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Department of Information Technology

PACKET SWITCHED UMTS PILOT SYSTEM

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ABSTRACT

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Universal Mobile Telecommunication System (UMTS) is a mobile communications system specified by Third Generation Partnership Project (3GPP). UMTS supports both circuit switched and packet switched data transmission and enables wireless high bit rate connection to Internet.

The objective of this thesis is to describe 3GPP compliant Third Generation (3G) packet switched pilot system implemented by Nokia Research Center (NRC). A new kind of protocol development approach has been used by combining the protocols made by different tools to one system.

As the traffic is soon changing to packet switched it is necessary to consider the methods for providing Quality of Service (QoS). These methods were studied by using the implemented Pilot system as a test platform. Pilot system was demonstrated in conferencies and it was delivered to several teleoperators.

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Universal Mobile Telecommunication System (UMTS) on Third Generation Partnership Project (3GPP) –organisaation määrittelemä matkaviestin-järjestelmä. UMTS tukee sekä piiri- että pakettikytkentäistä tiedonsiirtoa ja mahdollistaa langattoman, suurinopeuksisen Internet-yhteyden.

Diplomityön tarkoituksena on kuvata Nokia Research Center:n toteuttama kolmannen sukupolven 3GPP yhteensopiva pakettikytkentäinen koejärjestelmä. Työssä on käytetty uutta lähestymistapaa protokolla-kehitykseen, yhdistämällä eri työkaluilla tuotettuja protokollia yhdeksi kokonaisuudeksi.

Liikenteen vaihtuessa lähitulevaisuudessa suurelta osin paketti-kytkentäiseksi on mietittävä keinoja palvelunlaadun takaamiseksi. Näitä keinoja tutkittiin käyttämällä työssä toteutettua koejärjestelmää testialustana. Koejärjestelmää esiteltiin useissa konferensseissa ja se toimitettiin monille teleoperaattoreille.

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Mikko Tirronen

TABLE OF CONTENTS

GLOSSARY

3G	Third Generation
3GPP	Third Generation Partnership Project
AAL2	ATM Adaptation Layer 2
AAL5	ATM Adaptation Layer 5
ALCAP	Access Link Control Application Part
APN	Access Point Name
ASN.1	Abstract Syntax Notation One
ATM	Asynchronous Transfer Mode
BSC	Base Station Controller
BSS	Base Station System
BTS	Base Transceiver Station
CN	Core Network
CS	Circuit Switched
CVOPS	C-based Virtual OPerating System
DHCP	Dynamic Host Configuration Protocol
DS	Differentiated Service
DSCP	Differentiated Service Code Point
EFSA	Extended Finite State Automaton
EFSM	Extended Finite State Machine
EIR	Equipment Identity Register
FP	Frame Protocol
GDB	GNU Debugger
GGSN	Gateway GPRS Support Node
GMM	GPRS Mobility Management
GMSC	Gateway Mobile Switching Center
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
GTP-U	GPRS Tunneling Protocol – User Plane
HLR	Home Location Register
IP	Internet Protocol

ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union, Telecommunication Sector
MAC	Medium Access Control
MSC	Message Sequence Chart
MSC	Mobile Switching Center
MT	Mobile Terminal
MTP3B	Message Transfer Part 3 Broadband
NBAP	NodeB Application Part
NRC	Nokia Research Center
PDCP	Packet Data Convergence Protocol
PDN	Packet Data Network
PDP	Packet Data Protocol
PDU	Protocol Data Unit
PER	Packet Encoding Rule
PPP	Point to Point Protocol
PS	Packet Switched
PSTN	Public Switched Telephone Network
P-TMSI	Packet-Temporary Mobile Subscriber Identity
QoS	Quality of Service
RAB	Radio Access Bearer
RANAP	Radio Access Network Application Part
RB	Radio Bearer
RCS	Revision Control System
RF	Radio Frequency
RFC	Request For Comments
RLC	Radio Link Control
RNC	Radio Network Controller
RNS	Radio Network Subsystem
RNTI	Radio Network Temporary Identity
ROHC	RObust Header Compression
RRC	Radio Resource Control

RRM	Radio Resource Management
RTP	Real-time Transport Protocol
SAP	Service Access Point
SCCP	Signalling Connection Control Part
SCIU	SDT CVOPS Integration Utility
SDL	Specification and Description Language
SDL/GR	SDL/Graphical Representation
SDL/PR	SDL/Phrase Representation
SDT	SDL Design Tool
SDU	Service Data Unit
SGSN	Serving GPRS Support Node
SM	Session Management
SRNS	Serving Radio Network Subsystem
SSCF	Service-Specific Coordination Function
SSCOP	Service-Specific Connection-Oriented Protocol
TCP	Transmission Control Protocol
TE	Terminal Equipment
TFT	Traffic Flow Template
TI	Transaction Identifier
TOS	Type Of Service
UDP	User Datagram Protocol
UE	User Equipment
UMTS	Universal Mobile Telecommunications System
UTRAN	Universal Terrestrial Radio Access Network
VHE	Virtual Home Environment
VME	VERSAmodule Eurocard
VoIP	Voice over IP
VTT	Valtion Teknillinen Tutkimuskeskus (Technical Research Centre of Finland)
WAP	Wireless Application Protocol
WCDMA	Wideband Code Division Multiple Access

1. INTRODUCTION

During the last decade the number of the cellular phone subscribers has grown enormously. Second generation (2G) systems, especially Global System for Mobile Communications (GSM), has been a success story in the mobile communications industry.

The Universal Mobile Telecommunication System (UMTS) is a Third Generation (3G) system, introducing sophisticated packet data handling capabilities such as high bit rates, and enables fast access to Internet. The UMTS has also flexible Quality of Service (QoS) support, needed especially in the packet switched domain.

3G brings along a lot of completely new technology compared to the 2G systems, for example in radio interface the access technique is Wideband Code Division Multiple Access (WCDMA). 3G contains a lot of new features and enables users to have multiple concurrent applications. Thus the protocols that set up the services are complicated. Development work of the 3G systems is slow and it takes many years to have a real product at the market. Pilot systems enable the demonstration of the system when the product systems are still under development. Due to the fact that the competition of equipment supplier contracts is hard, it is beneficial to be able to demonstrate the know-how to possible customers.

This thesis was done as a part of a project where a UMTS Pilot system was implemented. The Pilot system is based on the Third Generation Partnership Project (3GPP) Release 99 specifications. It contains User Equipment (UE), Universal Terrestrial Radio Access Network (UTRAN) and Core Network (CN) emulators for the packet switched domain. System enables connectivity to the real packet core network and also possibility to use real WCDMA radio between UE and UTRAN. The aim of the project was to build a 3G system for demonstration of its applications and behaviour of applications over WCDMA radio. Also one target was to demonstrate the functionality of the 3GPP protocols.

This thesis is divided into two parts, theoretical and practical. Chapter 2 gives an introduction to the UMTS and the services UMTS enables. Evolution from the GSM system to UMTS is described in the same chapter. Packet switched domain from the protocol architecture point of view is described in chapter 3. Bearer Service concept and QoS issues are also introduced in same chapter. At the end of the chapter some principal signalling procedures are shown. In chapter 4 the practical part of the thesis is depicted. It describes how the UMTS Pilot system was implemented. In chapter 5 conclusions are made.

2. UMTS OVERVIEW

The GSM system is circuit switched, designed mainly for voice services. It is constrained by the data rate, therefore it is not suitable for multimedia services. Due to the circuit switching and system architecture, billing is based on used time and thus system is not useful for data transfer. The General Packet Radio Service (GPRS) introduces Packet Switched (PS) domain to the GSM system. The packet switched access enables all-the-time connectivity to network services. The GPRS is as well constrained by data rate and the lack of QoS quarantees.

Though the speech will remain the dominant service up to year 2005, the opportunity for increased revenue will come from advanced data and information services [16]. UMTS is the most promising Third Generation system, providing highly personalised and user friendly mobile telecommunication system. The new mobile services and applications are the driving force when introducing the UMTS to the mass market. By introducing a wideband radio access technique to the system, it is possible to use high bit rates. This makes it possible to have high-quality multimedia session, such as video conference.

GSM has achieved strong market penetration and is going to remain in use for a long time. Due to the fact that UMTS network is expensive to build, the coverage expands slowly. One of the UMTS design basis has been that the system must be able to operate also with GSM system. In the beginning the UMTS is going to work only in cities and the GSM covers sparsely populated areas. Therefore, in the first phase the terminal would be dual mode, enabling seamless access to both systems.

2.1 Services

In the beginning, in 3GPP Release 99, UMTS network is going to provide at least the same services as GSM. This means that from the end user point of view those services behave similarly but their implementation within the network is in many cases different than in GSM. In the later 3GPP Releases (Release 4 and 5), handling of the services within the network will change. [9] For example, circuit switched voice calls could be delivered as packet switched calls (Voice over IP, VoIP).

UMTS contains a service creation environment which allows UMTS operators and other entities create rapidly entirely new services [16]. These services are, for example location based services. Location based services are based on the mobile stations position. This enables a huge number of different kind of applications, such as emergency services and traffic information management. Virtual Home Environment (VHE) is a new service concept. It enables users to feel that they are connected to the home network even when roaming in other networks. VHE takes care that subscribers' service information and profile are transferred between the networks [9]. In the next chapter the packet data network architecture is described in the point of view of network entity, considering also to the evolution from the GSM to UMTS.

2.2 Network Architecture

UMTS is based largely on the GSM technology and the aim is to use as much from GSM with GPRS extensions as possible [9]. Existing network elements, such as Mobile Switching Center (MSC), Gateway GPRS Support Node (GGSN) and Home Location Register (HLR), can be extended to adopt the UMTS requirements, but some network elements must be replaced. Following paragraphs describes how the evolution from the GSM to UMTS is carried out.

The GPRS is the packet data enhancement to GSM, providing efficient handling of bursty traffic. GPRS introduces two new network elements to GSM system: Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN), see Figure 2.1. SGSN, GGSN and HLR compose so called packet core network. Using these nodes mobile can access the Internet.

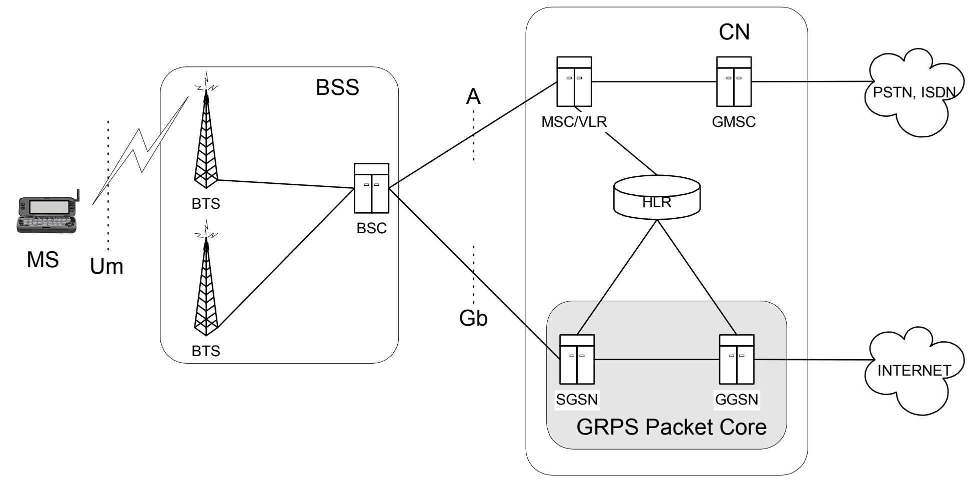

Figure 2.1 GSM with GPRS

The packet core network of the UMTS is evolved from the GPRS network: the names of the elements are same but the functionality is different. The UMTS radio access technique is completely new, called Wideband Code Division Multiple Access (WCDMA). Universal Terrestrial Radio Access Network (UTRAN) consists of one or more Radio Network Subsystems (RNS). RNS is composed of one Radio Network Controller (RNC) and one or more NodeB's, see Figure 2.2.

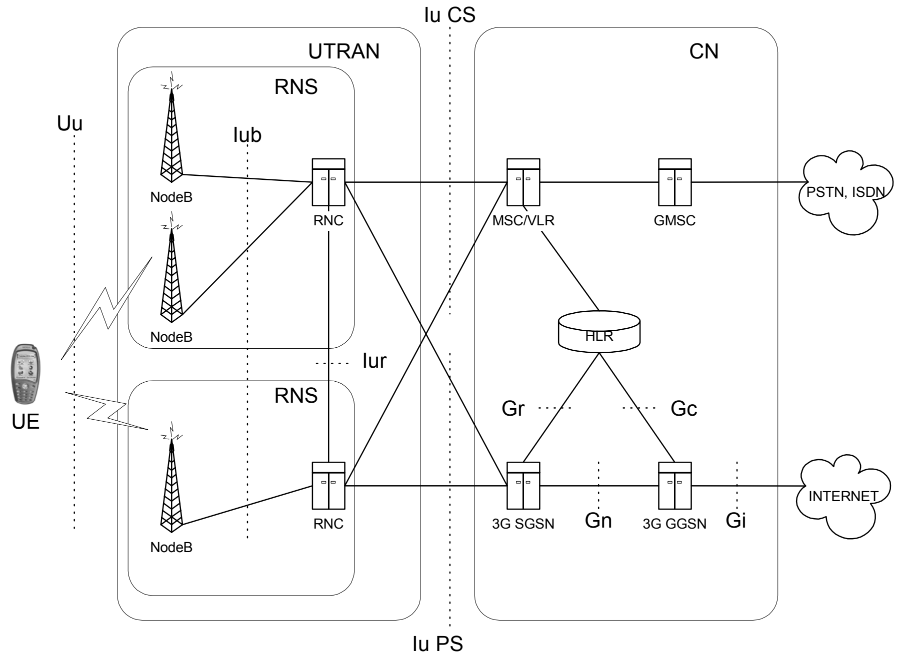

Figure 2.2 UMTS network elements

NodeB logically corresponds to the GSM Base Station [8]. The main function of the NodeB is to perform the air interface Layer 1 (L1) processing. It also performs some basic radio resource management operations, such as the inner loop power control. The inner loop power control means that NodeB commands the UE to adjust its transmission power once in every timeslot (0.667 millisecond) [9]. NodeB consist of one or several cells. Cell is the main logical resource of NodeB. The NodeB has Uu and Iub interfaces. Uu is the air interface in 3GPP vocabulary. Iub is the interface to RNC.

RNC logically corresponds to the BSC (Base Station Controller) in GSM. The main function of RNC is to control radio resources in its domain. The RNC has the Iub, Iur and Iu interfaces. Iu is the interface to core network. Packet switched Iu (Iu-PS) is interface to 3G SGSN and circuit switched Iu (Iu-CS) is interface to 3G MSC. Iur is used in soft handover cases between RNCs.

Soft handover means that data is send and received via several cells of NodeBs to UE. Set of cells through which the UE has simultaneously connection to the UTRAN is called active set [9]. RNC and UE combines incoming data from different NodeBs. The handover between GSM and UMTS is called inter-system handover, which is always hard handover. Hard handover means that the old connection to NodeB or BTS (Base Transceiver Station) is released before new NodeB or BTS is started to use.

The main difference between 2G SGSN and 3G SGSN is that the Mobile Management (MM) entity is divided between RNC and SGSN in 3G. This means that changes in UTRAN are not necessary visible to the PS domain, but RNC handles these situations. The SGSN is mainly responsible for MM related issues like routing area update, location registration, packet paging and controlling the security mechanisms related to the packet communication. [9] The Iu-PS interface connects 3G SGSN to the RNC.

The GGSN maintains the connections towards other packet switched networks, such as Internet [9]. It holds routing information of the UEs connected to it. From the external network's point of view, GGSN acts as a router, hiding the GPRS infrastructure from the external networks.

In UMTS the mobile terminal is called User Equipment (UE). The UE interfaces the user and the Uu interface. It consist of the Mobile Equipment (ME) and the Universal Subscriber Identity Module (USIM). USIM contains data and procedures to identify the user. ME is a physical device that sends and receives information over air interface.

HLR contains permanent subscribers information. The main functions of the HLR are subscriber data and service handling, statistics and mobility management [9]. The HLR of the GSM system can be used by modifying it for UMTS purposes.

3. UMTS PACKET SWITCHED DOMAIN

The UMTS network consists of two main domains, packet switched and circuit switched. In this chapter the packet switched domain is described from protocol models point of view. UMTS bearer concept and traffic classes are described and finally the most essential signalling procedures are depicted.

3.1 Protocol Architecture

Protocol model in UTRAN terrestrial interfaces (Iub, Iu, Iur) are designed according to the same generic reference model, as depicted in Figure 3.1 [5]. In the reference model terrestrial interfaces are divided into 3 planes: control, user and transport network control planes. The model is based on principle that the layers and planes are logically independent of each other and therefore replacing parts of the protocol stack to fit future requirements would be easy [5].

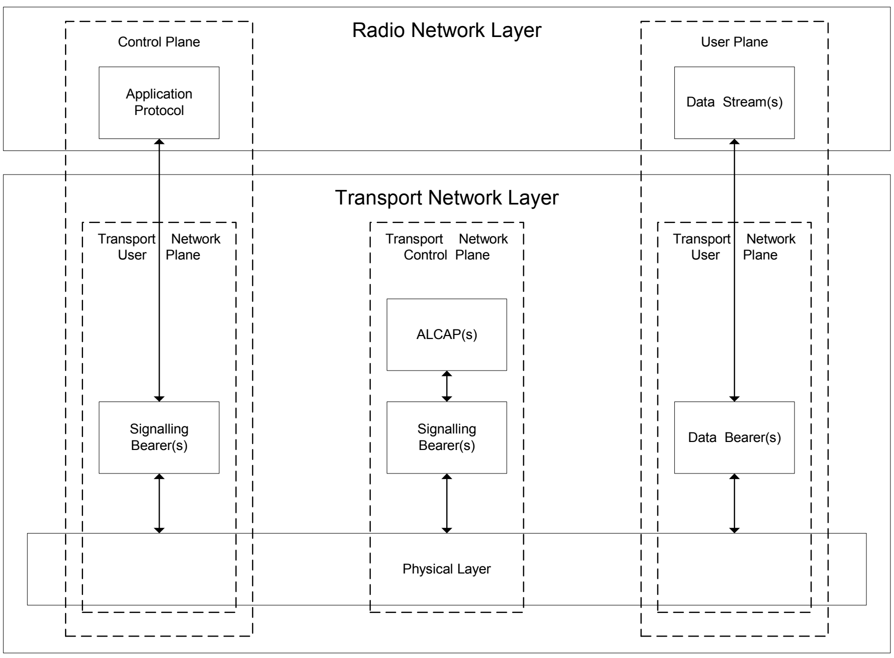

Figure 3.1 General protocol model for UTRAN terrestrial interfaces

UTRAN related issues are visible in the Radio Network Layer. Radio Network Layer consists of application protocols of the control plane and Data Streams of the user plane. Main task of the application protocols is to manage radio access bearers needed by data streams. Data streams are characterised by one or more frame protocols specified for that interface [5]. Transport Network Layer takes care of conveying radio network layer messages by setting and releasing signalling and data bearers.

Transport Network Control Plane includes Access Link Control Application Part (ALCAP) protocol that is used to setup data bearers. ALCAP is generic name for transport signalling protocol. If there are preconfigured data bearers then ALCAP is not needed at all.

3.1.1 Control Plane

Control plane is used for sending information needed for setting up, modifying and releasing bearers between network elements. Control plane protocols are depicted in Figure 3.2.

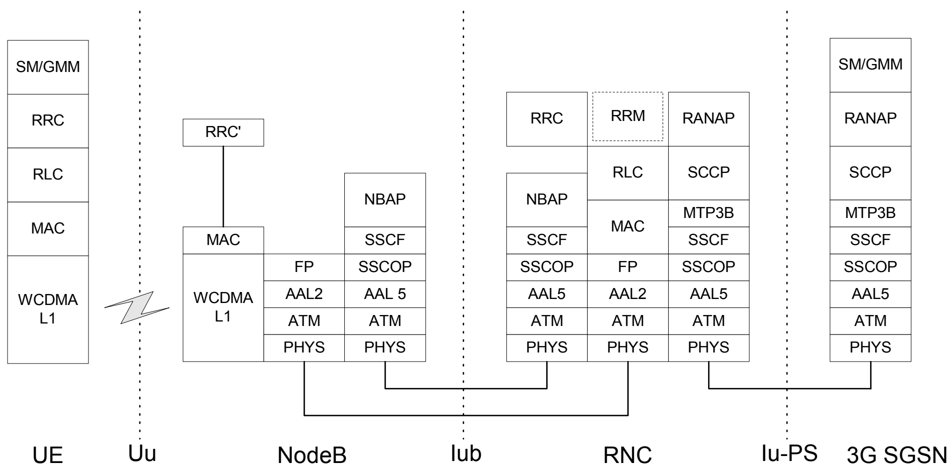

Figure 3.2 Control plane protocols

Uu protocols

Session Management (SM) protocol is used to activate, modify and deactivate packet data sessions. Its main function is to support Packet Data Protocol (PDP) context handling of the user equipment [3]. SM contains procedures for activating, modifying and deactivating PDP contexts. PDP context activation, PDP context modification and PDP context deactivation are described in chapter 3.3.

The main function of the *GPRS Mobility Management* (GMM) protocol is to support mobility of the user terminals, for example, by location updates and authentication. One of the main procedures of the GMM is Attach procedure. This procedure is used to inform the network of the users location and establishing GMM context to SGSN. The Attach procedure is described in chapter 3.3.

The *Radio Resource Control* (RRC) is a signalling protocol between UE and UTRAN. The RRC protocol can carry all higher layer (GMM/SM) signalling as a payload of a message. It is used to setup and release the signalling connection between UE and UTRAN. The signalling connection setup must be performed before any higher layer signalling can be transported. It is also used for setting up and releasing radio bearers for user plane traffic. Radio bearer is presented in chapter 3.3. The subset of RRC in NodeB (RRC') broadcasts cell-specific information or paging messages from network to UE.

The *Radio Link Control* (RLC) protocol is used both in control and user plane. It provides a logical link control over air interface. Main functions of the RLC are for example, segmentation/reassembly, concatenation, error detection, duplicate detection, retransmission and ciphering. RLC can operate in three modes: transparent, unacknowledged and acknowledged. Transparent mode means that the RLC does not add any protocol information to SDUs (Service Data Unit). In unacknowledged mode RLC transmits SDUs without guaranteeing delivery to the peer entity [9]. Acknowledged mode means that the RLC guarantees the transmission of SDUs to peer entity.

The *Medium Access Control* (MAC) is also used both in control and in user plane. It is responsible for controlling the communications over WCDMA transport channels provided by physical layer [9]. MAC performs scheduling of radio bearers (or logical channels). This is depicted more closely in chapter 4.8.1.

Iub protocols

The radio specific information is told to base station with *NodeB Application Part* (NBAP) protocol. NBAP is used to setup and release radio channels and then bind established resources to radio interface. NBAP is also used for configuring cells of base stations.

Frame Protocol (FP) is used to transfer data streams over AAL 2. Payload of FP usually contains one or more MAC PDU's (Protocol Data Units) and FP header contains synchronisation information between NodeB and RNC.

Iu protocols

Radio Access Network Application Part (RANAP) controls the resources in the Iu interface. In PS domain it is between RNC and 3G SGSN and in CS domain between RNC and 3G MSC. It provides Radio Access Bearer (RAB) management, by providing means for CN to control the establishment, modification and release of RABs between UE and CN. It also participates user mobility, by transferring a RAB to a new RNS when a user moves from the area of the serving RNS to another. [9] This service is called SRNS relocation. RANAP also transfers higher layer messages between UE and 3G SGSN (in PS domain), as the RRC transfers UE messages to RNC, RANAP transfers them from RNC to 3G SGSN.

Message Transfer Part Level 3 Broadband (MTP3B) and *Signalling Connection Control Part* (SCCP) belong to the Common Channel Signalling System 7 (SS7) protocol stack inherited from the GSM system. The purpose of SCCP is to provide connection-oriented and connectionless services. MTP3B is responsible for message routing.

<u>Others</u>

Radio Resource Management (RRM) is actually not a protocol, it is rather a collection of algorithms needed for establishing and maintaining a good radio path. The RRM contains algorithms for handovers, power control, admission control, load control and packet scheduling. The packet scheduler is discussed more detailed in chapter 4.8.1 because it is an essential part of the Pilot system.

Asynchronous Transfer Mode (ATM) transmission is based on fixed length data frames, cells. Length of the cell is 53 bytes, from which the 5 first bytes are for the header and following 48 bytes are for payload.

ATM Adaptation Layer (AAL) 2 and *AAL 5* are adaptation layers to the ATM. The main task of those layers is to split data to the ATM cells. Characteristics of the AAL 2 is a short payload, which reduces the delay of packetisation. Therefore it is suitable for carrying for example speech packets.

Service-Specific Connection-Oriented Protocol (SSCOP) provides mechanisms for the establishment, release and monitoring of signalling information exchanged between peer entities. It provides assured data delivery between connection endpoints. *Service-Specific Coordination Function* (SSCF) maps the SSCOP to the upper layer.

3.1.2 Transport Network Control Plane

Transport Network Control Plane exists only in Iu-CS, Iur and Iub interfaces. Because Iur and CS have not been implemented in Pilot system those are out of the scope of the thesis, so only Transport Network Control Plane in Iub is described here. The Iu-PS uses preconfigured AAL 5 connections, ALCAP is not needed. Transport Network Control Plane is present only when transport technology needs controlling.

On the Iub interface ALCAP consists of *Q.2630.2* and *Q.2150.2* protocols, see Figure 3.3. Q.2630.2 is used for setting up and releasing AAL 2 connections of the Iub. Q.2150.2 is a signalling transport converter between Q.2630.2 and SSCF.

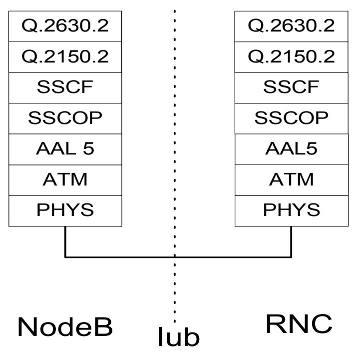

Figure 3.3 Transport Network Control Plane in Iub

3.1.3 User Plane

User plane protocols are used for transferring application level data, like IP packets of a video conversation. These protocols implement the actual radio access bearer service. User plane protocols are depicted in Figure 3.4.

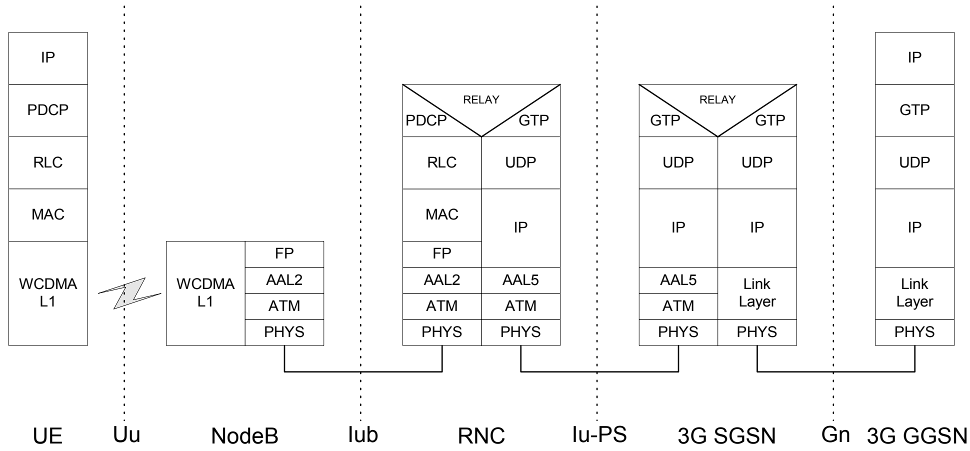

Figure 3.4 User plane protocols

Uu protocols

The main task of the *Packet Data Convergence Protocol* (PDCP) is to compress the redundant protocol control information. PDCP and header compression are described more detailed in chapter 4.6.

RLC and MAC were depicted more closely in chapter 3.1.1.

Iu protocols

GPRS Tunneling Protocol for packet data user plane (GTP-U) provides transport services for IP traffic in Iu-PS and Gn interfaces. It also does Differentiated Service marking for uplink packets. Differentiated Service is depicted in chapter 4.8.2.

Internet Protocol (IP) is a network layer protocol. It provides transmission of the datagrams from source to destination. It can be either version 4 or 6. *User Datagram Protocol* (UDP) provides transmitting datagrams on top of the IP protocol. It provides unreliable delivery of the sent datagrams. Real-time traffic is typically carried over UDP.

Others

FP, AAL 2, AAL5 and ATM were depicted in chapter 3.1.1.

3.2 UMTS Bearer Service

A typical end-user is not interested in how a certain service is technically provided, rather the user observes quality of the service. Quality usually means for example accessibility to the service and the delay of the connection. Bit rate is as well a significant factor when measuring the quality of a service.

UMTS has to support a wide range of applications having different QoS requirements. To meet these requirements a bearer concept was introduced. The bearer forms a logical connection through UMTS network with a certain QoS profile. The layered architecture of UMTS Bearer Service is shown in Figure 3.5 [2]. In bearer service architecture every layer hides the details of layers below. The QoS requirements are handled differently in various parts of the network. For example the requirements and parameters are different in air interface than in Iu interface.

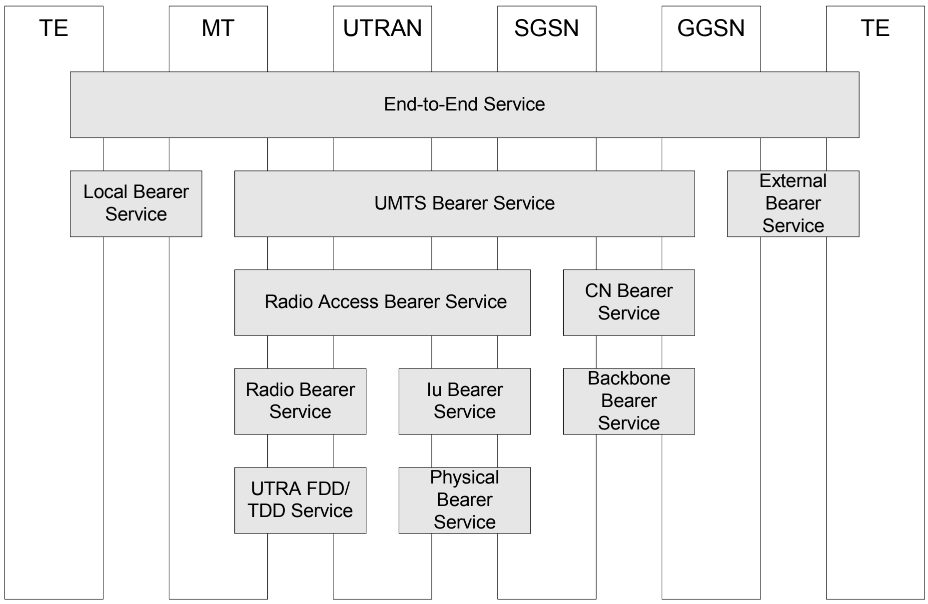

Figure 3.5 UMTS QoS architecture

Network services are considered to be end-to-end, between Terminal Equipment's (TE), with a certain QoS profile. The End-to-End Service uses services from Local Bearer Service, UMTS Bearer Service and External Bearer Service. The UMTS Bearer Service in turn uses Radio Access Bearer Service and CN Bearer Service to fulfill its QoS requirements. All the Bearer Services have a set of attributes representing their capabilities. The attributes of the UMTS Bearer Service are described in Table 1. [2].

Traffic Class	Conversational	Streaming	Interactive	Background
Maximum bit rate	X	X	X	X
Delivery order	X	X	X	X
Maximum SDU size	X	X	X	X
SDU format information	X	X		
SDU error ratio	X	X	X	X
Residual bit error ratio	X	X	X	X
Delivery of erroneous SDUs	X	X	X	X
Transfer delay	X	X		
Guaranteed bit rate	X	X		
Traffic handling priority			X	
Allocation/retention priority	X	X	X	X

Table 1 UMTS Bearer service parameters

Here is a short description of each attribute [2]:

- Traffic class indicates for what kind of application the bearer is suitable. Traffic class is discussed in a more detailed way in chapter 3.2.1.
- Maximum bit rate, maximum number of bits delivered (kbits/s).
- Delivery order indicates whether the UMTS bearer shall provide in-sequence SDU delivery or not.
- Maximum SDU size indicates the maximum allowed SDU size. It is used for admission control and policing.
- SDU format information, list of possible exact sizes of SDUs. UTRAN needs SDU size information to be able to operate in transparent RLC

protocol mode, which is beneficial to spectral efficiency and delay when RLC re-transmission is not used. Thus, if the application can specify SDU sizes, the bearer is less expensive.

- SDU error ratio indicates the fraction of lost or erroneous SDUs.
- Residual bit error ratio indicates the undetected bit error ratio in the delivered SDUs. If no error detection is requested, Residual bit error ratio indicates the bit error ratio in the delivered SDUs.
- Delivery of erroneous SDUs indicates whether SDUs detected as erroneous shall be delivered or discarded.
- Transfer Delay indicates the maximum delay accepted for sending the message to receiver.
- Guaranteed bit rate, guaranteed number of bits delivered (kbits/s). Guaranteed bit rate may be used to facilitate admission control based on available resources.
- Traffic Handling Priority (THP) specifies the relative importance for handling of all SDUs belonging to the UMTS bearer compared to the SDUs of other bearers. THP is used only in case of interactive traffic class.
- Allocation/retention priority specifies the relative importance compared to other UMTS bearers for allocation and retention of the UMTS bearer.

3.2.1 Traffic Classes

The UMTS QoS is based on the classification of different kind of traffic flows. The classification has to be robust and capable of providing reasonable QoS resolution [2]. Four different QoS classes has been defined:

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

The main distinguishing factor between these QoS classes is how delay sensitive the traffic is: Conversational class is meant for traffic which is very delay sensitive while background class is the most delay insensitive traffic class [2].

The conversational class is intended to be used to carry real-time, delay sensitive traffic flows. The best known application of this class is speech over circuit-switched bearers. Video telephony belongs to this category too. Maximum transfer delay is given by the human perception and hence limit for acceptable transfer delay is very strict, as a failure to provide it will result in unacceptable lack of quality.

Streaming class is also intended to carry real-time traffic flow, difference to the conversational class is that the traffic is not so delay sensitive. Streaming applications, like television channel downloading, are very asymmetric and therefore typically withstand more delay than more symmetric conversational services. This also means that they tolerate more jitter in transmission.

Interactive scheme applies when the end-user is on line requesting data from a server, for example web browsing and database retrieval. Because end-user is expecting the message (response) within a certain time, the round trip delay time is therefore one of the key attributes.

Background traffic is characterised by that the destination is not expecting the data within a certain time. Examples of the background applications are emails, Short Message Service, download of databases and reception of measurement records [2].

3.2.2 Methods for Providing Requested QoS in UTRAN

Radio interface has limited bandwidth, and increasing that is very difficult. Thus, it is necessary to use it efficiently. Some methods are designed to be used especially in error prone air interface. The RRM in RNC is mainly responsible for these methods. RRM contains various algorithms, which aim to stabilise the radio path enabling it to fulfill the QoS criteria set by the service using the radio path [9]. Those algorithms are, for example power control, load control, handover control, admission control and packet scheduling.

When real-time multimedia traffic is transferred over a limited bandwidth radiolink, it is necessary to reduce the overhead that results from transmitting the headers [13]. RRM can reduce the overhead by choosing some IP header compression algorithm (supported by the UE). Two different algoritms have been specified by 3GPP, RFC 2507 (Release 99) and RFC 3095 (Release 4). Latter is also known as RObust Header Compression (ROHC).

The user plane of the UMTS core network is based on IP, therefore it is necessary to have QoS provision for IP traffic. Differentiated Services method is the IP level QoS provision mechanism 3GPP has specified in UMTS Release 99.

Packet scheduling, header compression (ROHC) and Differentiated Service Marking are implemented to the Pilot system and therefore those are described in a more detailed way in chapters 4.8.1, 4.6 and 4.8.2 respectively.

3.2.3 PDP Contexts and Traffic Flow Templates

The PDP context is a term used in packet data sessions. It is a logical connection through UMTS network with a certain QoS profile, see Figure 3.6.

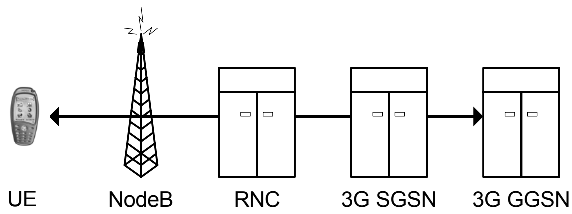

Figure 3.6 PDP connection

The PDP context contains all parameters describing the packet data connection [9]. One of the parameters is PDP address, it can be IPv4, IPv6 or Point to Point Protocol (PPP) address. GGSN allocates IP address to the UE. PDP address is used by UE application. Other attributes are for example QoS Requested and QoS Negotiated by the network and Traffic Flow Template in case of secondary PDP context. QoS attributes of PDP context maps directly to UMTS bearer parameters (see Table 1). It is possible to allocate multiple primary PDP contexts, each having own PDP address, to the UE (see Figure 3.8). The user can have multiple primary PDP contexts to same Access Point Name (APN). APN identifies packet data network to which the user wants to connect. It is also possible to activate several contexts to the same PDP address, first established is called primary and the following are called secondary PDP contexts.

Secondary PDP context uses the same PDP address than primary PDP context, but it has different QoS profile. It is linked to the primary PDP context with Linked TI field (see Figure 3.8). Secondary PDP context can be initiated only when there is previously activated primary PDP context.

Primary PDP context is used for transmitting data that does not fit any of the packet filters. Packet filter is described in the next paragraph.

Traffic Flow Template (TFT) consists of from one to eight packet filters, which are used to route the data packets to the right PDP contexts, see Figure 3.8. This information element is signalled to GGSN with the PDP context activation or modification procedure. It is also signalled to UE in case the UE terminates context activation or modification. Traffic Flow Template is used in case of secondary PDP context. A packet filter has also an evaluation precedence index that indicates the order of the evaluation in case of multiple filters. The UE (or network) needs to attach unambiguous TFT all except one PDP context to enable correct routing of the data.

When PDP context has been created and it is time to send application level data, such as VoIP packets, the right radio bearer is determined first. This is done by evaluating the filters in the traffic flow template in order of the evaluation precedence. If IP packet matches to the filter, then it can be routed to that bearer. Filtering can be based on various fields in the packet header, for example, IP addresses, next protocol, UDP/Transmission Control Protocol (TCP) port numbers and so on. If a filter is not found (or matching) packet is sent to PDP context that does not have filter (normally primary PDP context). Figure 3.7. describes the evaluation of the filters.

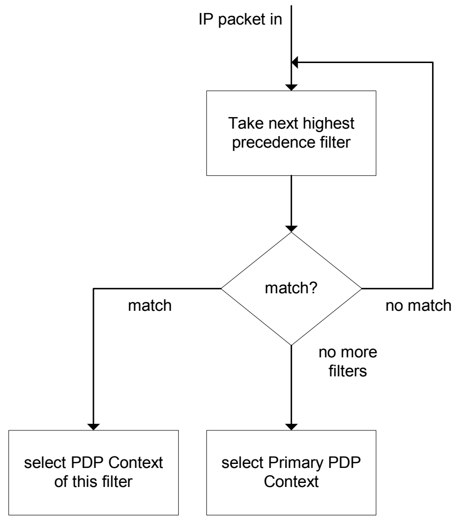

Figure 3.7 Evaluation of packet filters

Linked TI indicates the TI value assigned to primary PDP contexts for this PDP address and APN [1], see Figure 3.8. It is used when secondary PDP context is activated to identify the primary PDP context.

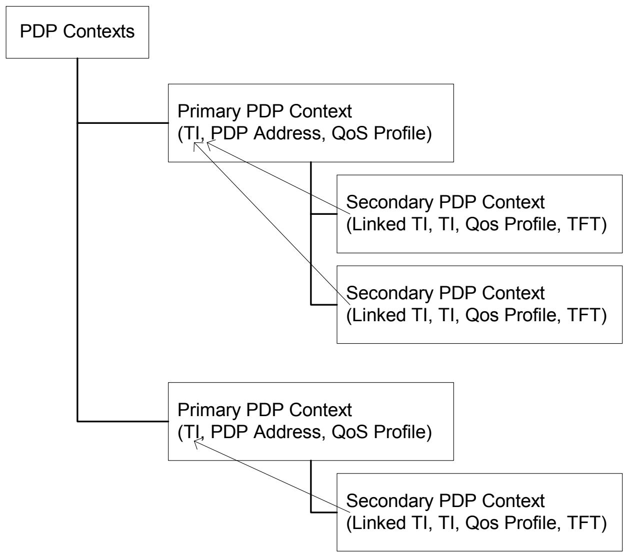

Figure 3.8 Relation between different concepts

3.3 Signalling Procedures

In this chapter elementary signalling procedures related to packet data are described. Main procedures are *Attach*, *PDP context activation*, *PDP context modification* and *PDP context deactivation*. There are a number of other important procedures too, such as cell update, handover and routing update. They are not considered because there were only one NodeB in Pilot system and thus those were not needed. All following procedures are assumed to be successful.

3.3.1 Attach

To be able to send or receive data UE must first register to an SGSN. This registration is performed by Attach procedure. The procedure is done automatically when UE is powered on. The procedure starts with the RRC connection setup, that is a signalling connection between UE and RNC. After the connection has been setup, UE sends ATTACH REQUEST message to the network, informing about its identity, type of attach and its capabilities. These capabilities are, for example, supported header compression algorithms. When network receives the request from the UE it checks if the user is authorised to use network services. If authentication is successful, network assigns a packet temporary mobile subsciber identity (P-TMSI) to the user. That is the identifier used to identify UE in subsequent transactions (from other UEs). The required signalling between different network entities in Attach procedure is described in Figure 3.9.

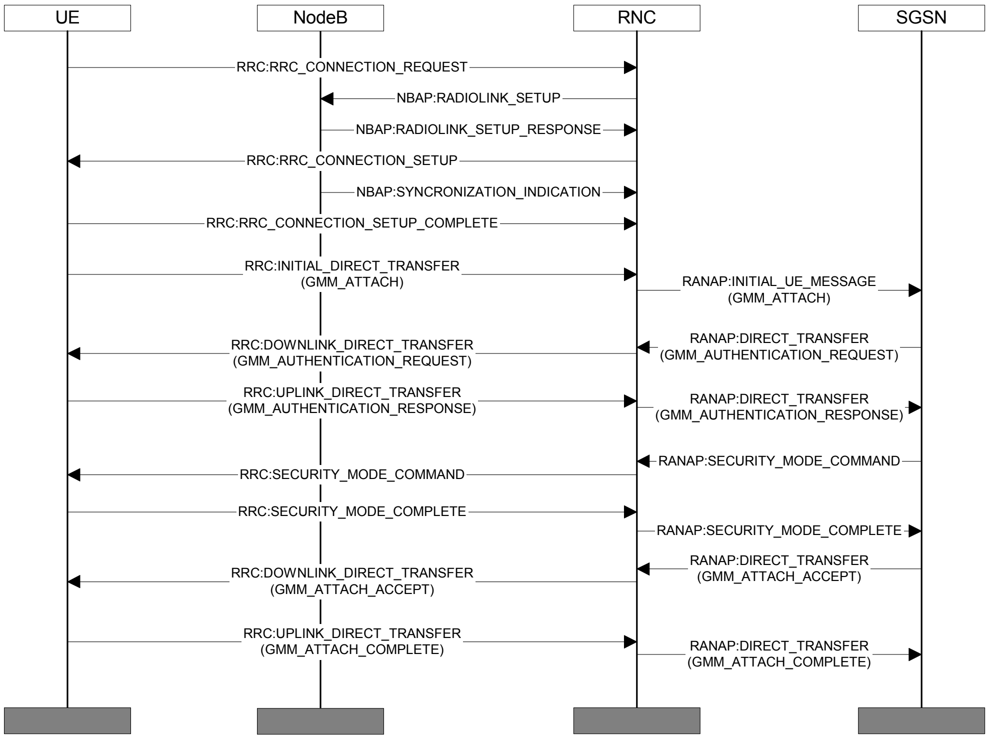

Figure 3.9 Attach procedure

3.3.2 Primary PDP Context Activation

If the UE wants to exchange data packets with the external packet data network (PDN) after a successful Attach procedure it needs to have a PDP address used in that PDN. The PDP context activation procedure is shown in Figure 3.10. It starts when the UE sends an ACTIVATE PDP CONTEXT REQUEST message to the network. The message contains, for example, PDP type (IPv4 or IPv6) and requested QoS. If IP address field is left empty, then dynamic addressing is used and the GGSN allocates an address for UE. After SGSN has received the message, Bearer Setup procedures are started, Radio Bearer (RB) is setup between UE and RNC and Iu Bearer is setup between RNC and SGSN. These two bearers compose Radio Access Bearer (RAB) with the requested QoS parameters, see Figure 3.5. SGSN informs GGSN about the new context by sending CREATE PDP CONTEXT REQUEST message. GGSN uses information in this message to create a new entry for its PDP context table and it responds to SGSN with the CREATE PDP CONTEXT RESPONSE message. Message contains IP address and QoS negotiated. SGSN builds ACTIVATE PDP CONTEXT ACCEPT message according to the message it got from GGSN and sends it to UE. RNC, GGSN and SGSN may downgrade the requested QoS profile depending on their capabilities. The subcriber profiles from HLR may also set limitations to QoS profile. The message contains negotiated QoS parameters. If the UE accepts negotiated parameters, the packet data exchange is enabled or otherwise the context is deactivated.

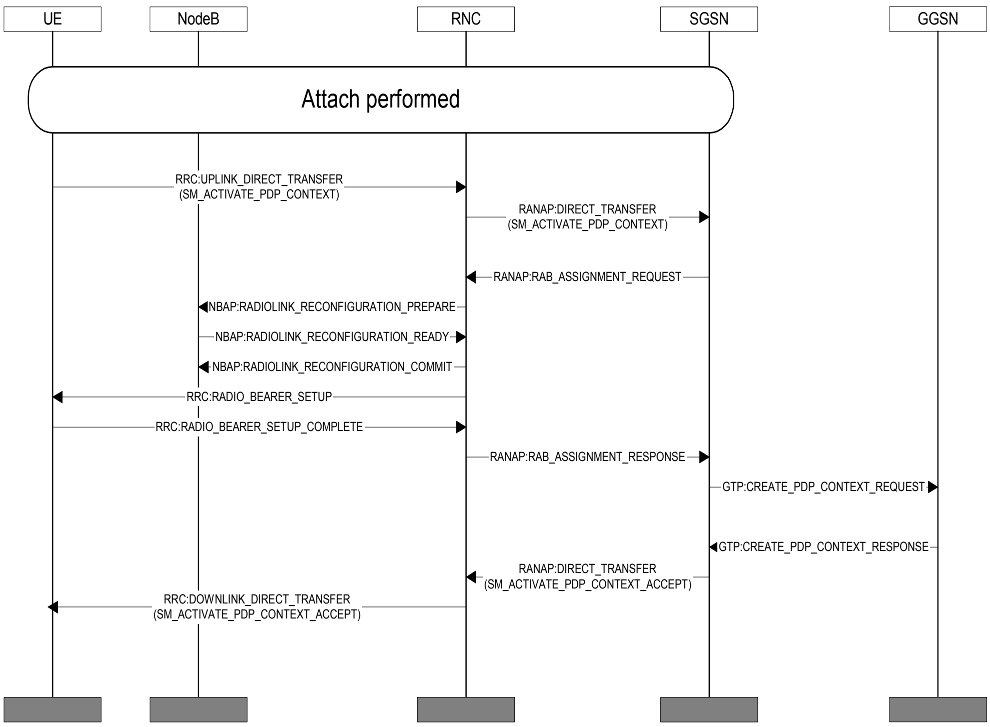

Figure 3.10 Primary PDP context activation procedure

3.3.3 Secondary PDP Context Activation

This procedure is used when a user wants to establish an additional PDP context with the QoS profile different than in PDP context previously established. Procedure starts when UE sends SM ACTIVATE SECONDARY PDP CONTEXT REQUEST message containing the Requested QoS, Linked TI and TFT, see Figure 3.11. If TFT is present, it shall be sent transparently through the SGSN to the GGSN to enable packet classification and policing for downlink transfer [3]. SGSN validates message by verifying that Linked TI is one of the active PDP contexts. Network starts the Bearer Setup procedures with the QoS negotiated parameters. After that it replies with the SM ACTIVATE SECONDARY PDP CONTEXT ACCEPT message. If the QoS parameters differ from what the UE requested, it shall either accept the negotiated QoS profile or start PDP context deactivation procedure [3].

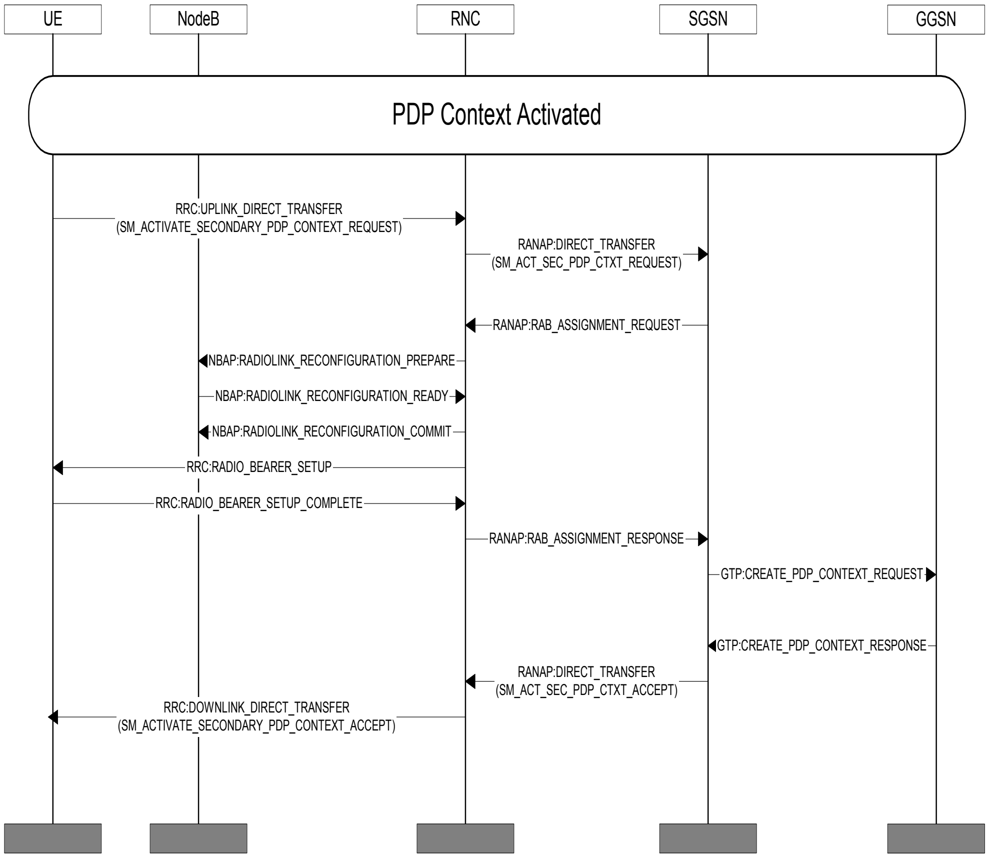

Figure 3.11 Secondary PDP context activation procedure

3.3.4 PDP Context Modification

If for example the radio network cannot anymore provide the QoS negotiated during the PDP context activation, there is a need to re-negotiate the QoS parameters; otherwise the radio network may become overloaded.

The PDP context modification procedure is used to modify QoS parameters of the active session. The modification procedure may start from UE, RNC, SGSN or GGSN. RNC may start the modification procedure by sending RANAP message RAB MODIFY REQUEST to the SGSN, see Figure 3.12, case 1. GGSN and RNC initiated modification procedures are described in Figure 3.12.

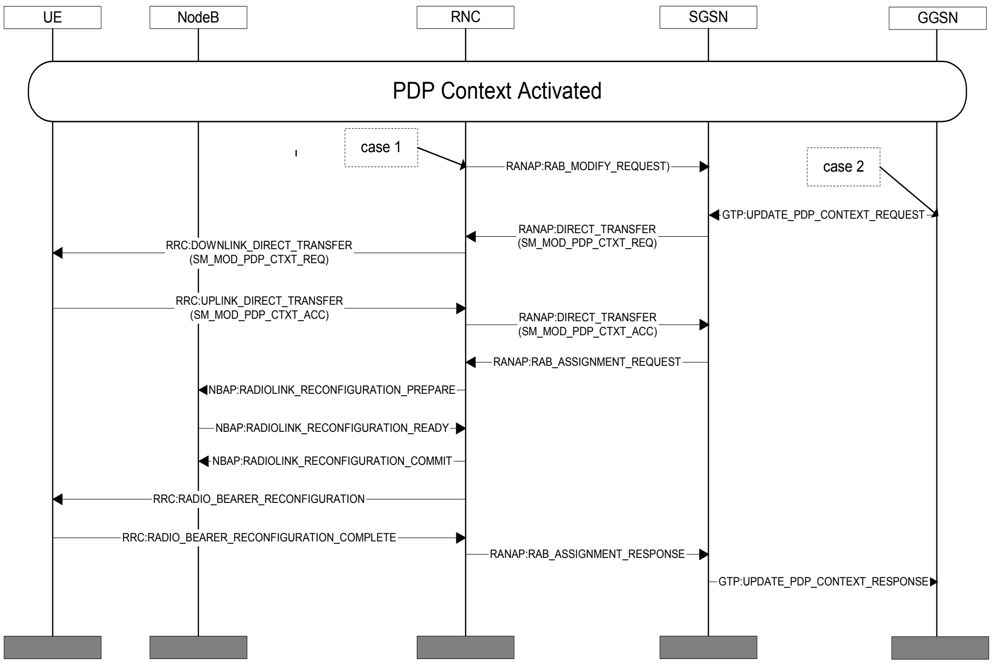

Figure 3.12 PDP context modification procedure

GGSN starts the procedure by sending UPDATE PDP CONTEXT REQUEST message to SGSN, containing QoS requested and optionally the PDP address, case 2 in Figure 3.12. The SGSN may restrict the QoS profile for example, due to the current load in network. Then SGSN sends MODIFY PDP CONTEXT REQUEST message to the RNC, containing QoS negotiated and the PDP address. Message is delivered in RANAP DIRECT TRANSFER message to RNC. From RNC to UE the message is carried in RRC DOWNLINK DIRECT TRANSFER. UE responds with SM MODIFY PDP CONTEXT ACCEPT if it accepts the negotiated QoS parameters. When RNC gets the message, it forwards it to SGSN by capsulating it in to RANAP DIRECT TRANSFER message. SGSN starts the bearer modification procedures with negotiated QoS parameters by sending RANAP RAB ASSIGNMENT REQUEST to RNC. After the bearers have been reconfigured, RNC responds with RANAP RAB ASSIGNMENT RESPONSE message. SGSN then returns UPDATE PDP CONTEXT RESPONSE to GGSN, with QoS negotiated.

3.3.5 PDP Context Deactivation

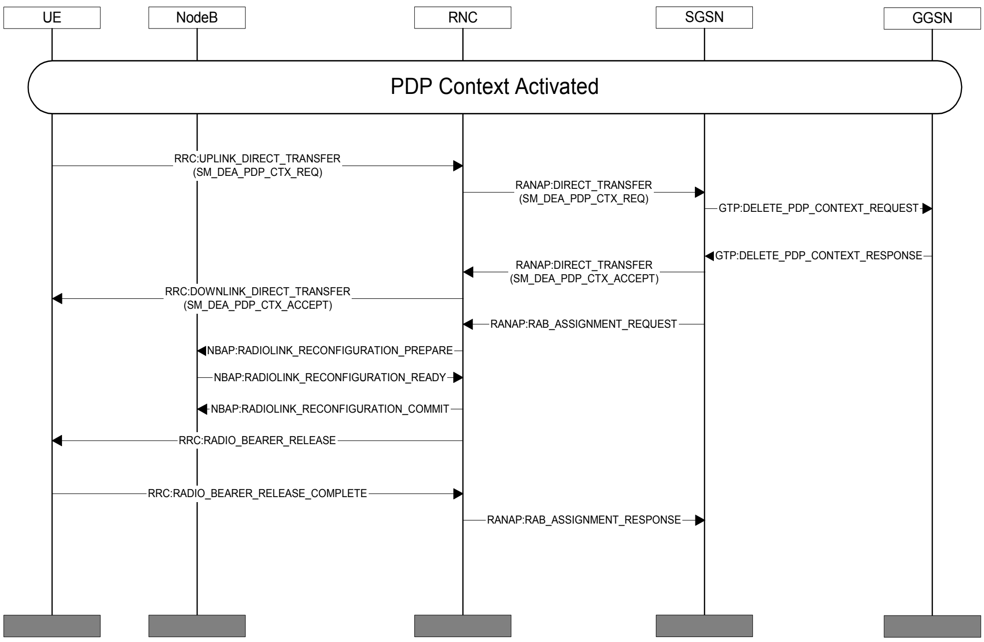

Figure 3.13 PDP context deactivation

PDP context deactivation is described in Figure 3.13. UE sends DEACTIVATE PDP CONTEXT REQUEST message to the SGSN. Message contains optional Teardown information element. Teardown is used to deactivate all the PDP contexts associated to this PDP address. SGSN sends DELETE PDP CONTEXT REQUEST to GGSN. GGSN deletes the context(s) and responds with DELETE PDP CONTEXT RESPONSE message. When SGSN receives the message, it sends

DEACTIVATE PDP CONTEXT RESPONSE to UE. It also starts bearer release procedure by sending RANAP RAB ASSIGNMENT REQUEST to the RNC. RNC releases bearers and replies with RANAP RAB ASSIGNMENT RESPONSE message.

4. UMTS PILOT SYSTEM

This chapter describes the implementation part of the thesis. First, some general information of the project is given in chapter 4.1. Then, the system architecture and the protocol architecture is depicted, in chapters 4.2 and 4.3 respectively. After that in chapter 4.4 there is a brief look on different tools and languages used in system development. Then the principles of the system integration are described in chapter 4.5. Chapter 4.6 describes the implementation of the PDCP. Interface to WCDMA radio parts is depicted is chapter 4.7. QoS issues of the system are described in chapter 4.8. Chapter 4.9 describes the usage of the system. Chapter 4.10 depicts the validation of the protocols.

4.1 General

The UMTS Pilot system has been developed in Nokia Research Center, Helsinki. The work started in November 1999 and is still ongoing. The purpose of the project was to build a platform for demonstrating 3GPP WCDMA radio and the behaviour and functionality of the latest 3GPP protocols. The aim was also to develop platform for testing behaviour of different IP based applications. The system includes UE, UTRAN and Core Network emulators. The protocols can be connected to real 3GPP WCDMA radio, or alternatively radio interface can be replaced with UDP socket. The radio interface was also developed in NRC Helsinki. The system can also be connected to Nokia's 3G SGSN product.

The Pilot system has been delivered to several teleoperators around the world. It has also been demonstrated in several conferencies and exhibitions, such as Cebit in Germany 2001.

The system enables setting up multiple PDP contexts: primary and secondary, for IPv4 and IPv6 based applications. It is possible to connect all kinds of IP based applications, also commercial, to the system. In demonstrations, for example, streaming, video conferencing and web browsing were shown.

Because the system was designed mainly for demonstration purposes there are some limitations compared to the commercial systems, for example, there is only one NodeB and no Iur interface. Also only one UE can be connected to the system at the time. Iub interface is "reduced" version, only NBAP protocol is present, lower layers are replaced by socket. Maximum data rate of the air interface (Uu) is limited to 384 kbits/s.

Protocols are implemented using both SDL (Specification and Description Language) and CVOPS (C-based Virtual Operating System). In chapter 4.3 this is shown in more detailed way. Almost all the protocol skeletons have been received from other NRC projects. Skeleton means that the protocols were not complete implementations, for example messages and primitives were not fully filled resulting that some required information elements and parameters were missing. Filling the messages (and primitives) with correct parameters took a quite a long time. PDCP and RRM (not exactly protocol) were output of this project. Also interface to WCDMA radio parts (layer 1) was implemented in the project. Thus, the main focus has been in integrating the protocols working together. Because the protocols were implemented in different projects using different tools, integration process became more complicated. A special tool had been developed in NRC for integrating SDL and CVOPS parts together; tool is called SDT and CVOPS Integration Utility (SCIU). Actually, rather than a tool it consists of modified kernels of CVOPS and SDT (SDL Tool). SCIU is discussed in more detailed way in chapter 4.4.4. The testing of the system took a large amount of time although SDT offers good tracing facilities. In the 3GPP WCDMA radio, messages are exchanged between protocol stack and radio parts in exact frequency of 10 milliseconds and that accuracy brought very challenging requirements to the real-time protocols. It meant that there was a need to optimise the performance of the protocol implementations.

The project was divided into three phases and in every phase additional features were added to the system. This division was done because the 3GPP protocol specification versions evolved continuously and therefore also protocol implementations needed to be updated. The author was in his part responsible for system integration and implementation of the PDCP protocol and also in integrating the system with the WCDMA radio parts. System integration and implementation of the required features to the protocols were the most time-consuming tasks. Both of them took several months in every phase. Testing the system (also with the WCDMA radio parts) was a continuous task through the project.

4.2 Pilot System Architecture

The Pilot system consists of three PC's with Linux operating system. UE and UTRAN machines are card PC's with the VME (VERSAmodule Eurocard) interface. VME interface is used in communication with baseband. Baseband is a unit that handles layer 1 processing of the data, channel coding, interleaving, rate matching etc. Radio Frequency (RF) sends the processed data to the air and receives the data from the air. Baseband and RF compose the Layer 1.

UTRAN PC has an ATM card for enabling ATM connection towards SGSN. Application PCs and UE Linux PC are connected to hub. Figure 4.1 presents the architecture with real WCDMA radio parts and RF parts.

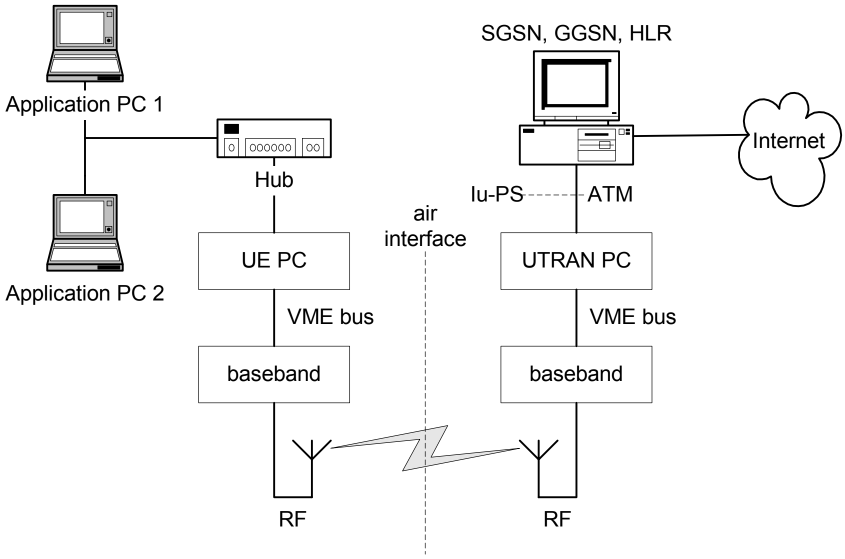

Figure 4.1 Pilot system with 3G WCDMA radio

It was possible to replace the WCDMA radio and RF with WCDMA radio emulator process inside UE and UTRAN machines, then the traffic in radio interface goes via UDP socket, as shown in Figure 4.2. The real WCDMA radio brought more complexity to the system. The reason for having WCDMA radio emulator was that then it was possible to test the functionality of the protocols before connecting them to the real WCDMA radio.

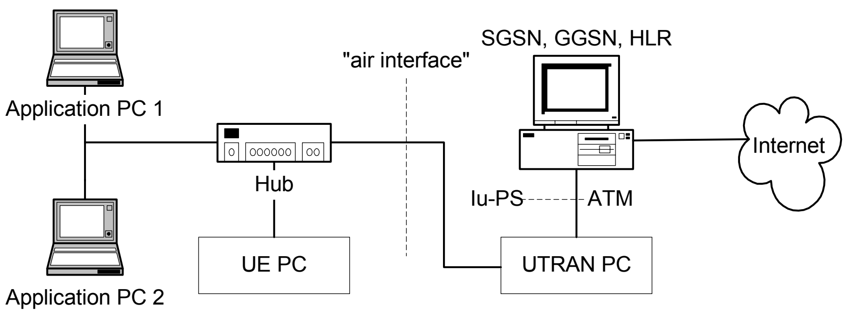

Figure 4.2 Pilot system without 3G WCDMA radio

4.3 Protocol Stack Architecture

This chapter presents the protocol architecture of the system. Main focus is in UE and in UTRAN (RNC and NodeB) parts and those are shown detailed below. In APPENDIX 1 the full protocol stack used in Pilot system is shown. It should be noted that SCIU is not actually a "block", but it is shown in the following pictures to clarify the integration of the protocols.

UE:

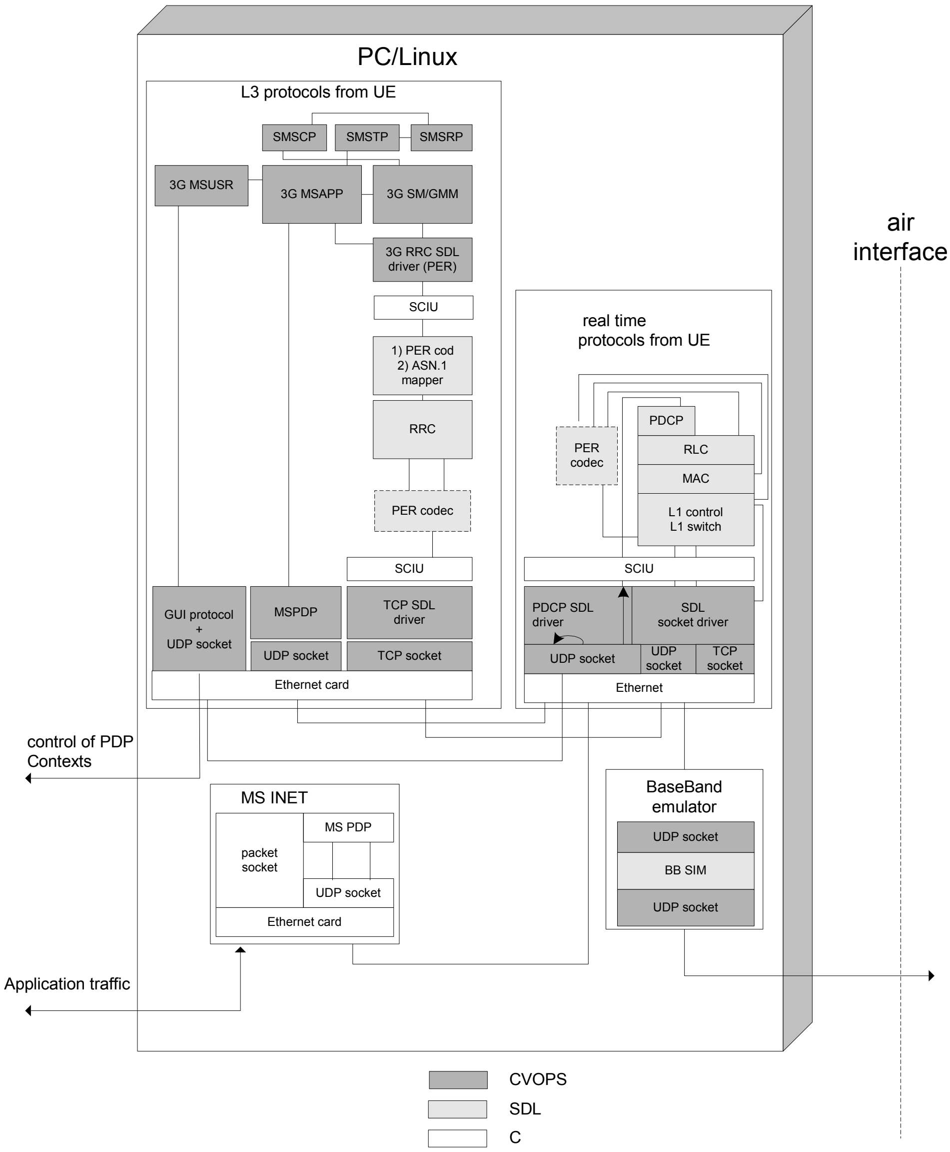

Figure 4.3 UE protocols

The UE protocol stack has been divided into two processes, L3 (Layer 3) process and real-time process. All L3 protocols are run in L3 process and all real-time protocols in real-time process. There are also two additional processes in the picture, Baseband emulator and MSINET. MSINET is basically needed for capturing and routing of the IP packets between UE Linux PC and Application PC. MSINET is discussed more detailed in chapter 4.9. Baseband emulator is used for replacing the functionality of the real 3G WCDMA radio when UDP socket is used in the radio interface.

The real-time process contains PDCP, RLC and MAC protocols. There are also some additional blocks, such as Packet Encoding Rule (PER) codec (described in chapter 4.5) and L1 control/switch. All above-mentioned blocks are coded using SDL. There are some blocks handling sockets, those are coded using CVOPS. Also SCIU drivers are needed, such as pdcp_sdt_driver and tcp_socket_driver. The general description of the SCIU drivers is given in chapter 4.4.4. Most of the protocols in L3 process have been implemented using CVOPS, such as GMM/SM. RRC and PER codec have been implemented with SDL.

UTRAN:

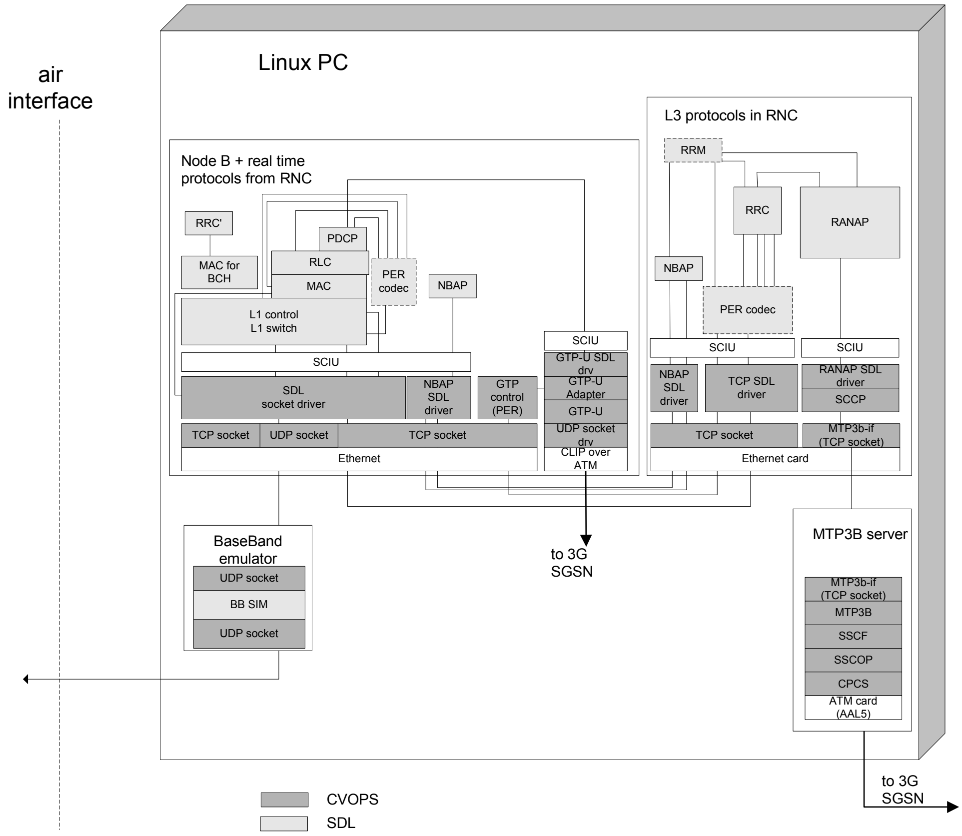

Figure 4.4 UTRAN protocols

UTRAN side has been divided into four processes, Baseband emulator, MTP3B server and NodeB with the real-time protocols from RNC and L3 protocols of the RNC. Functionality of the Baseband emulator is similar to the UE side. MTP3B server handles the signalling connections needed by RANAP and SCCP towards 3G SGSN. It is coded using CVOPS.

L3 process contains RANAP, RRM, NBAP and RRC protocols and PER codec block, those are SDL implementations. There are also SCIU drivers, for example nbap_sdt_driver and ranap_sdt_driver.

NodeB with real-time protocols from RNC contains NBAP, RRC', MAC, RLC and PDCP protocols and PER codec. Those were implemented with SDL. SCIU drivers and GTP-U is implemented with CVOPS.

4.4 Development Environment

In this chapter the development environment is described. First SDL and SDT are described. Then CVOPS and also brief description about SCIU and ASN.1 (Abstract Syntax Notation One).

4.4.1 SDL

SDL is a standard language for specifying and describing systems [7]. It has been developed and standardised by ITU-T (International Telecommunication Union, Telecommunication Sector) in the recommendation Z.100. SDL has designed especially for telecommunication industry and it has been found out to be suitable even for product level implementation [14].

SDL has a graphical representation (SDL/GR, SDL Graphical Representation) in addition to the textual representation (SDL/PR, SDL Phrase Representation). The graphical representation has made the language user-friendly and easy to use [7].

The SDL describes system structure in a hierarchical manner, see Figure 4.5 [15]. The highest level is a system level. The system level is composed of one or several blocks (Bl1 and Bl2) connected by channels together or to environment.

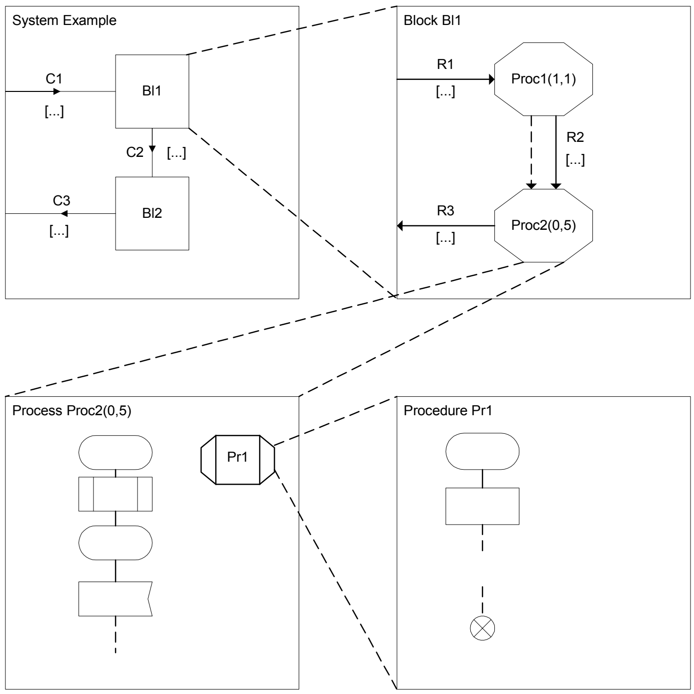

Figure 4.5 Hierarchical structure of SDL

C1, C2 and C3 are channels. A channel declaration includes a list of signals which can be sent via the channel. A block contains other blocks or processes (Proc1, Proc2). Those are connected together with signal routes (R1, R2 and R3). A signal route has a list of signals it can delivere. Dashed line with arrow indicates that the Proc1 can dynamically create process instances of Proc2. Process contains Extended Finite State Machine (EFSM) and it defines the behaviour of the system. A process may call procedures (Pr1). A procedure in SDL is similar to a procedure in other programming languages.

4.4.2 SDT

SDT is a set of tools for developing the real-time systems in SDL. It offers utilities for all phases in the software development process. SDT is based on the object-oriented design language SDL, and the trace language Message Sequence Chart (MSC). Both of these are formal, visual and standardised languages [15]. The main tool of the SDT is Organizer, it manages all the

SDL designs. Other tools are SDL editor, analysator, code generator and simulator. SDL editor offers a graphical user interface for handling of SDL diagrams. It also does real-time syntax checking for SDL designs. Analysation is carried out before the code generation, it checks syntactics and semantics of the design. The analyser also converts Graphical Representation to Phrase Representation. The code generator compiles the SDL representation to C code. There are several code generators for different purposes, for example Cbasic, Cadvanced and Cmicro. Cmicro is designed to be used in embedded systems. The generated code is much more compact than using the other code generators. Cbasic is mainly designed for simulation purposes and Cadvanced for building any kind of application. With help of the simulator it is possible to check how the system behaves when running it. It is useful for detecting run-time errors in the design. Using simulators MSC trace it is possible to see the signalling in a graphical format.

SDT enables the designer to embed external code to the system. That code could be written in C or C++. For example, PDU encoding and decoding functions are usually implemented by using (tool generated) C functions. The code SDT produces can be compiled to various platforms, such as Linux, Windows and VxWorks (a real-time operating system).

4.4.3 CVOPS

CVOPS is a tool and run-time environment for implementing communication protocols. It has been developed by the Technical Research Centre of Finland (VTT) [11]. It has been used for long time in the telecommunication industry.

CVOPS offers mechanisms for protocol development, such as scheduling, message passing, timers and memory handling. It also includes a special language for implementing protocol logic, Extended Finite State Automaton (EFSA).

A CVOPS system may consists of several protocols, which may all include multiple instances (connections). One instance from a protocol implemented with CVOPS is called virtual task (vtask). Protocol stack is implemented by connecting these vtasks together. CVOPS handles communication between vtasks. A vtask sends messages to other vtasks via its interfaces. Both sides of an interface have their own parameter functions, *putParameters* and *getParameters*. PutParameter is called by CVOPS when the message is sent from a vtask and getParameter is called when the message is received by a vtask. This is described in Figure 4.6 [11].

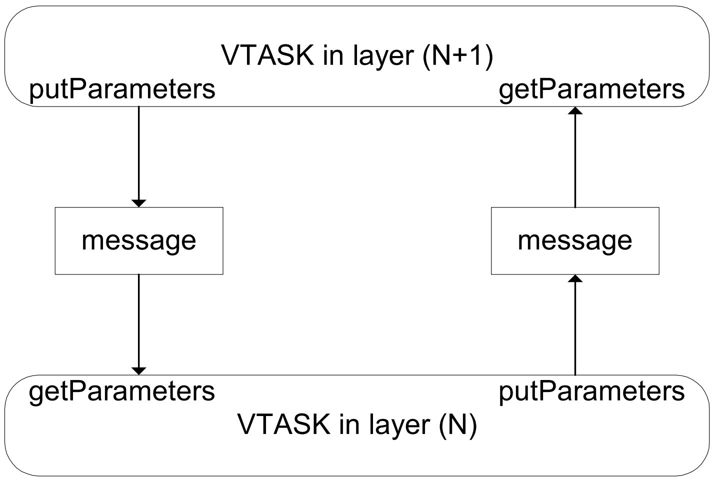

Figure 4.6 Parameter functions of CVOPS

The communication between CVOPS and environment goes through so called driver vtasks [10]. Driver vtask converts a message to the form it can be sent to the environment. Also when a message is received from the environment driver vtask creates CVOPS message and sends it to other vtasks [10].

4.4.4 SCIU

In NRC there are a lot of CVOPS and SDL protocol implementations (protocol libraries). It is beneficial to be able to use CVOPS and SDL protocols in a same system. The purpose of SCIU is to make CVOPS and SDL operate together.

The SCIU integrates CVOPS and SDT kernels so that they both have own message schedulers. CVOPS scheduler works as a main scheduler of the system. If it notices that there is a message in the SDT message queue, it calls SDT scheduler to handle that message. After the SDT scheduler has handled the message the control returns to the CVOPS scheduler.

SCIU driver is CVOPS vtask that is used to encode, decode, send and receive messages between SDT and other CVOPS vtasks. Because data types and messages are different in the SDT and CVOPS, the message parameters need to be converted to the data types used by the receiver of the message. Figure 4.7 shows how messages pass the driver [17].

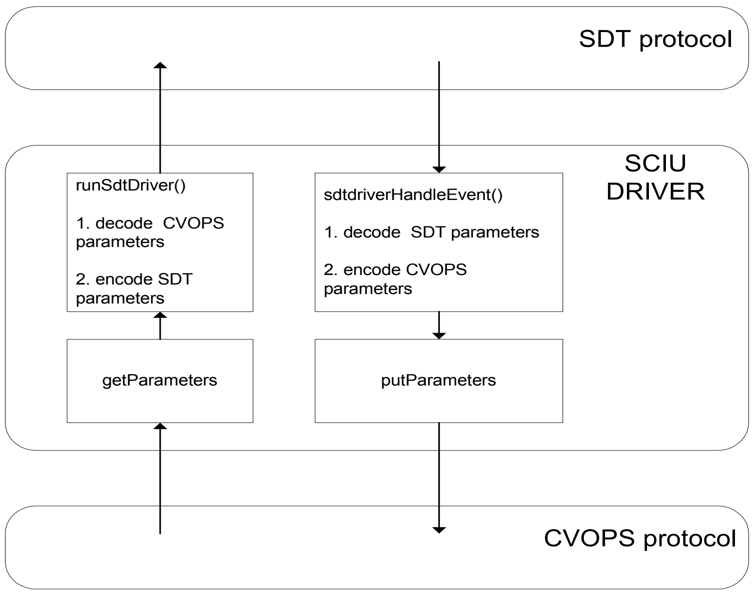

Figure 4.7 Message passing through SCIU driver

4.4.5 ASN.1

ASN.1 is a notation for defining data types and structures. It is widely used in communication industry, for example, 3GPP layer 3 protocol specifications use ASN.1 for defining messages. ASN.1 defines the abstract syntax of the data but it does not affect how the data is encoded. The basic principle is to define a small number of different data types by defining their possible values and give rules to combine these into increasingly complicated types. Packet Encoding Rule (PER) is one of the encoding rules that provides a transfer syntax for the ASN.1 types. ASN.1 and PER together specifies a machine-independent bit-pattern representation during transfer for ASN.1 types.

4.5 System Integration Principles

Most of the protocols were received from the other projects, so the main task was to integrate them. Because the protocols were coded using different tools and languages, the job was more difficult compared to the situation if they had all been coded with the same language. Also the primitive interfaces were defined differently, so there was a need to modify them. As depicted earlier in chapter 4.1, the protocols were not complete implementations, for example, some required parameters and information elements were missing. Therefore, it took a quite long time to implement those.

SCIU was needed to get SDL and CVOPS protocol implementations working together. Where CVOPS protocol was facing SDL protocol there was a need to implement a special vtask between them, called SCIU driver. The main task of the SCIU driver is to map the SDT primitives and their parameters corresponding CVOPS primitives and vice versa.

The UE stack has been divided so that the real-time protocols are in different process than the L3 protocols. The same is true on the RNC side.

The division was done because real-time protocols needed to achieve the requirements set by L1 and protocols itselves. Another issue was tracing, usually it was enough to see L3 level tracing and setting that on did not exhaust the real-time protocols.

The primitive interface between RRC and lower layers (PDCP, RLC, MAC and L1 control/switch) is complicated and because the primitives are defined using ASN.1, the parameter mapping in SCIU driver would have been troublesome without PER codec. In addition, because the ASN.1 tools generate C data structures from the ASN.1 definitions, the smallest change in the ASN.1 definition would have caused major modifications in SCIU driver. PER codec is a SDL block that uses encoding and decoding functions generated by the ASN.1 tools. PER coding of the structure before sending it to SCIU driver makes the mapping easier and there is a need to map only one parameter. On the receiver side, peer PER codec needs only decode the PER encoded parameter, after that the primitive can be passed to the receiver.

Figure 4.8 illustrates how the pdcp_ue_config_req –primitive is handled. When RRC sends primitive to PDCP, it goes first to PER codec. The PER codec encodes the message to transfer syntax and sends it to socket (actually there is SCIU driver between PER codec and TCP socket). On the receiver side, a message comes to PER codec and it is decoded to local syntax and the message is then passed on to PDCP.

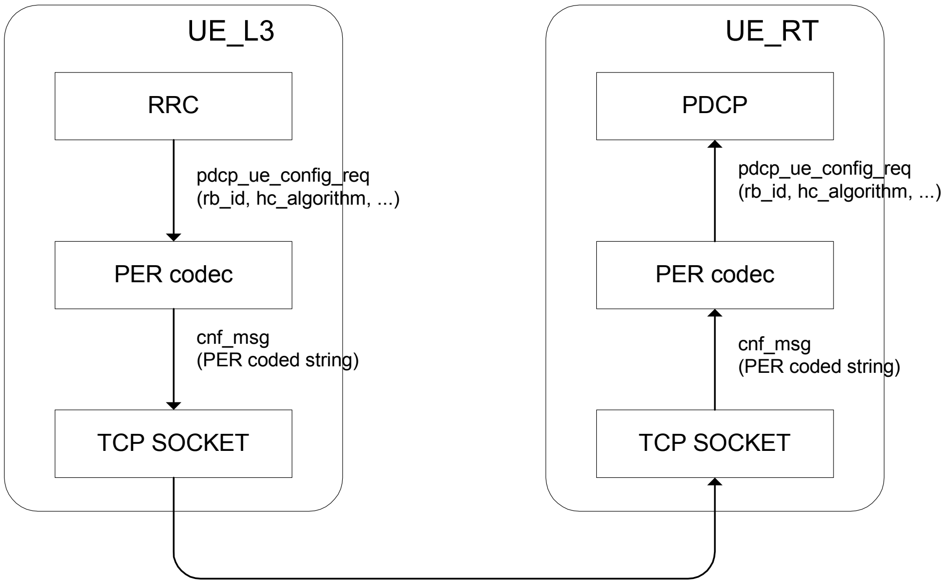

Figure 4.8 Primitive encoding and decoding

In the Pilot system the software development was done in Sun Solaris machines. A version control system was used there. The target environment, were the system was executed, was PC's with Linux operating system. The system was first tested in Solaris with real-time memory debugging tools (Purify) and other software development tools and after the system worked correctly in Solaris, it was transferred to Linux. It was not possible to test everything in Solaris because actual real-time testing with real application and WCDMA radio had to be done in Linux. One defienciency in software development was that the good memory debugging tools were not available in the target environment. Therefore the Gnu Debugger (GDB) was proved to be very valuable in Linux. Because in Solaris there were no ATM card, the system had to be changed slightly before it was possible to test in Solaris. ATM was then replaced with TCP socket.

The tracing of the system execution is essential and luckily SDT provides quite sophisticated tracing facilities for analysing the behaviour of the system. It is possible to set the tracing level depending on what kind of information is needed. Sometimes it is enough to see PDU level tracing but usually in a development phase developer needs to see also parameters of PDU and primitives. Using SDT trace it is possible to trace even all the assignments happening during the execution of the system. There is also developed a tracing method to the CVOPS. It is possible with it to examine for example, primitives, PDUs and their parameters and also the execution of the state machine. The SDT trace is too time consuming for the real-time testing because it usually produces too much information. Therefore, during the project an own tracing methods to the SDT kernel was developed. In software development a version management system is fundamental. In this project the version management system was chosen to be Revision Control System (RCS).

Another issue was integrating the protocols and WCDMA radio parts. Both sides had their own viewpoint to the system. It was valuable to have detailed descriptions of the interfaces and signalling charts. Due to the real-time environment the testing became complicated. WCDMA radio required 10 millisecond transmission window from MAC protocol. That needed some optimisation to SDL code, mainly to PDCP, RLC, MAC and L1 control/switch.

4.6 PDCP Implementation

The Packet Data Convergence Protocol (PDCP) is located in UE and in RNC. It is the topmost protocol of the radio link layer (Layer 2, L2), see Figure 4.9. The PDCP is used for compressing header information when data packets are sent over the radio interface. The PDCP has an interface to the RRC and RLC. It provides user plane radio bearers for carrying data packets.

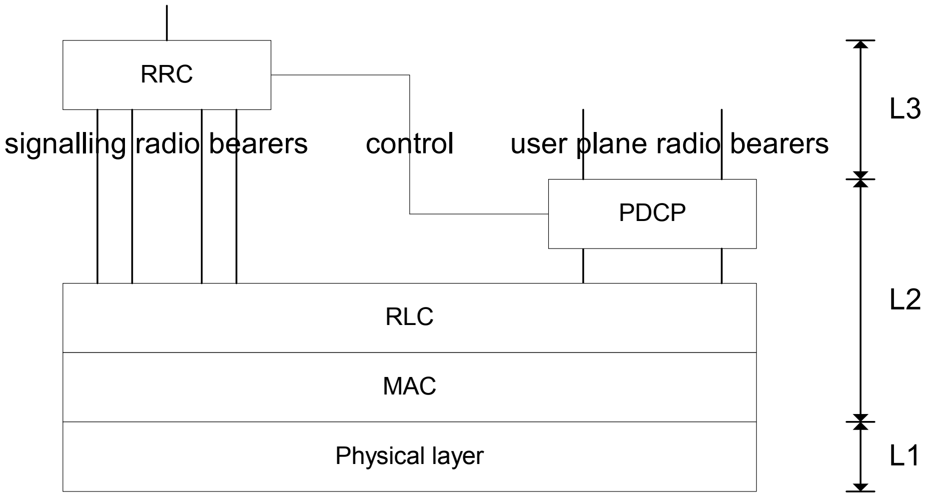

Figure 4.9 Environment of PDCP

PDCP entities are configured by RRC protocol. RRC may setup, reconfigure or release radio bearers and corresponding PDCP entities. Configuration is done via control interface. Every radio bearer is connected to one PDCP entity and one PDCP entity is connected to one RLC entity [4]. PDCP may use several different kind of IP header compression algorithms. The RFC 2507 is used to compress TCP/IP traffic, and the ROHC is able to compress IP/UDP and IP/UDP/RTP.

The signalling of the header compression algorithm to the PDCP layer is described in Figure 4.10. In the figure only the UE side is presented. The RNC side is similar to that. When the UE performs RRC Connection procedure to the network, that is a signalling connection between UE and UTRAN, it tells all the supported header compression algorithms to the RRM in message RRC CONNECTION SETUP COMPLETE (described in chapter 3.3.1). When a radio bearer is going to be setup between UE and UTRAN, RRM chooses whether to use the header compression algorithm or not. If it chooses an algorithm, it tells it with the RADIO BEARER SETUP message to the UE. Then UE knows how to configure the PDCP layer.

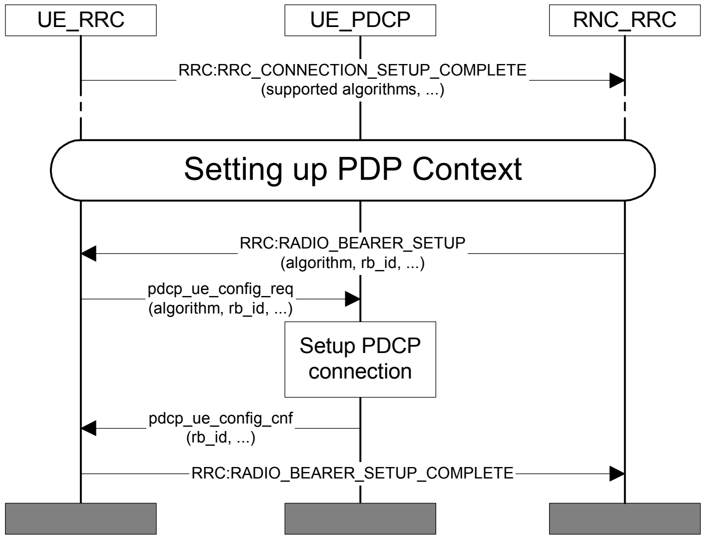

Figure 4.10 PDCP connection setup

SDL implementation of the PDCP contains two processes. One entity is called RED (Routing Encoding/Decoding), and the other one CONN (connection), see Figure 4.11. The RED is a static process, created when the system is started. It handles routing of the data packets and control signalling to the correct connections, to CONN process. Usually (in NRC's SDL protocol development) also encoding and decoding of the Protocol Data Unit (PDU) is done in the RED process [14]. But in case of PDCP there was a need to make it faster by reducing the number of the signals, and therefore encoding and decoding happens in the CONN process. RED handles creation of the dynamic CONN processes.

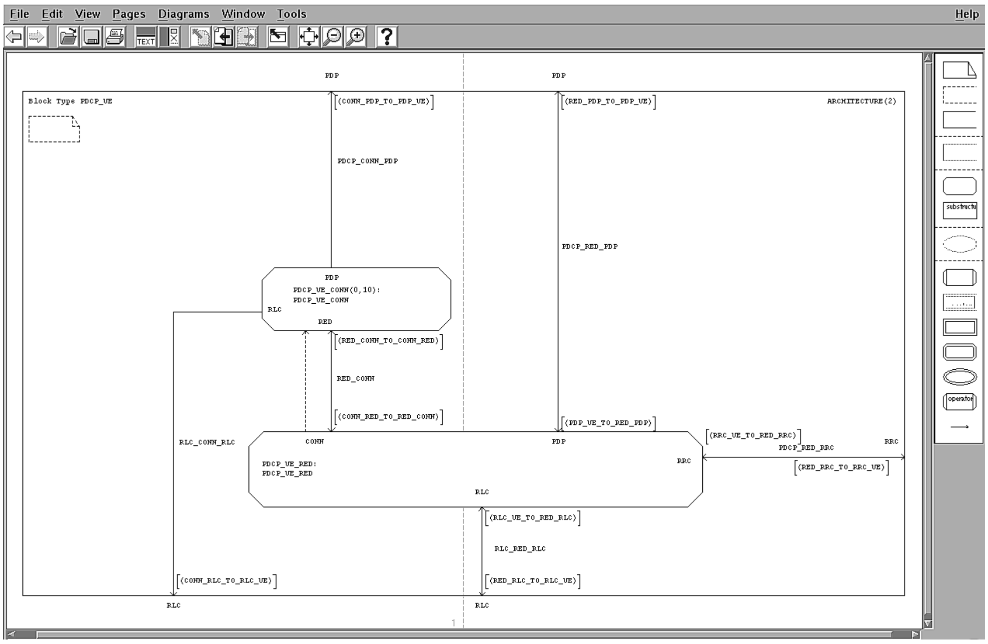

Figure 4.11 PDCP SDL block structure

PDCP uses three different RLC modes for sending data: acknowledged, transparent and unacknowledged modes. Figure 4.12 illustrates the signalling when data is compressed and sent from the UE to the RNC over the unacknowledged RLC mode.

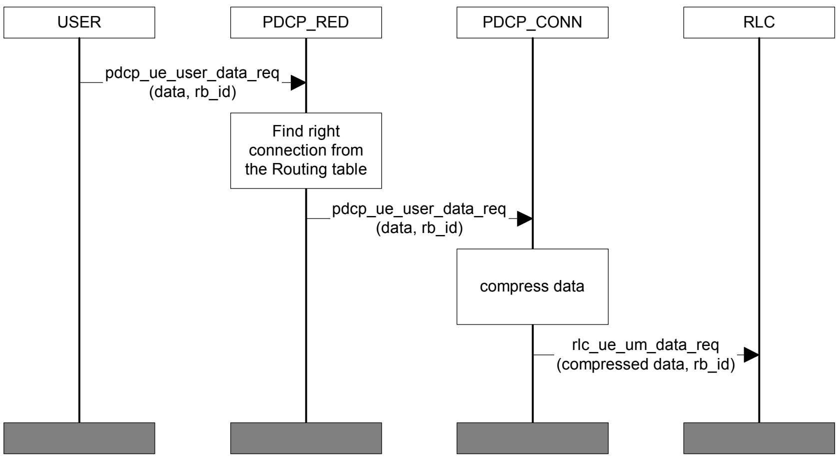

Figure 4.12 Data through PDCP, compressing

On the receiving side, Figure 4.13, signalling sequence is the opposite. *Data* is IP packet produced by the application, *compressed data* is compressed IP packet (for example compressed IP/UDP/RTP headers and uncompressed payload), *rb_id* identifies the radio bearer and *rnti* (Radio Network Temporary Identity) identifies the UE.

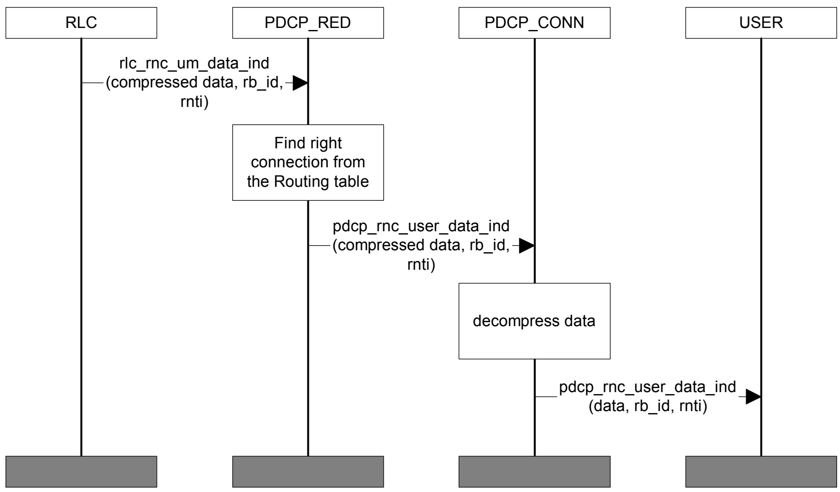

Figure 4.13 Data through PDCP, decompressing

<u>Header compression</u>

The Voice over IP call is usually IP/UDP/RTP based traffic. The packet then has IP header (20 bytes for IPv4, 40 bytes for IPv6), UDP header (12 bytes) and RTP header (8 bytes). Payload of the packet depends largely on the speech coding. It can be as low as 15-20 bytes [13]. In this case, most of the data transferred over air interface is header information.

Header compression of IP flow is possible due to the fact that the fields in the headers of IP packets are either constant or changing in a known pattern between consecutive packets in the same flow. It is possible to send only information regarding the nature of the changing fields of the headers with respect to the reference packet in the same IP flow. [6] Constant information in the headers, such as UDP port numbers and IP addresses, needs only to be sent in the beginning of the packet data session. Some other fields, such as RTP time stamp and RTP sequence number, change in a predictable way.

The Pilot systems PDCP implementation contains ROHC header compression algorithm. Algorithm has been implemented in NRC. It is coded using C language. Porting the algorithm to the PDCP and testing it with applications (used in project) was done during this project. Porting was done so that there is a clear function interface that can be called from the SDL code.

ROHC has been designed to be used in cellular environment. The cellular environment in this case means that the packets are sent over an unreliable link. It tolerates errors caused by error prone radio link. ROHC is able compress the original IP/UDP/RTP –headers (40 bytes in case of IPv4 and 60 bytes in case of IPv6) into 2 bytes. The compression efficiency is very significant.

Figure 4.14 describes how ROHC compression algorithm is called from the SDL code. Picture has been taken from PDCP_UE_CONN –process, leaving all unessential parts out. IP packet comes as a parameter of pdcp_ue_data_req –signal (v_frame equals IP packet). An incoming packet is converted to a form that can be handled by the ROHC algorithm. Then the compression algorithm is called and compress_str –parameter is defined to be of type "IN/OUT" meaning that it is passed by reference. The ROHC receives uncompressed IP packet in compress_str –parameter, compresses the packet and returns the compressed packet using the same parameter. After compression, a PDU is constructed (not shown in the picture) and send to RLC, by using unacknowledged transmission. The same function is called also in decompressing side.

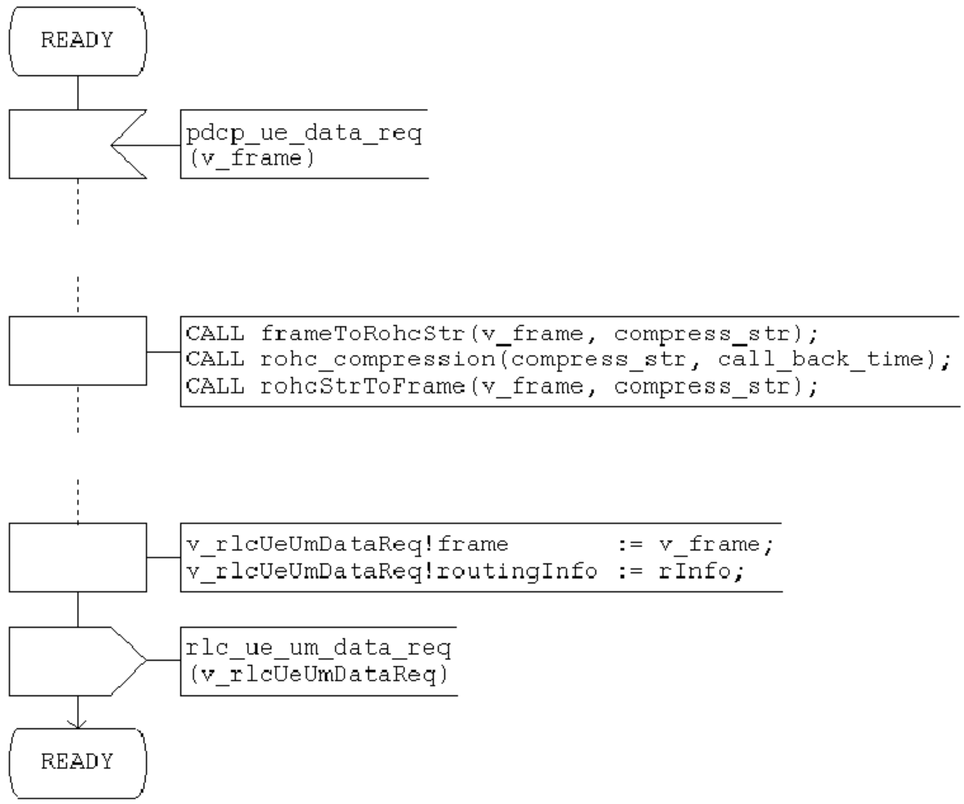

Figure 4.14 Calling compression function from SDL code

4.7 Interface to WCDMA Radio Parts

The system has an interface to real 3GPP WCDMA radio. The L1 Control block handles the configuration of the WCDMA radio entities. Because the configuration parameters were defined differently on protocol side and on the baseband side, there was a need to do mapping between them. This parameter mapping is done in L1 Control block.

Communication between protocols and baseband is done by using standard VME interface, see Figure 4.15. UE is on the left of the picture and it consists of a protocol, baseband and rf units. The UTRAN is located on the right of the picture.

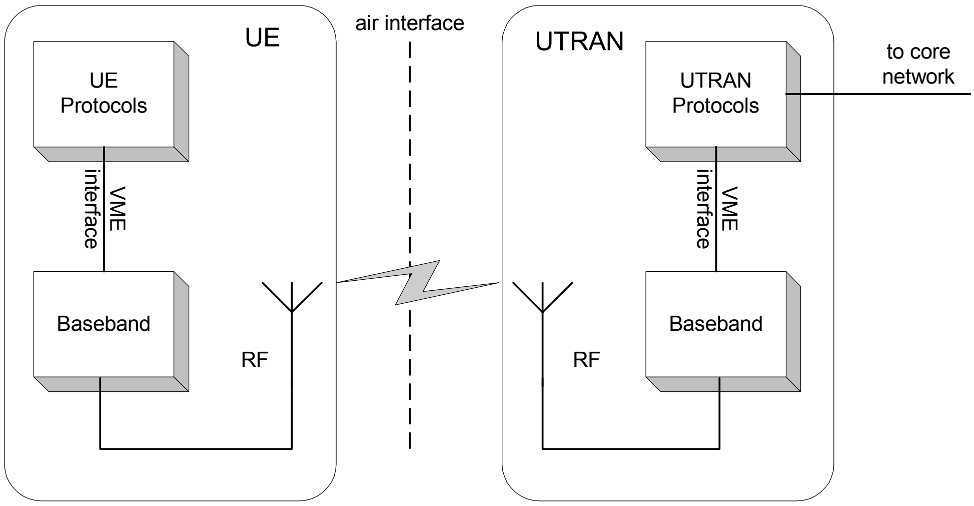

Figure 4.15 Baseband and RF

4.8 The Implementation of the QoS in the Pilot System

Following chapters describe what kind of Quality of Service methods have been implemented in the Pilot system. Header compression and packet scheduler are related to the air interface and Differentiated Service marking is related to core network. Header compression was described already in chapter 4.6.

4.8.1 Packet Scheduling

The packet scheduler manages radio resources by scheduling bandwidth between different traffic class bearers for both uplink and downlink directions. The packet scheduler is based on absolute priorities. The radio bearer with the highest priority is given possibility to send everything it has first and after that the radio bearer with the second highest priority etc. If there are radio bearers with the same priorities, they are all in a linked list where capacity is first served for the first one in the list and then the second one etc. Scheduling method is not fair: if the radio bearers with high priorities are requiring all the capacity, the radio bearers with lower priorities do not get anything. In the Pilot system there were no situation when two radio bearers had same priority values, so this kind of scheduling was adequate.

Capacity is based on transport blocks. A transport block equals to MAC PDU. A transport format is a format L1 offers data to MAC and vice versa. It consists of one or more transport blocks. MAC always selects the slowest transport format which can carry all the data RLC protocol has in its buffers.

In the Pilot system the priority value of the bearer is determined in Packet Scheduler unit of RRM. It is calculated from the values of traffic class and traffic handling priority, received in RAB ASSIGNMENT REQUEST message, as depicted in Figure 4.16. RRM calculates also the smallest transport format set that still fulfills bandwidth requirements informed in RAB ASSIGNMENT REQUEST message. The calculation is based on guaranteed bit rate field.

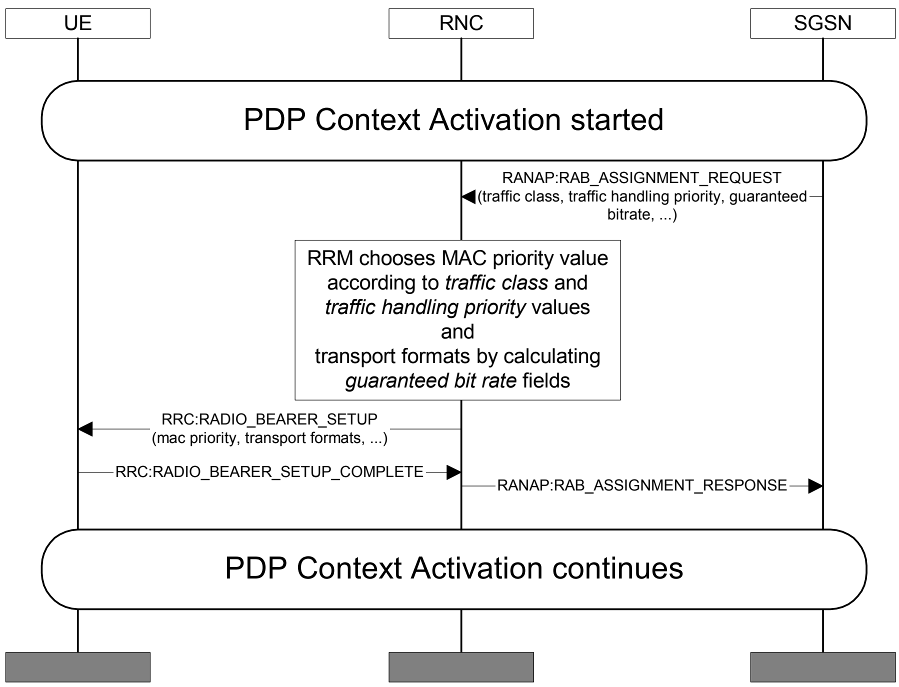

Figure 4.16 Setting the MAC priority and transport formats

The priority value and transport formats are then signalled to UE in RRC message RADIO BEARER SETUP. After receiving the message RRC configures the MAC with the right priority and transport format set values. Same happens in the UTRAN side before RADIO BEARER SETUP was sent.

4.8.2 Differentiated Service Marking

The Differentiated Services architecture provides a method to classify and partition allocation of resources among multiple traffic classes. IP packets are classified into service classes at the boundaries of the network. Differentiated Service marking is based on replacing IPv4 Type of Service (TOS) field, or in case of IPv6 Traffic Class field, with a Differentiated Service Code Point (DSCP). The DSCP tells how routers should treat the packet.

Figure 4.17 describes how the correct Differentiated Service Code Point is told to GTP-U in RNC. With that information GTP-U knows how to mark the IP packets going to uplink direction (from UE to GGSN).

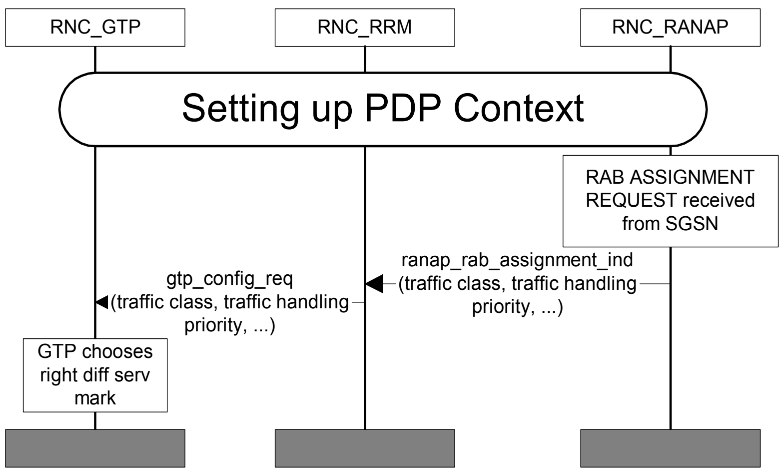

Figure 4.17 Differentiated Service marking

When RRM receives indication from RANAP to start radio access bearer setup procedure, it delivers traffic class and traffic handling priority parameters to the GTP-U. Based on that information GTP-U knows how to mark packets.

4.9 Interaction of the Application and Pilot System

This chapter describes how the application PCs and protocol stacks communicate together. First it is depicted how PDP contexts (primary and secondary) are activated and then how the user plane data traverse through the system.

Primary PDP Context Activation

MSINET is the process used for capturing user plane IP packets from ethernet to protocol stack. It also emulates DHCP (Dynamic Host Configuration Protocol) server to the application PC in case of IPv4 and takes care of stateless autoconfiguration procedure in case of IPv6. It connects IPv4/IPv6 application PCs to the protocol stack [12]. The primary PDP context activation procedure for IPv4 is shown in Figure 4.18.

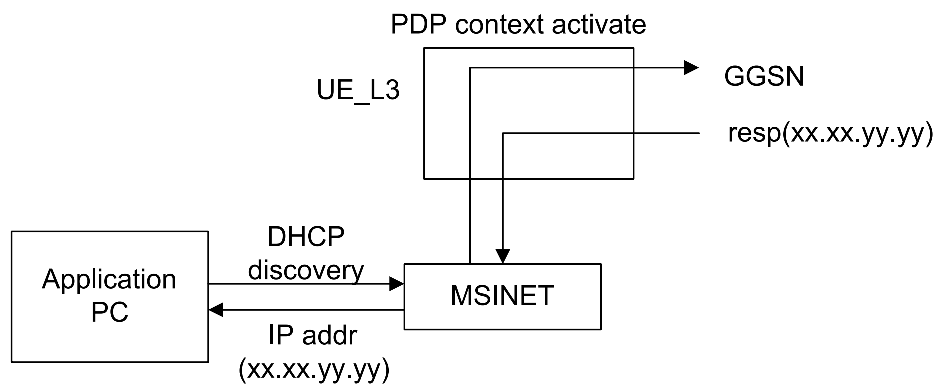

Figure 4.18 PDP context activation using DHCP

The PDP context activation procedure starts when the operating system of application PC sends a DHCP request for asking IP address from DHCP server. MSINET detects it and triggers the primary PDP context activation procedure with the predefined default QoS parameters. When the procedure is completed MSINET returns the IP address, allocated by the GGSN, to the application PC.

In case of IPv6 the situation is more complicated because there is no DHCP for IPv6 yet. The procedure starts when the MSINET detects the Router Solicitation message from the IPv6 application PC. This message contains application PC's link local address. Then MSINET triggers primary PDP context activation. With the PDP context activation accept message MSINET gets the Interface Identifier allocated by the GGSN. After that, as a data packet, MSINET receives IPv6 Router Advertisement containing network Prefix Information. Now MSINET can combine these addresses (Interface Identifier and Network Prefix) together and form IPv6 address for the application PC.

Secondary PDP Context Activation

The QoS and Session Manager is a graphical tool for setting up QoS parameters for secondary PDP contexts (i.e. applications). Using the tool it is possible to define own QoS parameters for video call, web browsing and other type of applications. The QoS and Session Manager conveys session activation, modification and deactivation requests sent from the application to the UE emulator which then start PDP context activation, PDP context modification or PDP context deactivation procedures for given application.

Figure 4.19 shows how the PDP context activation procedure is performed using 3G Concept Phone and the QoS and Session Manager.

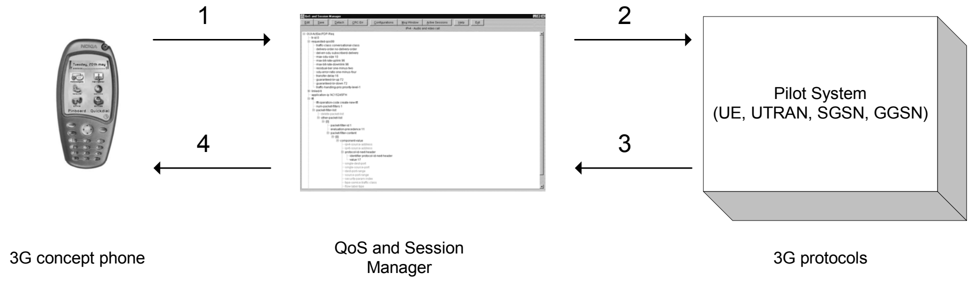

Figure 4.19 Environment of QoS and Session Manager

1. The end user selects the application from the UE. Let us assume that the application is a video call. The 3G Concept Phone then sends a request to QoS and Session Manager for reading QoS parameters for video call.
2. QoS and Session Manager requests UE protocol stack to activate secondary PDP context with the parameters defined beforehand for given application type in QoS and Session Manager.
3. Protocol stack responds with the secondary PDP context accept (in case of successful activation) to QoS and Session Manager. The message contains negotiated QoS parameters.
4. QoS and Session Manager informs 3G Concept Phone that the context has been setup and the application may start to signal the actual application (video call).

For the video call the QoS parameters of the secondary PDP context activation request can be as depicted in Figure 4.20. These parameters were described in chapter 3.2. There is also defined Traffic Flow Template for the video call. Filtering in this case is based on the protocol carried as a IP payload. Value 17 equals to UDP, so all UDP packets are routed to this PDP context (it also depends on evaluation precedence).

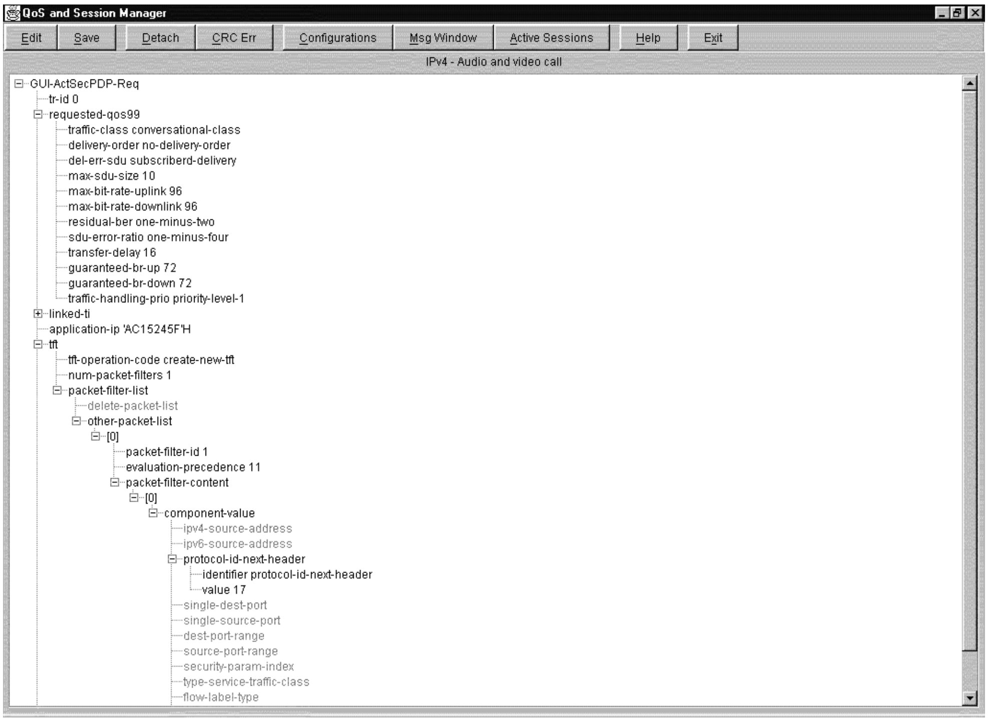

Figure 4.20 QoS and Session Manager

The end user can also see what kind of applications are already activated, Figure 4.21, by clicking "Active Sessions" – button.

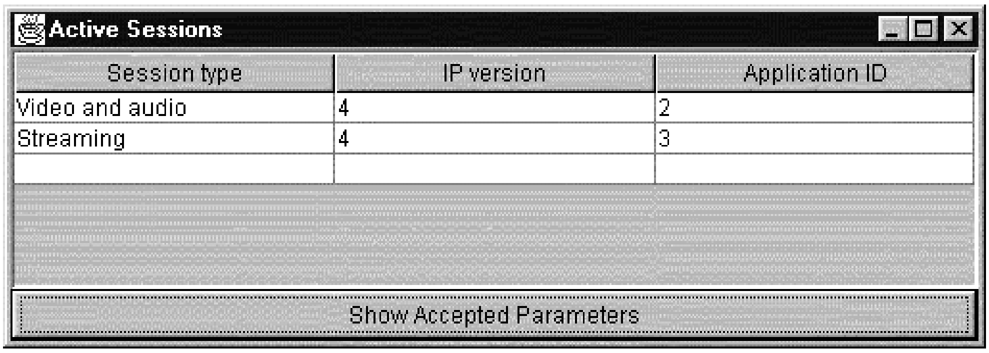

Figure 4.21 Active Sessions

When the context has been setup, the user can check from Active Sessions window what were the negotiated QoS parameters received from the network.

Data transmission

Because UE may have now multiple active PDP context, primary and one or more secondary contexts, it is necessary to route IP packets to the correct PDP context. Routing is based on Traffic Flow Template (see chapter 3.2.3), which is signalled to GGSN during the activation procedure. TFT contains filters for routing of the packets, see Figure 4.22.

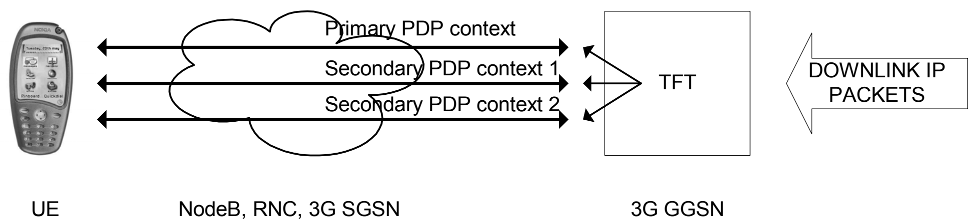

Figure 4.22 TFT filtering

Figure 4.23 shows the flow of the user plane traffic through the system from the application PC connected to the UE to the SGSN. Packets returning from the SGSN travel the same route.

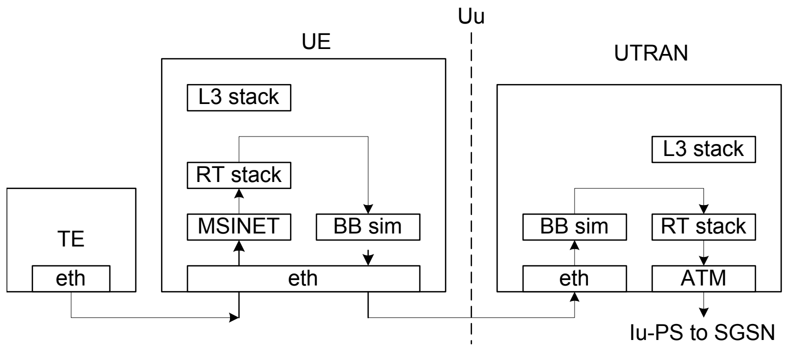

Figure 4.23 Packet flow

TE is a application PC and RT stack runs the real-time protocols, i.e. PDCP, RLC and MAC. In addition to that in the UTRAN side the GTP-U is also run in RT stack. BB simulator is used to replace the functionality of the real 3GPP WCDMA radio. Uu is the air interface.

4.10 Testing and Validation of the Protocol Implementations

The protocol implementations in the Pilot system have been done according to the 3GPP specifications. Same protocol implementations are used in several testers for testing Nokia's 3G products and product testing phases have proved 3GPP compliance.

The MTP3b server and the RObust Header Compression algorithm have been in a bake-off sessions where they have been tested against the implementations of other manufacturers.

5. CONCLUSION

As the second generation mobile telecommunication systems are mainly designed for speech, the third generation systems are designed taking into account technology needs for mobile multimedia telecommunication. The portion of data is going to bypass speech in the near future. Because the data is usually carried over packet switched connections, there is a need for classification the traffic into different traffic classes in order to ensure promised quality requirements.

In this thesis the focus was in implementation of the Third Generation Pilot system. From the QoS point of view some important methods were described, especially those interacting with radio interface. With the help of the Pilot system, is was possible to see in practise how different methods affected to the behaviour of the whole system.

The described QoS methods were, packet scheduling, ROHC header compression and Differentiated Service marking. MAC scheduling was proved to be practical, by prioritising different traffic class bearers/logical channels. ROHC reduced significantly the reduntant information sent over the air interface. The functionality of the Differentiated Service marking was not yet ensured because no routers supporting marking was available.

The SDL implementation of the PDCP was described. It was found that SDT is a very useful tool for designing and implementating telecommunication protocols. Especially it is valuable for rapid prototyping and piloting. The behaviour of the system and protocols is easy to observe with help of SDT tracing facility. Integrating SDT and CVOPS protocols was straightforward with help of the SCIU.

It was also found out that Linux operating system is capable enough for running real-time protocol implementations. That was proved by connecting protocols to the real 3G WCDMA radio interface.

The Pilot system was demonstrated in several conferencies during 2000 and 2001, such as Cebit, Cannes and UMTS conference in Barcelona. The system is also installed to Nokia's 3G truck and it tours around the Europe demonstrating the system to teleoperators. The system is also used as a testing platform of the 3G applications.

This project gave valuable knowledge to author both in understanding how UMTS operates and also how to develop and implement Pilot systems. In the near future the Pilot system is going to be connected to the Nokia's 3G SGSN product. The results of this project will be used as a basis of new WCDMA based pilot systems.

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APPENDIX 1: Protocol stacks