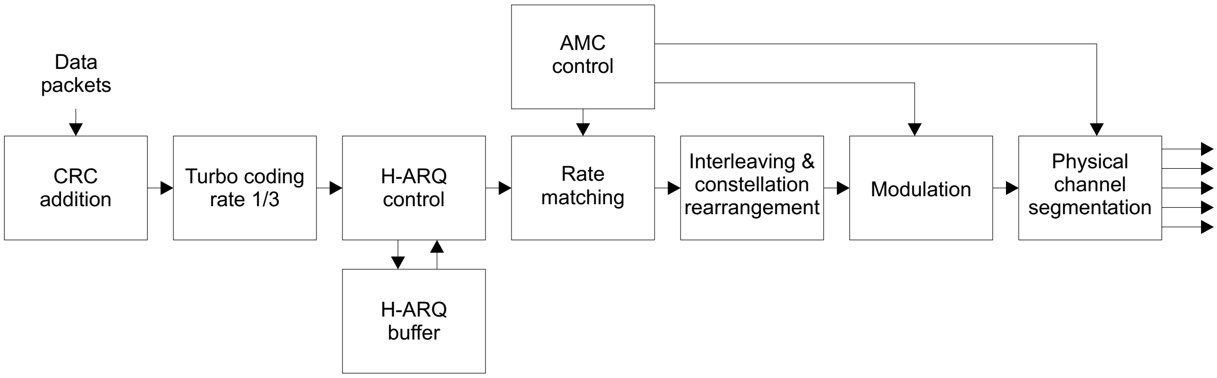


Figure 2.8: The HS-DSCH transmission scheme. Constellation re-arrangement is only used when 16-QAM is active, [3]

bits in the stream to match the available transmission capacity of the allocated physical channels. During this process, we obtain the effective code rate (ECR) as

$$ECR = \frac{Number\quad of\quad bits\quad going\quad into\quad the\quad turbo\quad encoder}{Number\quad of\quad bits\quad after\quad rate\quad matching} \tag{2.2}$$

When the link level performance has been simulated, it should be transferred into values that can be used at system or network level. One of the key tools for this is to use tables that map simple averaged powers, signal to noise ratios, or other simplified parameters into equivalent block or packet performance quantities. One approach for this is the actual value interface (AVI), which maps a given received per-packet average symbol energy to noise ratio, E_s/N_0, into a corresponding block error rate (BLER) [24].

In the AVI approach, a set of dynamic chip or symbol level simulations are performed to obtain an indication of the packet error (packet error indicator (PEI)) for a packet at a given average per-packet E_s/N_0. To illustrate the process of estimating the values for the AVI, an example of a simulation run is shown in Fig. 2.9. In this figure, the received E_s/N_0 values have been sorted in ascending order together with the corresponding PEI. In Fig. 2.9, the E_s/N_0 transition region, from where all packets are erroneously received to where all packets are successfully demodulated, has a width on the order of 2-3 dB for a given modulation and coding set (MCS) selection. An example of an AVI curve obtained using this strategy is shown in Fig. 2.10. When applying the AVI approach, the actual E_s/N_0 is used to look up an error probability, *p(error)*, in the AVI table for the selected MCS. A uniformly distributed random number, $y \in [0; 1]$, is generated to determine if the packet was successfully received ($y \geq p(error)$) [3].

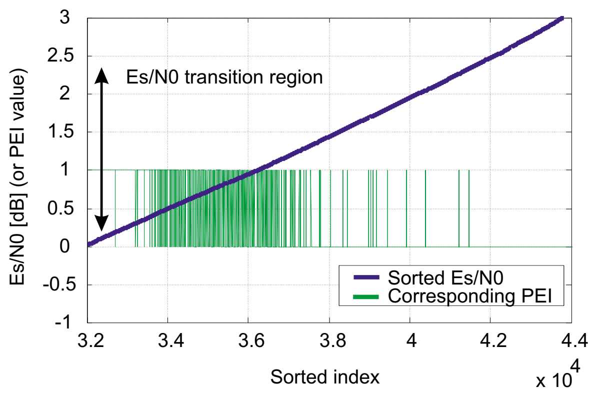


Figure 2.9: Principle behind the AVI format, [3]

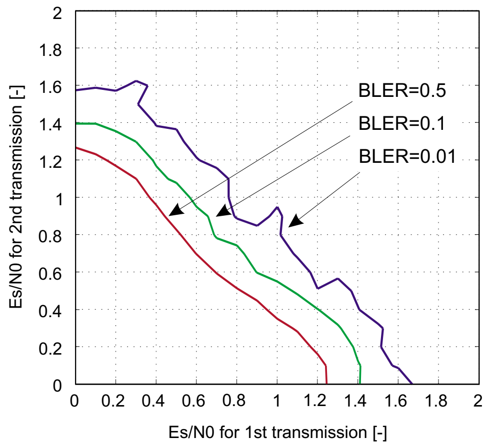


Figure 2.10: BLER contour plot, [3]

2.3.4 Hybrid Automatic Repeat reQuest mechanism

HARQ, combining soft information from additional retransmissions with the original soft information prior to decoding, greatly improves performance and adds robustness against link adaptation errors. It also compensates for errors made by the link adaptation mechanism and helps to correct the effective code rate. If all data is correctly decoded, an acknowledgement is sent to Node B, using the associated uplink control channel. But if the data is decoded incorrectly, retransmissions are requested immediately. Once the data has been retransmitted, the UE combines the previous version(s) of data with the retransmitted version. This procedure is called soft combining and usin it can increase the probability of successful decoding the data. Retransmissions are requested until the data has been decoded correctly or until the maximum predetermined number of attempts has been made. The latter again depends to some extent on the UE capabilities since the mobile needs sufficient memory to cache the soft data from intermediate retransmissions.

With the Selective Repeat (SR) ARQ protocol only erroneous blocks are repeated to reduce additional overhead. This demands reliable sequence numbers for every block, based on robust transmission of this information on the associated control channel. Most of the current implementations, with the goal of limiting the required amount of memory, are using Stop-And-Wait (SAW) ARQ, as described in [25, 26]. This simple ARQ implementation stops the transmitter from sending following blocks until the current one has been successfully received.

The HARQ mechanism resides in Node B, thus retransmissions can be requested rapidly. However, the increased decoding capabilities further burden the load in the cell by means of supplementary transmitted bursts, also introducing additional delay. Hence, in combination with an optimised scheduling algorithm, close interaction between HARQ and scheduling becomes mandatory for efficient operation ([27]).

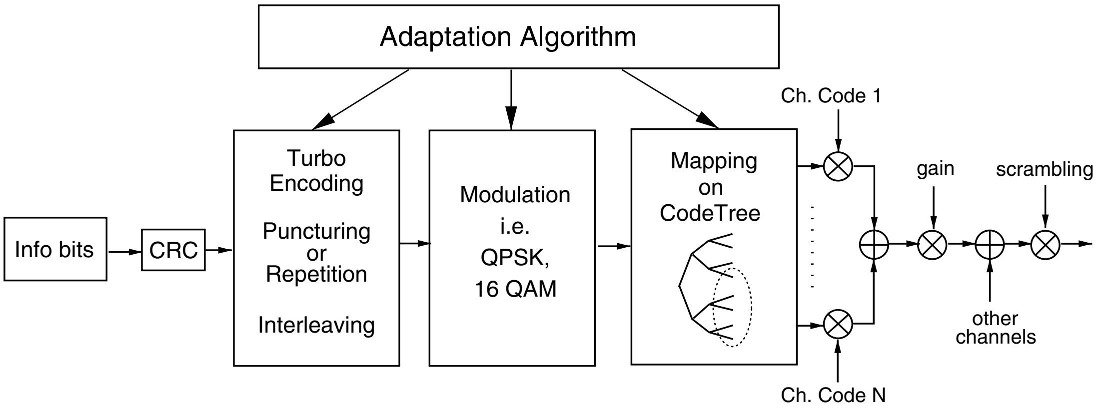


Figure 2.11: Physical layer processing
chain of AMC in HSDPA capable Node B, [1]

HARQ consists of the following three procedures:

Chase Combining (CC). If the received block doesn't have the correct Circular Redundancy Check (CRC) sequence, it is retransmitted and new values of soft bits are added to those of the first transmission to form a good set of data.

Incremental Redundancy (IR). Incorrect block is retransmitted with different redundancy version parameters

16QAM Constellation Rearrangement (CoRe). Different mapping of blocks of bits to symbols. This scheme makes full use of constellation rearrangement, but none of incremental redundancy: we send the same bits at each transmission using all possible bits-to-symbols mappings[28].

Chase combining was originally proposed in [29]. It provides a considerable gain in transmission power (3 dB in Gaussian environment) at the cost of slightly increased processing complexity and a buffer in the UE that is required to store the received values. Chase combining can be used at both bit and symbol levels. However, the improvement of performance is not sufficient to obtain target rates [27].

Incremental redundancy allows senders to send additional information in case retransmission is needed. In other words, bits which are punctured at the rate matching step of the first transmission can be sent at the second one. Using this technique one can prioritize sending systematic or parity bits, and at the same time, vary rate matching parameters, thus choosing not to puncture the same bits as at previous transmissions.

This technique greatly improves Turbo decoder's performance. The disadvantage is that the buffer size in the UE has to increase considerably, as well as processing complexity.

16QAM constellation rearrangement is a technique proposed in [29] that helps to increase performance as compared to chase combining while keeping processing complexity and buffer requirements comparatively low. Thus constellation rearrangement can be viewed as a low complexity alternative to incremental redundancy. As implied by its name, this technique is only applicable when 16QAM modulation is used, and consists in changing bits-to-symbols mapping

2.3.5 Turbo Codes

As mentioned in previous sections, HSDPA uses only Turbo Codes for forward-error-correcting (FEC). They work by inproving the energy efficiency of wireless communication systems. On the transmitter side, an FEC encoder adds redundancy to the data in the form of parity information. Then at the receiver, a FEC decoder is able to exploit the redundancy in such a way that a reasonable number of channel errors can be corrected. Because more channel errors can be tolerated with than without an FEC code, coded systems can afford to operate with a lower transmit power, transmit over longer distances, tolerate more interference, use smaller antennas, and transmit at a higher data rate.

For every combination of code rate (r), code word length (n), modulation format, channel type, and received noise power, there is a theoretical lower limit on the amount of energy that must be expended to convey one bit of information. This limit is called the channel capacity or Shannon capacity [30]. Engineers and mathematicians have come up with new codes that achieve performance close to Shannon capacity, but as recently as the early 1990s the gap between theory and practice for binary modulation was still about 3 dB in the most benign channels, those dominated by additive white Gaussian noise (AWGN). In other words, the practical codes found in cell phones, satellite systems, and other applications required about twice as much energy (i.e., 3 dB more) as the theoretical minimum amount predicted by information theory. For fading channels, which are harsher than AWGN, this gap was even larger.

Turbo codes were proposed in [30] with initial results that showed them to achieve energy efficiencies within only a half decibel of the Shannon capacity. Further benefits have pushed the use of turbo codes in various other telecommunication areas, such as: NASA's deep space communications (CCSDS), digital video broadcasting (DVB-T) and both 3G cellular standards (UMTS and CDMA2000).

Some issues of this approach have been investigated and explained in [31]. The main problem with turbo codes is complexity or the number of iterations required to execute the algorithm compared to the conventional convolutional code system, which is 32 compared to 1 in the latter. For this reason the constraint length of a turbo code's constituent encoder is shorter (L = 4 in UMTS) than that of a conventional convolutional code (L = 9 in UMTS and CDMA2000).

One method proposed in [31] to reduce complexity is to halt the decoder iterations once the entire frame has been completely corrected. This will prevent over-iteration, which corresponds to wasted hardware clock cycles. However, if the decoder is adaptively halted, then the amount of time required to decode each code word will be highly variable.

Another issue is that of channel estimation and synchronization. In order to transform the received signal into LLR form, some knowledge of the channel statistics is required. For an AWGN channel, the SNR must be known. For a fading channel with random amplitude fluctuations, the per-bit gain of the channel must also be known. In addition, it is necessary to synchronize the frame, that is, the decoder needs to know which received bit in a stream of received data corresponds to the first bit of the turbo code word. While such carrier, symbol, and frame synchronization problems are not unique to turbo-coded systems, they are complicated by the fact that turbo codes typically operate at very low SNR. As the performance of synchronization algorithms degrades with reduced SNR, it is particularly challenging to perform these tasks at the low SNRs common for turbo codes. One solution to these synchronization problems is to incorporate the synchronization process into the iterative feedback loop of the turbo decoder itself and has been propesed and investigated in [32].

2.3.6 Packet Scheduling

Packet Scheduling is a mechanism that determines the user that will receive a transmission in a given time interval. To a large extent in determines the overall behavious of the systems as it is a key design element of packet-data systems. Maximum system throuput can be obtained by assigning all available radio resources to the user with the currently best radio-channel conditions, with fairness beeing added as a key factor in all real-world implementations. There are different scheduling alghorithms that can be used to match an operators policies.

In HSDPA, scheduling of the transmission of data packets over the air interface is performed in the base station based on information about the channel quality, terminal capability, QoS class and power/code availability. Scheduling is fast because it is performed as close to the air interface as possible (in the Node B) and because a short frame length is used (2ms frame).

A more thorough discussion on this topic will be included in the third chapter of this thesis.

2.4 3GPP HSPA+ R8

Current real-world implementations of the 3GPP UMTS releases include all versions up to Release 6 (HSUPA). According to studies done by [8] with market data correct on the 19th August 2009, 14 HSPA+ systems support 21Mbps peak Downlink speeds and 1 HSPA+ system supports 28 Mbps peak Downlink speed.

Currently, telecoms operators investigate the feasibility of implementing Release 8 in order to take advantage of these technology changes:

- 42 Mbps Peak Data Rates;
- Multicarrier Enhances Broadband Experience - R8 Doubles data rates and lowers latency for all users due to an aggregation of multiple 5 MHz carriers to create a bigger data pipe (Fig. 2.12
- Multicarrier Increases Bursty Appl. Capacity;
- Standardized Support for Femtocells
- Low cost implementation - Incremental and cost-effective upgrade that leverages existing assets as NodeB and RNCs require only software upgrades and the technology is backward compatible with WCDMA and HSPA devices

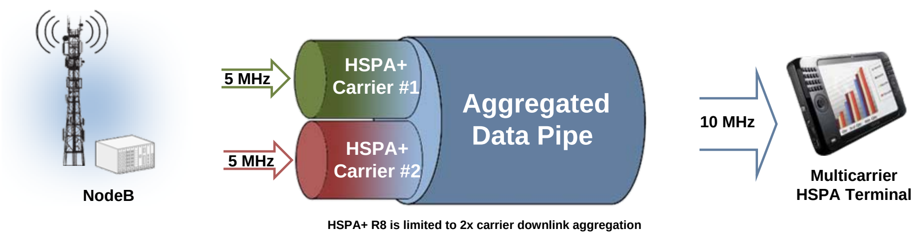


Figure 2.12: HSPA+ R8 Multicarrier operation, [4]

2.5 Summary

This chapter has discussed the most fundamental features of 3rd generation telecommunications systems leading up to a description of UMTS Release 99 and Release 5 (HSDPA) technologies.

An overview if the UMTS Release 99 system architecture described the main components, their functionality and the technologies that are in use. The discussion points out the communication features of the UTRAN, with the RNC and Node B systems creating and maintaing Radio Access bearers for communicaiton between UE and the CN. This is also the point in the system where the handover, power and admission control, as well as packet scheduling and code management take place. The UE and CN elements of the system are briefly described to better understand their role in the communication process.

The chapter continues with an analys of the changes implemented in UMTS systems to match the Release 5 specifications that define HSDPA. Layer 1 and 2 changes are discussed in terms of physical and transport channel mappings, frame size, modulation types and much more. The key technologies brought forward by HSDPA include the introduction of AMC, Hybrid ARQ, Fast packet Scheduling and the mandatory use of turbo codes.

The core idea of AMC is to dynamically change the Modulation and Coding Scheme (MCS) in subsequent frames with the objective of adapting the overall spectral efficiency to the channel condition. From the discussion presented in previous sections we can conclude that correct selection of MCS is quite important in order to increase the system throughput, increase spectral efficiency and avoid prediction of turbo codes. There are no standard methods for MCS selection used in HSDPA but we have presented three models widely used in system simulations: Threshold Decision Model, Markov's Model and Actual Value Interface. Most of the network symulators implement the AVI approach as it provides accurate results with less processing than other models, The AVI algorithm maps a given received per-packet average symbol energy to noise ratio, E_s/N_0, into a corresponding block error rate (BLER). When applying the AVI approach, the actual E_s/N_0 is used to look up an error probability, *p(error)*, in the AVI table for the selected MCS.

Fast Hybrid Automatic Repeat Request (HARQ) algorithms rapidly request the retransmission of missing data entities and combine the soft information from the original transmission and any subsequent retransmissions before another attempt is made to decode a data packet. The HARQ mechanism resides in Node B, thus retransmissions can be requested rapidly and it consists of the following three procedures: Chase Combining, Incremental Redundancy and 16QAM Constellation Rearrangement.

The chapter ends with an analisys of the forward-error-correcting facility in HSDPA, implement exclusively using turbo codes. Turbo codes are an improvement of convolutional codes where Recursive Systematic Code (RSC) is used. The recursive nature enables turbo codes performance to be improved after each iteration. Together with RSC, interleaving technique considerably contributes to the high performance and robust of Turbo codes. One drawback of Turbo codes is the complexity; however, this disadvantage is reduced thanks to the introduction of log-MAP in Viterbi algorithm.

As one of the key new features in the HSDPA model, the packet scheduling and QoS principles, algorithms and performance will be discussed in the following chapter. This will provide a clearer understanding of packet scheduling, how it might be included in a system simulator and it's gain in performance over the UMTS model.

Chapter 3

Packet Scheduling and Quality of Service in HSDPA

3.1 Introduction

Packet Scheduling is the mechanism determining which user to transmit to in a given time interval. It is a key element in the design of packet-data system as it to a large extent determines the overall behaviour of the system. Maximum system throughput is obtained by assigning all available radio resources to the user with the currently best radio-channel conditions, while a practical scheduler should include some degree of fairness. By selecting different scheduling algorithms, the operators can tailor the behaviour of the system to suit their needs.

In HSDPA, scheduling is performed as close to the air interface as possible, in the base station, based in information about the channel quality, terminal capability, QoS class and power/code availability. The goal of the Packet Scheduling is to maximise the network throughput while satisfying the QoS requirement from the users. With the purpose of enhancing the cell throughput, the HSDPA scheduling algorithms take advantage of the instantaneous channel variations and temporarily increase priorities of the favourable users. Since the users channel quality varies asynchronously, the time-shared nature of HS-DSCH introduces a form of selection diversity with important benefits for the spectral efficiency.

UMTS and HSDPA are designed to provide not only the traditional circuit switched services, but also new multimedia services with high quality images and video for personal communication. To enable such flexibility, UMTS allows a user/application to negotiate the bearer characteristics that are the most appropriate for carrying the information [18]. Different classes of bearers have been standardised to provide optimal service to the application's QoS requirements. The number of classes is limited to four to ensure that the system is capable of providing reasonable QoS resolution [33].

3.2 QoS Architecture

3.2.1 UMTS QoS Classes

The 3GPP created a traffic classification attending to their general QoS requirements. The QoS refers to the collective effect of service performances that determine the degree of satisfaction of the end-user of the service. The four classes are [33]:

- Conversational class.
- Streaming class.
- Interactive class.
- Background class.

The main distinguishing factor between the classes is how delay sensitive the traffic is. Conversational class is meant for traffic very delay sensitive whereas background class is the most delay insensitive.

The traffic corresponding to the conversational class refers to real time conversation where the time relation between information entities of the stream must be preserved. The conversational pattern of this type of communication requires a low end-to-end delay to satisfy the stringent requirements of human perception. A service example is telephony speech, voice over IP or video conferencing. According to [34], HSDPA focuses on streaming, interactive, and background traffic classes but not on conversational traffic, which will therefore not be discussed. The main characteristics of these classes are described in following subsections.

3.2.2 Streaming UMTS QoS Class

3GPP defines this scheme as a one-way transport that applies to real-time streaming. The fundamental characteristic of this QoS class is that the communication has to preserve the time relation (or variation) between information entities (i.e. samples or packets) of the stream, although it does not have any requirements on low end-to-end transfer delay. In order to allow end-to-end delay variations larger than accepted by the human perception, todays streaming applications apply time alignment prior to decoding at the receiving end. The time alignment function initially delays by buffering the received stream before starting the decoding process, which enables to cope with delay variations up to the limits provided by the buffering capabilities. Then, the client can start playing out the data before the entire file has been transmitted.

3.2.3 Interactive and Background UMTS QoS Classes

[35] defines the interactive class as a Non Real Time communication where an on-line end user requests data from a remote equipment. The communication is characterized by the request response pattern of the end user. Therefore, one of the key properties of this scheme is the service response time. The service response time can be defined as the period elapsed since the instant of the data request until the end of the message reception, which determines the degree of satisfaction perceived by the end user. Another characteristic of the interactive services is that the message must be transparently transferred. Moreover, interactive traffic can be bursty. Typical services of interactive class are web browsing, wap, e-mail service (server access), data base retrieval, e-commerce, network games, etc.

The background class is also defined by [35] as a Non Real Time communication type that is not delay sensitive and optimised for machine-to-machine data exchanges. the key note is that this class is targeted for applications that do not require an immediate action, and message delays in the order of seconds, tens of second, or even minutes may be acceptable because the destination is not expecting the data within a certain time [33]. This scheme is the most delay insensitive of all the UMTS QoS classes. Some services examples are background delivery of e-mails (server to server), Short Message Service (SMS), Multimedia Messaging Service (MMS), FTP file transfers, fax, etc.

Packet Scheduling Input Parameters

The packet scheduler has access to various input information that it can use to serve the users in a cell. This information can be classified in resource allocation, UE feedback measurements, and QoS related parameters. As indicated in Fig. 3.1 , the HSDPA users report channel quality indicator (CQI) measurements and pilot measurements. However, only the pilot measurement is made available to the RNC for quality based HSDPA access decisions, while the CQI reports are only accessible in the Node-B [5]. The relevant parameters for this paper are described below:

Resource Allocation

HS-PDSCH and HS-SCCH Total Power : Indicates the maximum power to be used for both HS-PDSCH and HS-SCCH channels. This amount of power is reserved by

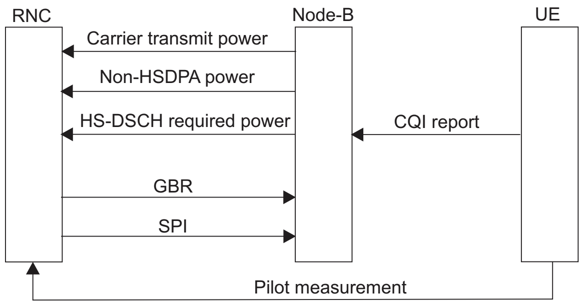


Figure 3.1: Measurement and parameter interface between the RNC, the Node-B, and the UE, [5]

the RNC to HSDPA. Optionally, the Node B might also add unused amount of power (up to the maximum base station transmission power). Note that the HS-SCCH represents an overhead power (i.e. it is designated for signalling purposes), which could be non negligible when signalling users with poor radio propagation conditions.

HS-PDSCH codes : Specifies the number of spreading codes reserved by the RNC to be used for HS-PDSCH transmission.

Maximum Number of HS-SCCHs : Identifies the maximum number of HS-SCCH channels to be used in HSDPA. Note that having more than one HS-SCCH enables the Packet Scheduler to code multiplex multiple users in the same TTI, and thus increases the scheduling flexibility, though it also increases the overhead.

UE Channel Quality Measurements The UE channel quality measurements are used to abstract information about the user's achievable data rates on a TTI basis. Methods used for link adaptation are also valid for packet scheduling purposes(i.e. CQI reports, power measurement on associated DPCH, or the HARQ acknowledgments).

QoS parameters at Node B

Allocation and Retention Priority (ARP) : The Node B has information of the UMTS QoS attribute ARP, which determines the bearer priority relative to other UMTS bearers

Scheduling Priority Indicator (SPI) : This parameter is set by the RNC when the flows are to be established or modified. It is used by the Packet Scheduler to prioritise flows relative to other flows [36]

Common Transport Channel Priority Indicator *(CmCH-PI)* : This indicator allows differentiation of the relative priority of the MAC-d PDUs belonging to the same flow [37]

Discard Timer : Is to be employed by the Node B Packet Scheduler to limit the maximum Node B queuing delay to be experienced any MAC-d PDU.

Guaranteed Bit Rate *(GBR)* : Indicates the guaranteed number of bits per second that the Node B should deliver over the air interface provided that there is data to deliver [34]. It is relevant to note that the corresponding mapping from UMTS QoS attributes (such as the traffic class, or the throughput) to Node B QoS parameters is not specified by the 3GPP (i.e. this is a manufacturer implementation issue). Furthermore no specifications are given for the interpretation of these Node B QoS parameters by the Packet Scheduler.

Other parameters

UE Capabilities : It can limit maximum number of HS-PDSCH codes supported by the terminal, the minimum period between consecutive UE receptions, or the maximum number of soft channel bits the terminal is capable to store.

HARQ manager : Notifies the packet scheduler when a certain HARQ retransmission in required.

3.3 Packet scheduling in HSDPA

QoS aware packet scheduling for Release 99 dedicated channels is traditionally conducted as a function of the users traffic class (TC), traffic handling priority (THP), allocation retention priority (ARP), and potentially also other UMTS bearer attributes as discussed in [38]. However, for HSDPA, the TC and THP information is not available in the Node-B for MAChs packet scheduling, so a new QoS interface has been defined for HSDPA between the RNC and the Node-B. Fig. 3.2 shows an overview of the HSDPA specific QoS parameters that the RNC can set as well as the feedback measurements from the Node-B to the RNC [21].

The HSDPA QoS parameters that can be used to guide the behaviour of the MAC-hs scheduler include the guaranteed bit rate (GBR), the scheduling priority indicator (SPI), and the discard timer (DT).The feedback measurements from the Node-B to the RNC in Fig. 2 (the carrier transmit power, the average non-HSDPA power, the HS-DSCH

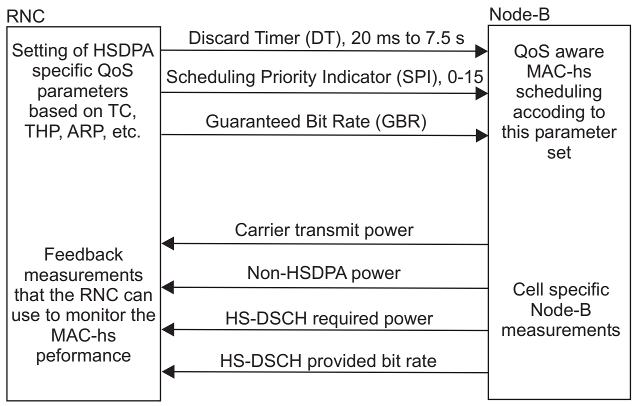


Figure 3.2: HSDPA QoS interface between the RNC and the Node-B, [5]

required power, and the HS-DSCH provided bit rate) can be used for HSDPA admission control decisions as discussed in [5].

3.3.1 Packet scheduling principles

We can define the operation task of the packet scheduler as :

1. To select a user to be served in every TTI, from those connected to the cell. AND
2. To maximize the cell throughput satisfying the QoS attributes belonging to the UMTS QoS classes of the cell bearers

Figure 3.3 illustrates the flow of actions by which the Node B scheduler selects the user to be served. To be noted that CQI represents Channel Quality Indicator while TCP stands for Transmit Power Control.

It can be observed that the Node B scheduler dictates the distribution of the radio resources among the users in the cell. A key aspect in the behaviour of packet schedulers has been pointed out in [39]: the scheduling algorithms that reach the highest system throughput tend to cause the starvation of the least favourable users (low G Factor users). This behaviour interacts with the fairness in the allocation of the cell resources, which ultimately determine the degree of satisfaction among the users in the cell.

3.3.2 Packet scheduling algorithms

The actual methods that implement the fairness policies, due to the pace of the scheduling process, also divides the methods in two main groups:

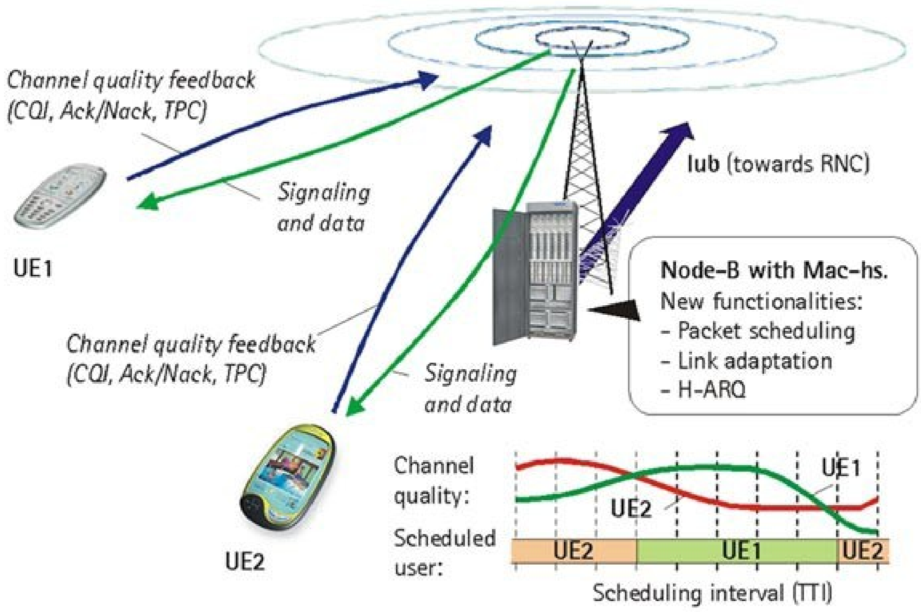


Figure 3.3: Fast Packet Scheduling, Courtesy: Nokia website

1. PS methods that base the scheduling decisions on recent UE channel quality measurements (i.e. executed on a TTI basis) that allow to track the instantaneous variations of the users supportable data rate. These algorithms have to be executed in the Node B in order to acquire the recent channel quality information. These methods can exploit the multiuser selection diversity, which can provide a significant capacity gain when the number of time multiplexed users is sufficient [40].

2. PS methods that base their scheduling decisions on the average users signal quality (or that do not use any users performance metric at all).

Fast Scheduling Methods

Maximum C/I (Max. CI) This scheduling algorithm serves the user with largest instantaneous supportable data rate (in every TTI). Although this principle has benefits in terms of cell throughput, it lacks throughput fairness because users under worse average radio conditions are allocated lower amount of radio resources. Nonetheless, since the fast fading dynamics have a larger range than the average radio propagation conditions, users with poor average radio conditions can still access the channel.

Proportional Fair (PF) This algorithm serves the user with largest relative channel quality [41]:

$$P_i = \frac{R_i(t)}{\lambda_i(t)} \quad i = 1, .., N \tag{3.1}$$

where $P_i(t)$ - user priority;
$R_i(t)$ - instantaneous data rate experienced by user i if it is served by the PS;
$\lambda_i(t)$ - user throughput.
The main property of this algorithm is that it asymptotically allocates the same amount of power and time resources to all users if their fast fading are iid (identically and independently distributed) and the rate $R_i(t)$ is linear with the instantaneous EsNo [41]. It is worth mentioning that the very last assumption does not hold in HSDPA due to the limitations of the AMC functionality.

Fast Fair Throughput (FFTH) This method aims at providing a fair throughput distribution among the users in the cell (in a max-min manner), while taking advantage of the short term fading variations of the radio channel.

Slow Scheduling Methods

Average C/I (Avg. CI) This scheduling algorithm serves in every TTI the user with largest average C/I with backlogged data to be transmitted. The default averaging window length employed in various studies for the average C/I computation is 100ms.

Round Robin (RR) In this scheme, the users are served in a cyclic order ignoring the channel quality conditions. This method outstands due to its simplicity, and ensures a fair resource distribution among the users in the cell.

Fair Throughput (FTH) There are various options to implement a fair throughput scheduler without exploiting any a priori information of the channel quality status [42].

Table 3.1 below summarizes the Packet Scheduling methods and their properties [38]

3.3.3 Packet Scheduling Performance

Diverse studies have been analysed the influence of QoS aware schedulers on overall HSDPA network performance. Approaches vary from focusing on the shared responsibility of QoS algorithms in the RNS [43], to analysing the QoS demands of the different traffic

PS Method	Scheduling rate	Serve order	Radio Resource Fairness
Round Robin (RR)	Slow (≈100 ms)	Round robin in cyclic order	Proportional throughput fairness & Same amount of average radio resources
Fair Throughput (FTH)	Slow (≈100 ms)	Served user with lowest average throughput	Max-min throughput fairness
C/I based (CPI)	Slow (≈100 ms)	Served according to highest slow-averaged channel quality	Unfair distribution of radio resources in favour of high G Factor users
Proportional Fair (PF)	Fast (≈ Per TTI basis)	Served according to highest relative instantaneous channel quality	Proportional throughput fairness & Same amount of average radio resources under certain assumptions
Fast Fair Throughput (FFTH)	Fast (≈ Per TTI basis)	Served according to highest equalized relative instantaneous channel quality	Max-min throughput fairness under certain assumptions
Maximum CI (Max CI)	Fast (≈ Per TTI basis)	Served according to highest instantaneous channel quality	Unfair distribution of radio resources in favour of high G Factor users

Table 3.1: Summary of Analysed Packet Scheduling Methods

classes, their throughput and delay characteristics [44]. Multiple aspects of QoS provisioning are handled by a proposed packet scheduler that adapts to varying traffic and radio channel conditions in [45].

[6] outlines different options for implementing QoS mechanisms and some of the challenges that arise with the new distributed architecture for HSDPA. Three mechanisms can be used to implement QoS control for HSDPA:

- Dynamic allocation of transmission resources for HSDPA (power and channelization codes used by MAC-hs PS);
- Quality based HSDPA access control of new HSDPA users;
- QoS aware MAC-hs packet scheduling.

The analysis of QoS aware MAC-hs packet scheduling is done based on the concept of available HSDPA cell capacity (Fig. 3.4) being time-variant due to dependence on the number of allocated users and their experienced radio channel quality.

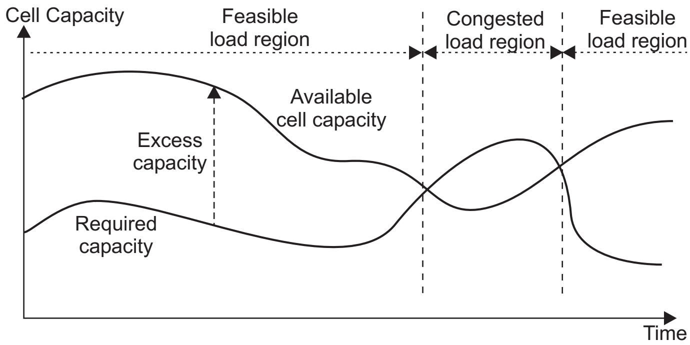


Figure 3.4: Time-variant HSDPA cell capacity, excess capacity, and required capacity. [6]

While operating in the feasible load region, a scheduler can use each user's SPI to apply a hard or soft prioritization strategy. The former will only benefit few users with high SPI. Using the latter approach a MAC-hs packet scheduler can be designed to take advantage of the so-called multi-user scheduling diversity gain, which typically increases with the number of users the scheduler can choose from [46].

The 3GPP specifications omitted information related to the behaviour of MAC-hs scheduler according to SPI settings, therefore it is left up to the Node B manufacturer. After this behaviour is defined, the scheduling algorithm can be designed by following the general mathematical framework formulated in [47], where the first step is to define utility functions for each user depending on its GBR and SPI. In the congested mode the MAC-hs scheduler can no longer serve all the allocated users at their GBR.

By means of dynamic system simulations for different traffic types, the gains in performance of HSDPA over Release 99 for different services are outlined and explained.

The highest cell throughput is obtained for best effort traffic with no QoS constraints and PF scheduling, where the cell capacity reaches 2.2 Mbps for HSDPA, which is 180% higher than that obtainable for Release99. For CBR streaming, in the scenarios tested, approximately 11 users per cell were supported for HSDPA, which yields a gain of 75% over Release99. These results are obtained by using a QoS-aware admission control algorithm, without it the streaming capacity decreasing by approximately 25%.

3.4 Conclusions

In this chapter we analyzed the four QoS classes defined by 3GPP. Closer attention is given to the streaming, interactive and background classes that HSDPA focuses on.

Due to a decrease in the frame size to 2ms, packet scheduling is called Fast Scheduling in HSDPA. Packet scheduler is also moved from MAC layer of RNC to in physical layer

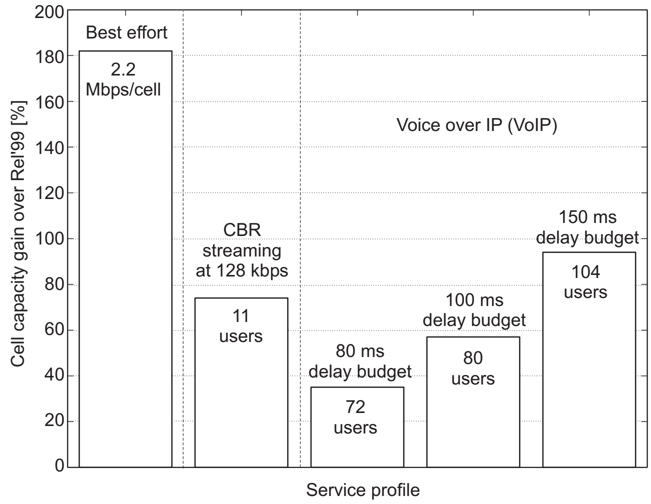


Figure 3.5: HSDPA capacity gain over Release99, [6]

of Node B hence improves the scheduling speed. Besides the Fast scheduling methods in which the scheduling decision is made based on the instantaneous value, slow scheduling methods decide the transmitted user and rate based on the average values. There is a trade off between cell throughput and user fairness among the scheduling methods. The selection of packet scheduling algorithm depends upon the specific situation where either throughput or fairness is biased.

From the results it has been concluded that Max CI is the scheduling method providing highest HSDPA cell capacity for best effort NRT traffic, although the unfairness of this policy causes the starvation of users under poor average radio propagation conditions. This starvation has very negative effects on the performance of the TCP protocol, and moreover, it can cause the dropping of the poor users if the starvation is prolonged.

When minimum user throughput guarantees are to be provided to the overall user community, the Proportional Fair scheduler clearly outperforms the remaining analysed schedulers for all the tested cases.

At this part in the paper we have touched on all the points that make part of the HSDPA model so that we can build a system simulator model based on it. The thesis will continue with a discussion on the network simulators in use, and focusing on one of the preferred tools in the industry, OPNET Modeler. The modeller and the built-in UMTS model are analysed with a view to understanding the means to changing the default model to match HSDPA specifications.

Chapter 4

UMTS Simulation Model

Previous chapters have discussed background information on the HSDPA specification. This chapter will introduce the simulation software and UMTS model used as a basis for building an HSDPA model. It will focus on features of UMTS specifically needed to be re-designed in order to achieve a valid HSDPA simulation system. The design choices are based on the differences between the two releases presented in previous chapters.

4.1 UMTS model

As new telecommunication technologies are brought forward from the abstract plane to real-world environments, a means of testing their applicability, usability and performance is needed. This may be achieved by means of experiments but with some significant drawbacks such as high costly and complex, inability to fully control the testbed environment to be able to repeat tests in a manner that proves their effectiveness and validity.

The two main network simulators currently in use to model the behaviour UMTS networks in academia, commercial and industrial communities[48] are NS2[49] and OPNET Modeler[50]. NS2 is discrete event simulator targeted at networking research, with substantial support for simulation of TCP, routing, and multicast protocols over wired and wireless (local and satellite) networks. It includes modules for both UMTS and HSDPA. As an open source software, it is widely used by the research community but it has had little use in the industry due to a lack of formal support (only user groups provide required information).

OPNET Modeler provides a comprehensive development environment supporting the modelling of communication networks and distributed systems. Both behaviour and performance of modelled systems can be analysed by performing discrete event simulations. The OPNET Modeler environment incorporates tools for all phases of a study, including model design, simulation, data collection, and data analysis. It also includes a UMTS model that emulates the network architecture using the elements in Figure 4.1.

4.2 UMTS Model features and limitations

The main features to have been built into the model are outlined bellow:

Power control Outer loop power control is supported;

GPRS attach The GPRS attach procedure informs the SGSNs when the user equipment (UE) is at power-on and of its GPRS capability;

PDP context activation On receipt of PDUs (protocol data units), the UE or network activates a PDP (Packet Data Protocol) context if one is not already activated. The PDP context activation includes the requested QoS (Quality of Service) profile associated with the traffic class of the PDUs received.

RLC Modes Three RLC modes are supported: acknowledged mode (AM), unacknowledged mode (UM), and transparent mode (TrM). RLC modes impact throughput and delay due to their different algorithms

Admission Control Two admission control algorithms are modelled: a default algorithm and a throughput-based algorithm.

Priority handling of data flows based on traffic class at MAC - Each traffic class is assigned a different priority and the MAC can handle data flows of different priority levels;

Traffic classes The four traffic classes defined in UMTS are supported: conversational, streaming, interactive, and background.

One QoS profile per traffic class of a UE - Each traffic class is associated with a configurable QoS profile (consisting of: data rate, priority level, preemption capability, vulnerability and more). This QoS profile is the QoS requested by the UE in the PDP context activation procedure.

W-CDMA air interface (FDD mode only) - Only the FDD mode is supported. Packet dropping probability is based on curves obtained from another set of simulations of the W-CDMA air interface (accurate to the waveform level).

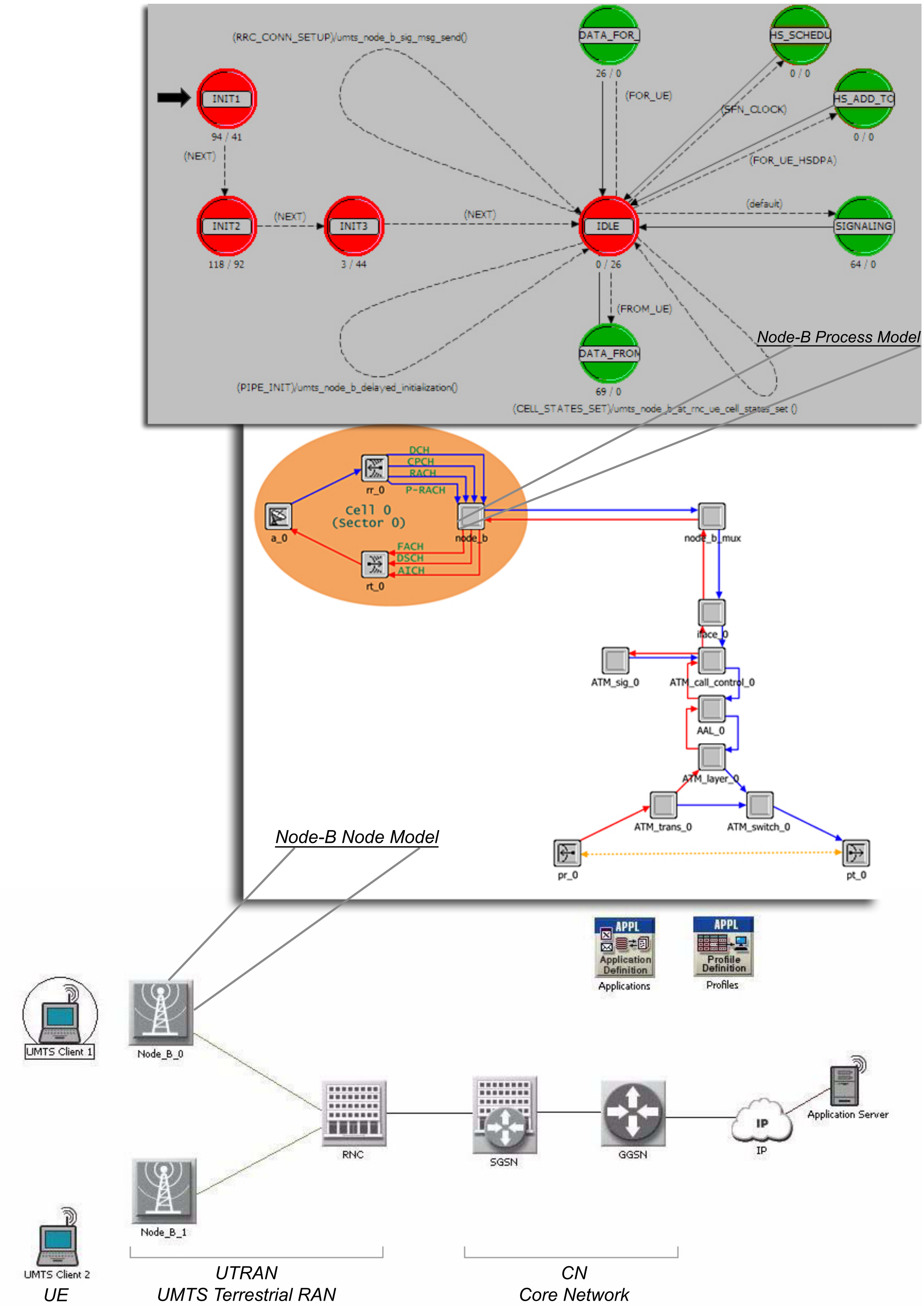


Figure 4.1: UMTS Node and Process models in OPNET

CELL_DCH (DSCH / DCH) - The model supports the DSCH (Downlink Shared Channel), which can be used by UEs in CELL_DCH state for downlink communications. Each RNC deploys a single DSCH for each cell it manages.

CELL_FACH (FACH / RACH) - A UE in the CELL_FACH state uses the RACH channel for uplink transmissions and the FACH channel for downlink transmissions. FACH scheduling follows a weighted round-robin approach and allows you to assign weights according to QoS class.

DCH Uplink and downlink dedicated signalling channels (DCHs) are used for all signalling messages between UE and UTRAN/CN. Signalling DCHs are established when needed and modelled with high fidelity. DCHs are configurable on a per-QoS basis for each RNC.

Logical and transport channels UMTS models allow the mapping of logical channels to transport channels, and enable a queueing scheme for logical channel to transport channel multiplexing.
UE can have up to 4 data transport channels for uplink and downlink, and 1 signalling transport channel for uplink and downlink. Three scheduling schemes can be used for logical channel multiplexing in the MAC: Strict Priority, Weighted Round Robin, and Modified Weighted Round Robin.

Intra-RNC hard and soft handovers - UMTS models both hard and soft handovers of the UEs between the Node-Bs of the same RNC. Softer handoversamong the cells/sectors of a Node-Bare also modelled

Although the designers went to great lengths to implement as many features as possible, there are some aspects of the model where work still needs to be done and they need to be taken into account when designing a simulation scenario.

GPRS detach It is assumed that a UE remains attached for the remainder of the simulation.

QoS Parameters (up/down-link data rates only) Uplink and downlink data rates are the only negotiable QoS parameters. Maximum Bit Rate Uplink/Downlink are the desired data rates for a UE. Guaranteed Bit Rate Uplink/Downlink are the lowest data rates a UE can accept. The UE will accept either maximum bit rate or guaranteed bit rate through a negotiation or renegotiation process initiated by the RNC.

QoS negotiation/renegotiation QoS negotiation during the relocation procedure is not supported. QoS negotiation/ renegotiation are solely initiated by the RNC and the UE will always accept the negotiated/ renegotiated bit rates from the RNC.

IP Support between RNC and Node-B (only UDP for Iub, IP or ATM) Only UDP will be used for Iub transport layer. Only IP transport will be supported for Node-B to RNC transport. Simultaneous use of ATM and IP interfaces for Node-B to RNC transport will not be supported.

Spreading tree No support for parallel spreading codes on dedicated channel. Dedicated channels do not support parallel codes. This limits the maximum data rate a single channel can support.

HSDPA No support for high-speed downlink packet access.

No mobility prior to attachment No mobility is modelled prior to the attachment of the UEs to the closest Node B. UEs begin monitoring their location after attachment

Inter-RNC / inter-SGSN handovers UE handovers are not supported between RNCs or between SGSNs

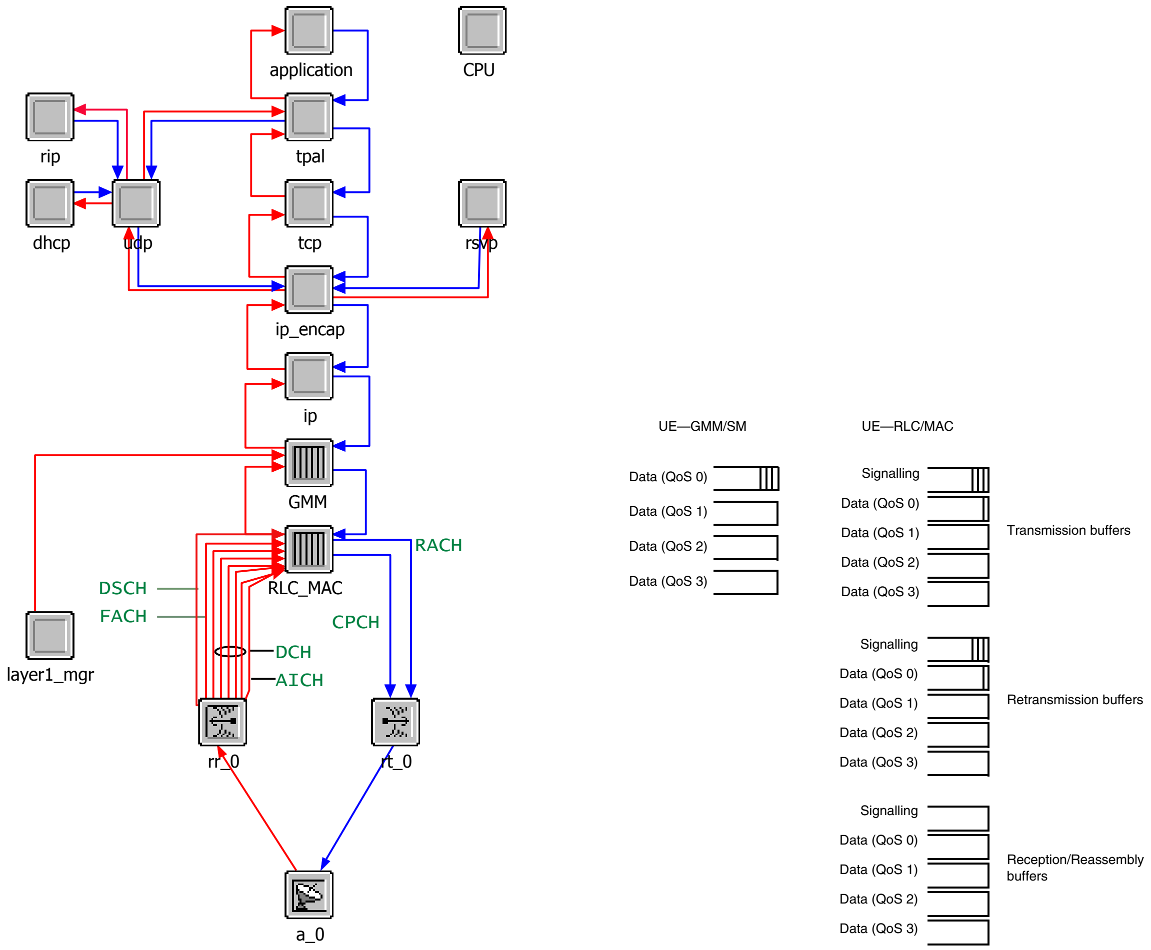
FACH Handover Handover is not supported for UEs using the common channels FACH/RACH (that is, the UEs in the CELL_FACH state).

4.3 UE

Three types of UE models are implemented in OPNET: simple mobile stations (umts_station), advanced workstations (umts_wkstn), and advanced servers (umts_server). Mobile nodes have the capability of supporting movement vectors associated in order to simulate movement throughout the simulation.

The advanced UMTS workstation and server model shown in Figure 5.5b includes an application layer that feeds directly into the GMM layer. It also includes the RLC/MAC layer, a radio transmitter and receiver, one antenna and the full TCP(UDP)/IP protocol stack between the application layer and GMM layer.

The GMM layer contains functions from the GMM, GSM, and RRC layers. It has mobility management functions (such as GPRS attach), session management functions (such as PDP context activation), and radio resource control functions (such establishment and release of radio bearers). The RLC/MAC layer contains the RLC and MAC layers. It includes priority handling of data flows, the three types of RLC modes, and segmentation and reassembly of higher-layer packets.



(a) UE Node Model

(b) Queue Structure for GMM and RLC/MAC Layer, [7]

Figure 4.2: UE - Node Model and Queue Structure

The links between the radio transmitter and the RLC/MAC layer and between the radio receiver and the RLC/MAC layer represent transport channels. On the uplink, there can be one random access channel (RACH), one common packet channel (CPCH), and one dedicated channel (DCH) where signaling and data traffic converges. Each transport channel in the dedicated channel has a unique spread code that distinguishes it from other transport channels.

On the downlink, there can be one forward access channel (FACH), one downlink shared channel (DSCH), one acquisition indicator channel (AICH), and one dedicated signaling channel per user, and up to four data channels. The number of signaling and data channels on the downlink is equal to the number of signaling and data channels on the uplink; the exception to this is the DSCH, which has one extra channel. Each channel is assigned a different spread code and traffic on all channels can be sent simultaneously.

The RLC is assigned a logical channel according to an applications DiffServ characteristics. MAC uses a queuing scheme in combination with logical channel weights/priorities to multiplex and schedule data from logical to transport channels. Logical channels transport control/data between L2/RLC and L2/MAC. Transport channels transport data between L2/MAC and L1. Users can map higher layer data to logical channels using either ToS or DiffServ priority handling, and multiplex logical channels to transport channels using a queuing scheme. This capability allows custom classes to be defined (i.e., prioritizing certain cell phone traffic sources over others), and increases the granularity of application performance metrics to observe scheduling behavior.

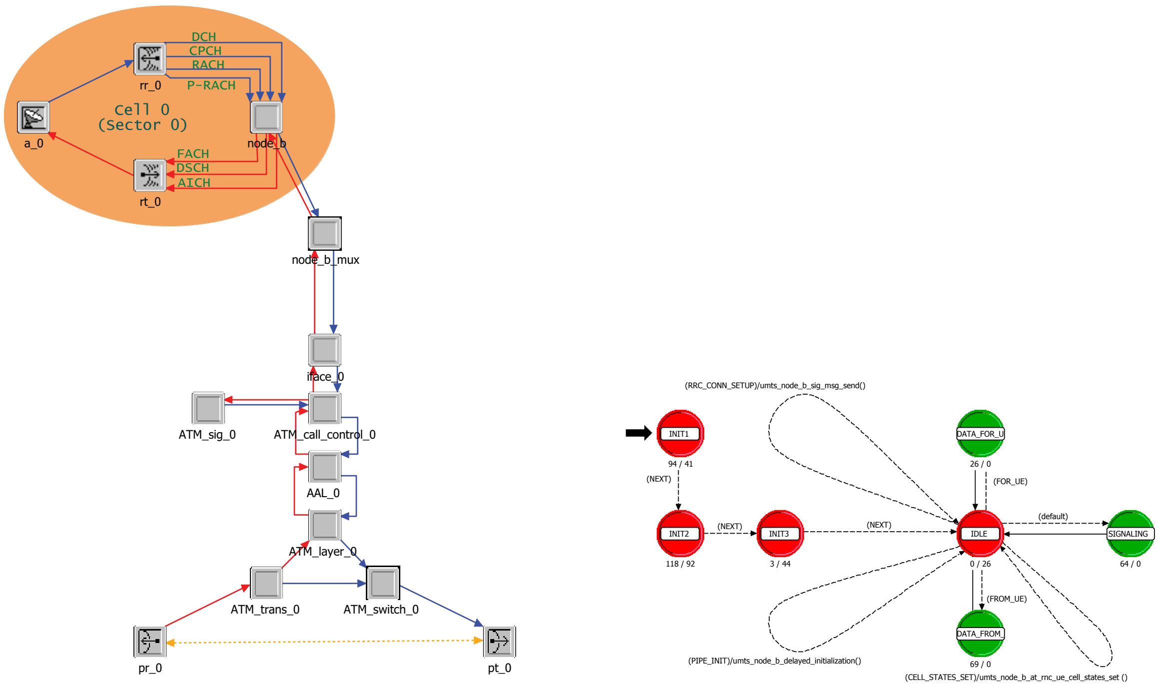
The queue structure at the GMM and RLC/MAC layers is shown in Figure 4.2b. The GMM layer has four queues, one for each QoS class the UE can support. When a data packet from the application layer arrives at the GMM layer, it is forwarded to the RLC/MAC layer if a channel has already received a RAB setup message for the RAB of the packets QoS class. Otherwise, the packet is enqueued at the GMM layer in the queue corresponding to its QoS profile. The RLC/MAC layer uses queues to transmit packets coming from higher layers, to retransmit packets in RLC acknowledged mode, and to receive packets from lower layers and reassemble them to build the PDUs from these packets. Each category requires one queue for signaling and four queues for each QoS supported.

4.4 Node B

The Node-B manages the networks air interface for UEs in the same sector as the Node-B. There are both ATM and IP-enabled Node-Bs. The model suite includes a single-sector Node-B, a three-sector Node-B, and a six-sector Node-B. An RNC connects to one or more Node-Bs to communicate with the UEs of the network and to manage multiple calls.

The Node-B node models include one node_b processor module for each sector it manages. The node_b processor module is connected to an ATM or IP protocol stack, a transmitter module, and a receiver module. Each packet stream between the node_b module and the transmitter represents a downlink channel and each stream between the node_b module and the receiver represents an uplink channel. In the downlink direction, packets are forwarded to the transmitter on the FACH or DSCH streams, or on the dedicated channel via the kernel Procedure (KP) op_pk_deliver(). In the uplink direction, all packets travel over the RACH, CPCH (not modeled in the current release), or DCH streams. All DCH packets converge at the DCH input stream, regardless of their channel or spreading code.

When the simulation starts, Node-Bs initialize the data structures used in the pipeline stages, sets radio transmitter and receiver attributes for all UEs and Node-Bs in the UMTS



(a) NodeB Node Model

(b) UMTS NodeB Process Model

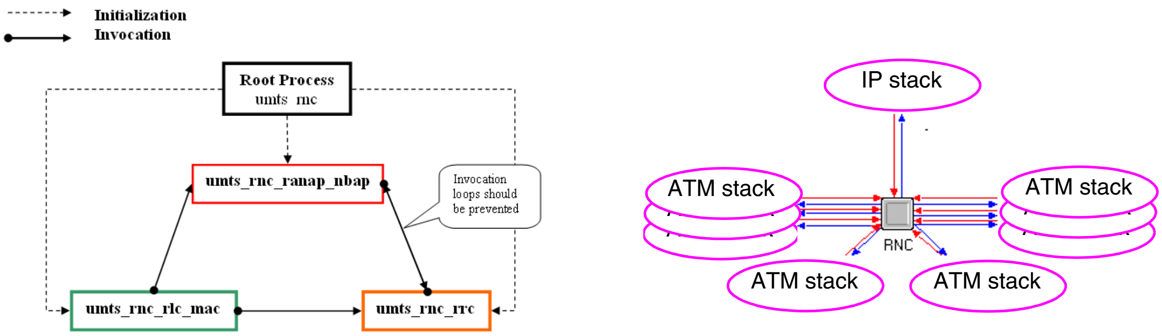
Figure 4.3: Node B - Node Model and Process Model

network (only the first Node-B to start performs this task), and initializes ATM-VC or IP connections to the RNC for each QoS class and signalling data channel.

Besides relaying packets between UEs and the RNC, the Node-B also assists the RNC with radio resource management through NBAP (Node-B Application Protocol) signalling messages. When the RNC receives a request to add a new radio link, it informs the Node-B of the addition of this link for the call. The Node-B then responds to the request with assigned spreading code for the radio link. A similar communication happens between Node-B and RNC for radio link deletions. RNC informs Node-B about the deletion request, and Node-B frees the spreading code assigned for that link, before responding to the RNC. When the RNC receives a request to modify a radio link, it informs Node-B of the modification of this link. Once complete, Node-B responds to the RNC.

4.5 RNC

The RNC Node model (4.4b) consists of the ”RNC Manager” and three child processes that perform the functionality of the RNC. The RNC Manager has nine ATM or IP stacks attached to it, one of which connects to the SGSN(s) servicing the RNC. The other eight will connect to Node-B ATM or IP stacks. The RNC process models can determine which type of node exists at the other end of any given connection, so the RNC can connect any of these stacks to either a Node-B or SGSN so long as no more than one RNC connects to it and at least one Node-B connects to it. The total number of supported node-Bs can be increased by adding more ATM or IP stacks to the node structure.

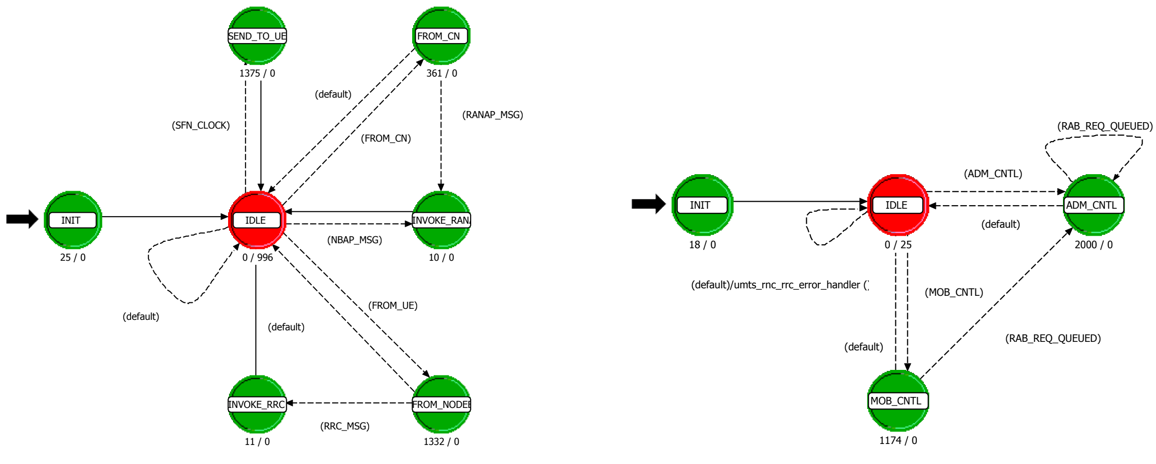


(a) RNC Model Root and Child Processes (b) RNC - Node Model

Figure 4.4: RNC, [7]

The RNC model suite (4.4a)includes four process models: umts_rnc (the RNC Manager), umts_rnc_rlc_mac, umts_rnc_ranap_nbap, and umts_rnc_rrc.

The umts_rnc_rrc process implements initialization and user equipment (UE) signaling with Radio Resource Control (RRC) procedures, including admission control, handover, and mobility related procedures.



(a) RNC RLC/MAC (b) RNC RRC

Figure 4.5: RNC - process models

The umts_rnc_rlc_mac process receives stream/remote interrupts to the RNC module. Once created, it is responsible for invoking other child processes accordingly. It also maintains arrays of queues that each serve a specific purpose: transmission, reception, retransmission, segmentation, and reassembly. Each position in the array represents the set of buffers (or queues) that are assigned to a specific channel. Some of these channels are assigned and released dynamically during the simulation while others are assigned for the duration of the simulation. The RNC designates an equal number of slots in this array for each Node-B it services. A queue array created for a Node-B has the structure depicted in Figure 4.6.

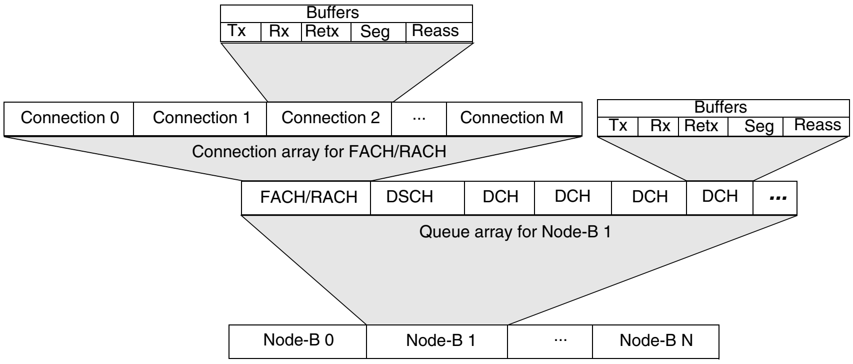


Figure 4.6: RNC - Queue Allocation Structure, [7]

The active connections for the FACH/RACH and DSCH channels are stored in two distinct arrays. When the simulation starts, the RNC dedicates slot 0 to the FACH/RACH and slot 1 to the DSCH, both of which point to the appropriate connection arrays. After startup of the UEs via the GPRS Attach procedure, the RNC establishes a signalling DCH for each UE. As the RNC creates DCHs, it dedicates slots in the array in the section it reserved for the Node-B serving the UE that the RNC establishes the channel for. As the simulation progresses and as the UEs send service request messages to get DCH RABs, the RNC creates channels for the new RABs that do not run over common channels. The RNC also designates unused slots in its queue arrays to service the UEs newly established RABs. If the newly created RAB runs over common or shared channels, a new connection slot is assigned.

4.6 Uu - Air Interface

To model specific W-CDMA behavior, the wireless module includes 13 pipeline stages to model the radio interface. In order to model specific changes in the air interface between the UE and the UTRAN, some of these stages can be modified.

The pipeline stages include modelling for:

- Pathloss model used to compute the received power at the receiver (dra_power.ps.c; Node B - Attribute): vehicular outdoor, pedestrian outdoor, indoor office;

- Background noise pipeline stage (dra_bkgnoise.ps.c) includes thermal noise and noise figure of the mobile and base station receiver;

- Interference noise (dra_inoise.ps.c)is calculated in the following pipeline stages

  RACH and AICH channels (umts_ue_dra_inoise.ps.c; umts_utran_dra_inoise.ps.c);

  DCH and FACH channels (umts_ue_dra_inoise.ps.c; umts_utran_dra_inoise.ps.c);

- Bit-Error-Rate pipeline stage (dra_ber.ps.c) modified to include the signal-to-noise ratio (SNR) versus block error ratio (BLER) curves that depend on the coding scheme and rate and transmission time interval for each transport channel, and the transport format combination chosen. The model supports convolutional codes rate half and rate third in AWGN and in multipath conditions with three equal paths and assumes perfect power control. Bounds on the BLER have been developed under these different conditions as specified in [24].

  For a specific target block error rate the required E_b/N_0 is closely approximated as:

$$(E_b/N_0)_{dB} \approx \frac{logB + b_0 - 2 - logP_B}{-b_1} \tag{4.1}$$

4.7 Conclusions

We considered the features and limitations of the UMTS model implemented in OPNET, and we can conclude that it is adequate to be modified to match our research goals.

The UE node model implements the DSCH transmission channel and it uses data structures pointing to packet objects and queues that can be used to track information needed for AMC and HARQ. The model also accepts changes to it's attributes list, therefore we can integrate the specified changes according to 3GPP standards.

The Node-B's MAC has been clearly designed using a state machine, making it possible to add packet scheduling functionality by linking new states to the existing ones.

All of the radio bearer settings and transport channel to physical channel mappings are set in the RNC module. They need to be adjusted and extended the make use of the multi-code HS-DSCH transport channel, that can be modeled based on the existing DSCH channel.

Chapter 5

HSDPA Simulation Model Design

5.1 HSDPA model

This section provides information about the parts of the UMTS model that have to go through changes in order to emulate HSDPA functionality. The changes to the model have been put in effect by changing the existing UMTS model. Changes have been made in the coding implemented in each model, the state machines that control their functions and to attributes of various models in order to emulate 3GPP technical documentation of UMTS Release 5 (HSDPA).

In order to make changes to OPNET process models we need to understand how processes are buit and how they communicate with each other. Processes in OPNET Modeler are designed to respond to interrupts and/or invocations. Interrupts are events that are directed at a process and that may require it to take some action. They may be generated by sources external to a process group, by other members of a process group, or by a process for itself. Interrupts typically correspond to events such as messages arriving, timers expiring, resources being released, or state changes in other modules. After a process has been invoked due to an interrupt, it may invoke other processes in the group and these may in turn invoke other processes, etc. An interrupt's processing is completed when the first process that was invoked blocks. The Process Editor expresses process models in a language called Proto-C, which is specifically designed to support development of protocols and algorithms. Proto-C is based on a combination of state transition diagrams (STDs), a library of high-level commands known as Kernel Procedures, and the general facilities of the C or C++ programming language. A process model's STD defines a set of primary modes or states that the process can enter and, for each state, the conditions that would cause the process to move to another state. The condition needed for a particular change in state to occur and the associated destination state are called a transition.

We focused the work on the key features brought by HSDPA:

- Fast link adaptation techniques based on multiple Modulation and Coding Schemes,
- Fast Hybrid Automatic Repeat Request (HARQ) algorithms,
- Short TTI (2ms)
- Scheduling in the Node-B shares the HS-DSCH among the users

5.2 Short TTI

One of the changes implemented in the HSDPA specifications is the short Transmission Time Interval with 3 sub-frames (2ms) instead if 15 (10ms) in the UMTS model. Before making changes to the buit-in model an analysis of the timing delays in the UMTS model is required.

The OPNET environment models the following timing delays:

- Encoder delay
- Processing delay
- Buffering delay
- Propagation delay (configurable)
- IP delay (configurable)

Figure 5.1 shows how these delays are implemented in the model. The encoder delay represents all delay incurred by the encoder in the first and subsequent frames of a burst (T_{delay}). At the RLC/MAC layer, data is first buffered for one transmission time interval (TTI), which can last from one to eight times the length of one radio frame (10-80 ms). Data is then processed (coded, interleaved and so on).

The processing delay is the time required by the transmitter and receiver to process the packet. The processing delays at the UE, RNC, and SGSN/GGSN are labeled t_{pc1}, t_{pc2} and t_{pc3}, respectively.

At the UE and UTRAN, packets can be sent on a frame boundary if the channel is not already busy. For example, if a packet at the UE is received from higher layers at least t_{pc1} before the frame boundary, the packet can be sent at the next frame boundary, if it is available. Otherwise, it waits an additional transmission time interval.

At the receiver, the buffering time (T_{buffer}) represents the time needed by the receiver to buffer all of the radio frames required to decode the signal. The propagation delay

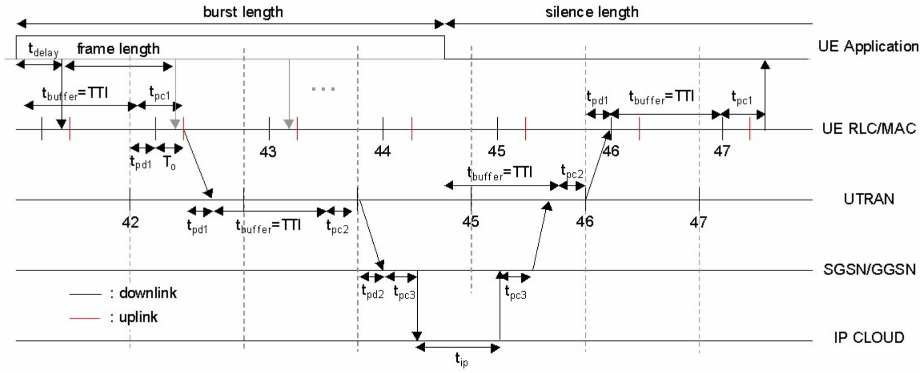


Figure 5.1: Delay in RAN and CN network, [7]

is based on the distance and on the type of channel link: t_{pd1}represents the propagation delay between the UE and UTRAN and t_{pd2} represents the propagation delay between the UTRAN and SGSN/GGSN.

The OPNET implementation of the UMTS specifications allows the operator to pick only from predefined values for the TTI (10, 20, 40 or 80ms). This is done through the UI at the RNC module in the Channel Configuration Attribute. When analysing the process models and the pipeline stages for it is soon obvious that the designers made the decision to hard-code the whole model based on the following assumptions:

- TTI value range = 10, 20, 40, 80
- Acces Slot Length = 15

Upon a closer investigation the following models, processes and pipeliine stages need to be mdified to accomodate for the change to a short TTI:

- umts_defs.h
- umts_rnc_rlc_mac.pr.c
- umts_rnc_ranap_nbap.pr.c
- umts_rlc_mac.pr.c
- rnc_support.ex.c
- umts_layer1_mgr.pr.c
- umts_rach.pr.c
- umts_rnc.pr.m
- umts_node_b.pr.m
- umts_ue_dra_error.ps.c
- umts_ue_dra_power.ps.c
- umts_rach.pr.c

Although consistent changes have been made on all the models that deal with the air medium, one issues still remains. the model loops continuously in the RACH process model during the Preamble-Adquisition procedure. Further debugging showed this to be due to the packets arriving at a simulation time before they are expected (due to the short TTI) at the Node B. The UE ramps up the transmission power in steps until it reaches the amximum allowed and then times out.

The AICH Transmission Timing parameters in the RNC module are used to synchronize the Preamble-Acquisition Indicator transmissions. They are modelled after UMTS specifications with two different configurations for this set of parameters: T_{ppmin} (3 or 4 Access Slots (AS)), T_{pa} (1.5 or 2.5 AS) and T_{pm} (3 or 4 AS) [16]. Changing the number of AS or changing the size of one AS can lead to some packets arriving in time. Unfortunately as these changes do not follow 3GPP standards (they have other dependent models) , they cause the model to crash.

An analysis of the RACH process is needed to further understant the model in question and to allow for future research to be carried out and modify the system accordingly. When the UE is in the CELL_FACH state, the RACH (random access channel) is used to transmit data in the uplink direction. When packets are buffered at the RLC/MAC layer, the RLC/MAC spawns the umts_rach process, which models the random access channel. The umts_rach process model, shown in Figure 5.2, follows the slotted aloha contention algorithm. The process uses the preamble ramp-up procedure to begin sending preambles. Once it receives an acknowledge from the node B, umts_rach notifies the RLC/MAC so that data messages can be sent.

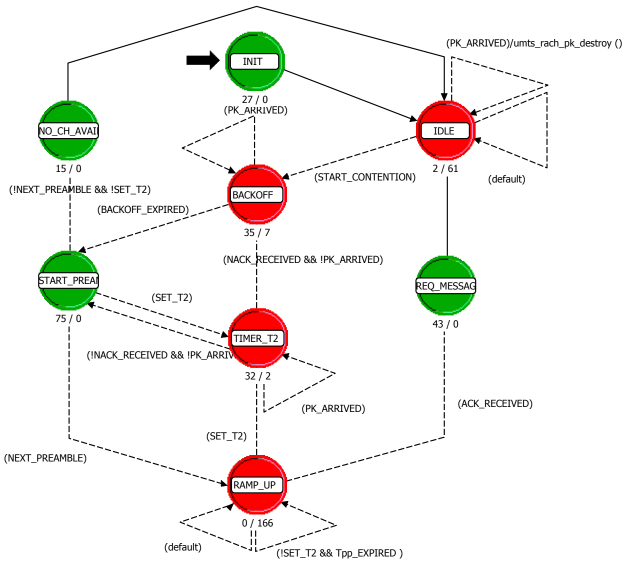


Figure 5.2: UE - RACH Process Model

5.3 AMC and HARQ

In HSDPA a process at the UE measures the SINR for each received packet and reports the corresponding CQI back to the Node-B. Upon receiving the CQI value, the Node-B process can use a corresponding set of values, the Modulation and Coding Set, in order to obtain a specific BLER. Implementing this procedure requires the use of Actual Value Interface (AVI) as specified in [24]. Previous work has been done [51] but not fully released to the research community, therefore we present a possible example table that can be used to map SINR values to an MCS set in 5.3a. For example, a CQI of 16 for a UE of category 7 means that the Node-B can transmit 3565 bits to this UE in the next TTI using 16-QAM modulation and an effective code rate of 0.37.

When the modulation type used by the Node-B transmitting packets over the Uu interface the modulation can change from QPSK to 16-QAM. In order to match this setting the radio transmitter's modulation attribute (5.3b) need to be changed. This attribute specifies the name of a modulation table used to look up the bit error rate as a function of the effective signal-to-noise ratio.

CQI value	Transport Block Size	Number of HS-PDSCH codes	Modulation	Data rate (Mbps)	ECR (per code)	SINR (dB)
0	Out of Range					
1	137	1	QPSK	0.07	0.15	-6.5
2	173	1	QPSK	0.09	0.19	-5.5
3	233	1	QPSK	0.12	0.25	-4
…	…	…	…	…	…	…
16	3565	5	16-QAM	1.78	0.37	14.68
17	4189	5	16-QAM	2.09	0.44	16.59
…	…	…	…	…	…	…

(a) SINR to MCS mapping table for UE categories 7, 8, [51]

Attribute	Value
name	rt_0
channel	(...)
modulation	qam16_ber_snr
rxgroup model	umts_dra_rxgroup
txdel model	umts_dra_txdel
closure model	umts_dra_closure
chanmatch model	umts_utran_dra_chanmatch
tagain model	umts_dra_tagain
propdel model	dra_propdel
icon name	ra_tx
Communication Direction	Downlink

(b) Antenna Modulation Type

Figure 5.3: AVI table and Antenna Modulation Parameters

Due to time restrictions, the current implementation has only the following changes:

- SINR measurement for each packet at UE MAC level

  Extensions to the data structure in the physical layer packet sent on the radio air interface (UmtsT_Physical_Channel_Settings) to include CQI value measurements;

  Added global data structure, referenced by a pointer variable, to hold current values of CQI, SINR and Eb/N0 for all currently connected UEs;

- conversion of SINR value to CQI attribute

New function implemented in the umts_rlc_mac module to convert the SINR value to a CQI attribute. This attribute is stored in the global poiter for later refference;

- Sending the CQI attribute to the Node-B

  Because the implemented model of the physical layer packet was extended, it is updated with the current value of the CQI for the sending UE, before beeing sent over the air interface.

OPNET suports HARQ with Chase combining only for the WiMMAX module, although the implementation for the HARQ functionality is written to preserve generality of the module as much as possible, so that it can be implemented to the UMTS model as well. The following features are included:

1. Giving an HARQ channel to the caller depending upon the buffer size and buffer aggregation.

2. Accepting an incoming packet and encoding it with HARQ information.

3. Calculating the coding gain of the received packets. This is done by default using an additive SNR scheme.

4. Supporting implicit and explicit acknowledgements

5. ACK processing and retransmittion

This package also defines interfaces that allows the user to model lost transmission packets and/or grants. In some cases, such losses are unavoidable, and the package provides functionality to handle these losses.

5.4 UE model extensions and HS-DSCH

3GPP defined in [20] 12 different categories for the UEs, based on their capabilities. For example, UE category k for k = 1 or 2 can only support data rates up to 1.2 Mbps using 5 simultaneous physical channels (codes) and has minimum inter TTI interval $min_TTI(k)$ of 3. If user u_i from category k is scheduled to transmit in TTI t, the earliest TTI in which u_i can be scheduled next is $t + min_{TTI}(k)$. Theoretically, category 7 can support up to 7.21 Mbps using 10 codes (under perfect link conditions) and has minimum inter TTI interval of 1. Other attributes such as spi, downlink gbr and ue category are introduced in Release 5. For each UE added attributes as shown in Figure 5.4 can be used to emulate those specifications.

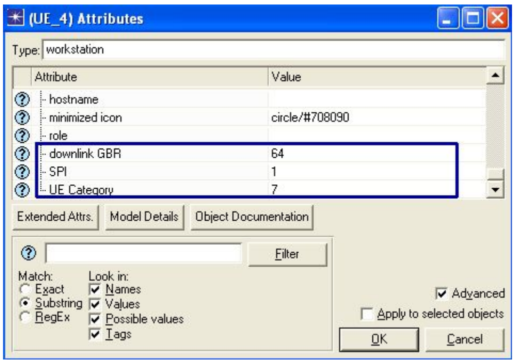


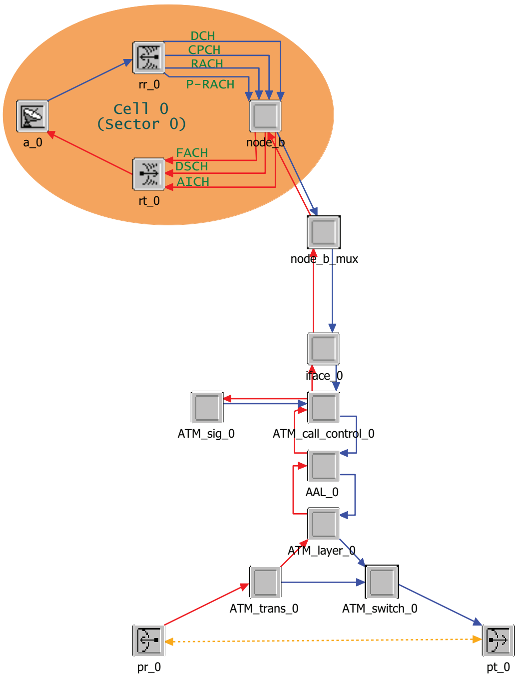
Figure 5.4: UE - HSDPA Attributes

Another key feature of the Release 5 specifications is the addition of the HS-DSCH channel.Instead of dedicated channels as in release 99, HSDPA uses shared wireless downlink channels for high-speed data. A new downlink transport channel HS-DSCH (High Speed Downlink Shared Channel) was introduced, that carries data to the selected UE (or UEs) during each transmission time interval (TTI) of two ms. The transported bits from HS-DSCHs are mapped onto up to 15 physical downlink shared channels (HS-PDSCH) each using a separate orthogonal cdma code with a spreading factor of 16. The associated High Speed Shared Control Channel (HS-SCCH) is used to communicate control information between the UE and Node-B.

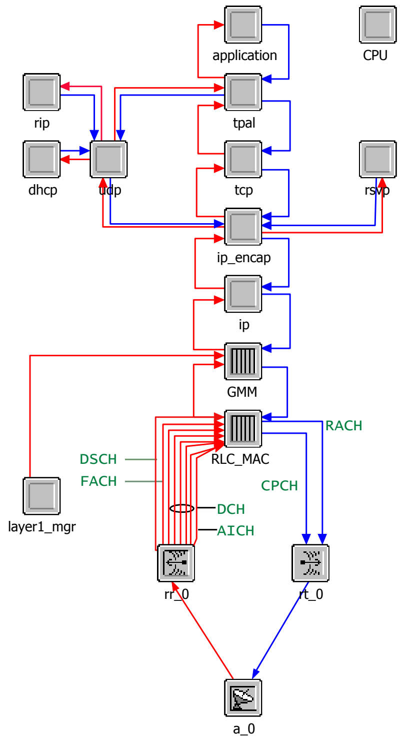
RNC allocates channelization codes and power for HSDSCH and HS-SCCH transmissions based on load and performance measurements provided by Node-B. These measurements include but are not limited to the total carrier power, non-hsdpa power and the HS-DSCH required power. HSDSCH required power is reported per SPI and is an estimate of the total power needed to serve all admitted user with that SPI at their GBR. The RNC also receives an Ec/N0 measurement of the CPICH channel from the new user seeking admission. Based on all these reports, the RNC can estimate whether the user can be granted access without deteriorating the services to the existing users.

The DSCH channel needs to be modified in order to implement HS-DSCH, with additional changes in the RNC and UE models to guide the transmission on the downlink through the new channel.

At the RNC, the Channel Configuration attribute is desigend to model the dedicated, common, and shared transport channels carrying data and signaling traffic. For data channels, channel parameters for each UMTS service class can be configured. To be noted that the configurable transport channel parameters depend on the channel type. For example, RB Mapping info (sub-attribute) does not apply to common channels because it is specified on a per-UMTS-class basis for the UEs in CELL_DCH state. The RB Mapping



(a) Node B - Node Model with Transport Channels



(b) UE - Node Model with Transport Channels

Figure 5.5: AVI table and Antenna Modulation Parameters

Info (sub-attribute of Transport channel Parameters) configures the parameters required to map the radio bearers to different channel types for the UEs that are in CELL_DCH state. The radio bearers for UEs in CELL_FACH state are mapped to FACH and RACH for down link and uplink, respectively. The downlink data transfer will only use the HS-DSCH if the channel mapping are set correctly Figure 5.6.

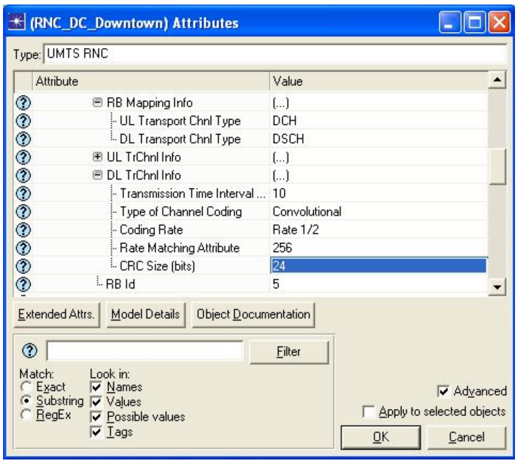


Figure 5.6: RNC Channel Mapping

5.5 Packet Scheduling

The Node-B receives guaranteed bit rate (GBR) and scheduling priority indicator (SPI) values from the RNC, which it can use to make scheduling decisions.

Figure 5.7 shows the process model for an HSDPA Node-B. When a packet destined for a UE arrives from RNC, it is added to the transmission queue of that user in state HS_ADD_TO_BUFFER. During each TTI, in state HS_SCHEDULE, the scheduling candidate set (SCS) is created comprised of all the users that have non-empty transmission queues and are elligible to receive. A user is elligible, if the number of elapsed TTIs since its last reception exceeds its min_TTI. Users are chosen from SCS to be scheduled to receive packets based on the scheduling algorithm being used.

Scheduling algorithms are implemented in OPNET through queuing scheme used on the transport channel queue to multiplex packets from multiple logical channels onto a single transport channel. There are three distinct algorithms implemented that can be varied for each of the four QoS profile configurations:

- Strict Priority - the logical channels' configured priorities will be used by the queuing scheme.

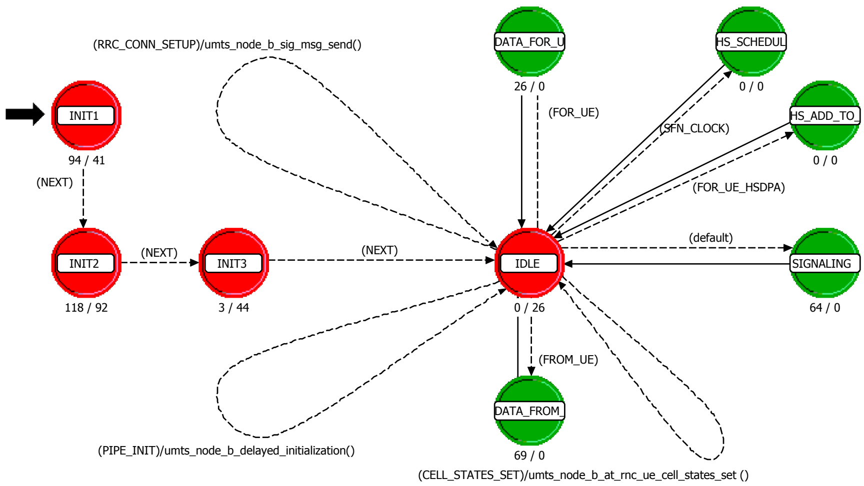


Figure 5.7: HSDPA Node B - process model

- Weighted Round Robin - the logical channels' configured weights will be used by the queuing scheme
- Modified Weighted Round Robin - similar to the standard algorithm but with minor changes

5.6 Conclusions and Future Work

The process of building a network model that is capable of mapping the HSDPA specifications to a working system needs to take into careful considerations the aspects if the system that need to be tested. A model that can be used for packet scheduling performance testing needs to have an accurate mapping of the MAC-hs layer in the HSDPA specifications, as well as implement the short TTI and move the retransmission functionality of the ACK/NACK in the Node-B module. Extending the OPNET Modeler network models, which are rich in features and implemented concepts, requires in depth knowledge of the design used for their coding. Besides the 3GPP standards, an advanced knowledge of C++ and pointer structure use is essential in understanding the Modeler's workflow.

At this stage the model created in OPNET is a proof of concept with implemented features of HSDPA rather than a full fledged model, but it shows the capabilities of the simulation environment and means of extending it. Further applications of a HSDPA model would require further extensions with a few features to be implemented as follows:

- Multiple code allocation / multiple HSPSDCH physical channels mapped to the HSDSCH transport channel to achieve high downlink speeds (up to 10.8Mbps)

- Implement the use of the HARQ module taking the WiMAX implementation as an example

- Change the Transmission Time Interval value for a frame to 2ms, the equivalent of 3 slots

- Upgrade the uplink channel to match UMTS R6 specifications (HSUPA) for speeds of up to 5.7Mbps. If implementing HSUPA the following key features need to be implemented:

  HSUPA unlike HSDPA doesn't use AMC, instead it uses Turbo coding with QPSK modulation.

  Power control is used in HSUPA unlike HSDPA

  HSUPA uses 10ms TTI (and optionally 2ms)

  HSUPA supports Intra Node B "softer" and Inter Node B "soft" Handover

  The scheduling in HSUPA is distributed between the UE and the Node B

Future extensions to the model could follow the industry evolution to next generation UMTS Releases Figure 5.8, as it will become challenging not only to model just single technology models, but also emulate the use of hybrid UEs running in UMTS/HSDPA/HSPA+ networks

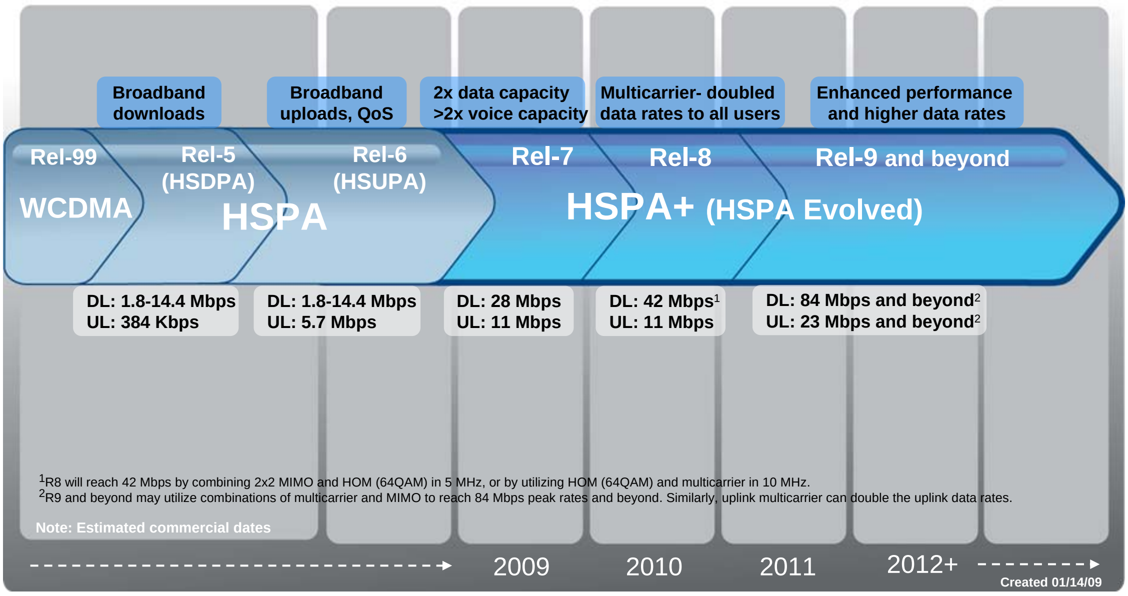


Figure 5.8: HSPA+ R8 Evolution, [4]

Appendix A

Abbreviations

Short Term	Expanded Term
WCDMA	Wideband Code Division Multiple Access
HSDPA	High Speed Data Packet Access
HSPA	High Speed Packet Access
UMTS	Universal Mobile Telecommunication System
3G	Third Generation
IMT-2000	International Mobile Telecommunications-2000
UTRA	Universal Terrestrial Radio Access
FDD	Frequency Division Duplex
TDD	Time Division Duplex
3GPP	3rd Generation Partnership Project
Mcps	Mega-chips per second
RAN	Radio Access Network
UTRAN	UMTS Terrestrial RAN
CN	Core Network
UE	User Equipment
RAB	Radio Access Bearers
RNS	Radio Network Subsystems
RNC	Radio Network Controller
QoS	Quality of Service
SGSN	Supporting GPRS Support Node
GGSN	Gateway GPRS Support Node

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