Radio Access Bearer Management Protocol in 3rd Generation Mobile Telecommunications System

Master's Thesis

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HELSINKI UNIVERSITY OF TECHNOLOGY Department of Computer Science and Engineering	ABSTRACT OF MASTER'S THESIS

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The development of third generation mobile telecommunication systems, often simply called 3G, is proceeding rapidly. The Universal Mobile Telecommunications System (UMTS) is one of the 3G systems.

This thesis gives an overview of the UMTS system. The UMTS network architecture with the network elements is described. The protocol stacks in different interfaces are presented. Both circuit- and packet-switched types of services are included in UMTS from the beginning. The services UMTS offers are shortly covered.

The main topic of this thesis is the bearer management in UMTS. The bearer concept of UMTS is given and explained. The relation between the UMTS services and bearers is presented. All the different types of UMTS bearers and their purpose of use are described. In UMTS the functionality of the bearer management is distributed to the different network elements. The co-operation of the network elements in bearer management is covered. Some example signaling flows are given to clarify the details of bearer management functions.

In this thesis one specific protocol participating the bearer management is studied in detail. This protocol is Radio Access Network Application Part (RANAP). The role of RANAP in bearer management is described. An implementation of RANAP is also presented.

Keywords: UMTS, 3G, bearer, bearer management, protocol, RANAP

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Kolmannen sukupolven matkapuhelinjärjestelmien kehitystyö on hyvässä vauhdissa. Universal Mobile Telecommunication System (UMTS) on yksi kolmannen sukupolven matkapuhelinjärjestelmistä, joita usein nimitetään lyhyemmin 3G:ksi.

Tässä työssä annetaan yleiskuvaus UMTS-järjestelmästä. UMTS-järjestelmän arkkitehtuuri verkkoelementteineen käydään pääpiirteissään läpi. Myös eri rajapintojen protokollapinot kuvataan yleisesti. UMTS-järjestelmä tarjoaa edeltäjistään poiketen alusta alkaen sekä piiri- että pakettikytkentäisiä palveluja. Tärkeimmät palvelut ja niiden ominaisuudet esitellään lyhyesti.

Diplomityön pääpaino on UMTS-järjestelmän siirtoväylien hallinnassa ja siihen liittyvässä toiminnallisuudessa. Tarkastelussa lähdetään liikkeelle esittelemällä UMTS-järjestelmän siirtoväylän käsite. Palveluiden ja siirtoväylän suhde sekä erityyppisten siirtoväylien käyttötarkoitus käsitellään kattavasti. Varsinaisten siirtoväylien hallintatoiminnat esitellään käymällä läpi verkkoelementtien välinen yhteistyö siirtoväylien pystyttämisessä ja purkamisessa. Siirtoväylien hallintaan liittyviä yksityiskohtia selvitetään esimerkeillä ja signalointikaavioilla.

Työssä käsitellään tarkemmin erästä siirtoväylien hallintaan osallistuvista protokollista nimeltään Radio Access Network Application Part (RANAP). RANAP-protokollan tärkeimmät tehtävät ja sen osuus yleisessä siirtoväylien hallinnassa käydään läpi yksityiskohtaisesti. Lisäksi esitellään RANAP-protokollasta tehty esimerkkitoteutus.

Avainsanat: UMTS, 3G, siirtoväylä, siirtoväylän hallinta, protokolla, RANAP

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Espoo, 4th January 2000

Janne Tervonen

TABLE OF CONTENTS

ABBREVIATIONS

3G	The Third Generation Mobile Communication Systems
3GPP	3rd Generation Partnership Project
3GPP2	3rd Generation Partnership Project 2
AMPS	Advanced Mobile Phone Service
APN	Access Point Name
ARIB	Association for Radio Industry and Business
ASN.1	Abstract Syntax Notation One
ATM	Asynchronous Transfer Mode
BER	Bit Error Rate
BS	Base Station
CC	Call Control
CDMA	Code Division Multiple Access
CM	Connection Management
CN	Core Network
CS	Circuit-Switched
DECT	Digital European Cordless Telephone
DL	Downlink
EDGE	Enhanced Data Rates for GSM Evolution
FDD	Frequency Division Duplex
FDMA	Frequency Division Multiple Access
ETSI	European Telecommunication Standards Institute
FPLMTS	Future Public Land Mobile Telephone System
GPRS	General Packet Radio System
GSM	Groupe Special Mobile (old)
GSM	Global System for Mobile Communications (current)
HLR	Home Location Register
HSCSD	High Speed Circuit Switched Data
ID	Identifier
IETF	Internet Engineering Task Force
IMT-2000	International Mobile Telecommunications - 2000
IPR	Intellectual Property Right
ISDN	Integrated Services Digital Network
IP	Internet Protocol
ITU	International Telecommunication Union
ITU-R	Radio Communication Study Groups of ITU
ITU-T	Telecommunication Standardization Sector of ITU
MAC	Medium Access Control
MM	Mobility Management
MOC	Mobile Originated Call
MPEG-4	Motion Picture Expert Group 4
MSC	Mobile Switching Center
MSC	Message Sequence Chart
MSS	Mobile Satellite System
MT	Mobile Termination
MTC	Mobile Terminated Call
NBAP	Node B Application Part
NMT	Nordic Mobile Telephones

NSAPI	Network Service Access Point Identifier
OSI	Open System Interconnection
PER	Packet Encoding Rules
PCS	Personal Communication Services
PDC	Personal Digital Cellular
PDP	Packet Data Protocol
PDU	Protocol Data Unit
PID	Process Identifier
PS	Packet-Switched
QoS	Quality of Service
RAB	Radio Access Bearer
RAN	Radio Access Network
RANAP	Radio Access Network Application Part
RB	Radio Bearer
RLC	Radio Link Control
RNC	Radio Network Controller
RNS	Radio Network Subsystem
RRC	Radio Resource Control
SCCP	Signaling Connection Control Part
SDL	Specification and Description Language
SDT	SDL Design Tool
SGSN	Serving GPRS Support Node
SIM	Subscriber Identity Module
SM	Session Management
SMS	Short Message Service
SRNC	Serving Radio Network Controller
SS7	Signaling System Number 7
TDD	Time Division Duplex
TDMA	Time Division Multiple Access
TE	Terminal Equipment
TI	Transaction Identifier
TTA	Telecommunications Technology Association
TTC	Telecommunications Technology Committee
UDP	User Datagram Protocol
UE	User Equipment
UL	Uplink
UMTS	Universal Mobile Telecommunications System
UTRAN	UMTS Terrestrial Radio Access Network
UWC	Universal Wireless Communications
VLR	Visitor Location Register
WARC	World Administrative Radio Conference
WCDMA	Wideband Code Division Multiple Access
WWW	World Wide Web

1 INTRODUCTION

The mobile communication market has enjoyed enormous growth during the past years. Operators worldwide get new customers at an increasing rate, new networks are being built and mobile phone manufacturers sell more and more handsets in the expanding markets. Twenty years ago the mobile communication markets were almost non-existent, but the worldwide deregulation of the telecommunications sector and rapid development of radio technology made mobile phones available for everyone. Now mobile communication is an important part of our everyday lives.

As a new and highly profitable field of technology mobile communication is intensively researched. The traditionally slow moving telecommunications sector is developing constantly due to new emerging technologies and innovations. The next large-scale development step will be the introduction of the third generation mobile communication system.

UMTS (*Universal Mobile Telecommunications System*) is one of the third generation systems. It offers a wide variety of services. Both circuit- and packet-switched services are included from the beginning. To support the different requirements of the services a bearer concept was introduced to UMTS. The functions needed for the bearer management are distributed to the different UMTS network elements and protocols. The main objective of this thesis is to give an overview of the UMTS bearer management and to describe the distribution of the responsibilities in it. The focus is on one specific protocol, RANAP, and on its role in the bearer management.

This thesis is organized as follows. Chapter 2 gives general information about the third generation mobile communication systems and UMTS. Also the international standardization process of the third generation systems is presented. Chapter 3 concentrates on the UMTS system and its network architecture. The bearer management in UMTS is described thoroughly in Chapter 4. The details of the RANAP protocol are given in Chapter 5. In Chapter 6 the experimental part of the thesis, an example implementation of the RANAP protocol, is presented. Finally, Chapter 7 concludes the thesis.

2 3RD GENERATION MOBILE COMMUNICATION SYSTEMS

2.1 Background

The mobile communications era started in 1980. The first mobile communication systems offered only speech services with relatively high price. These first generation systems were based on the same basic principle of cellular network as all the successors: wide-area, public network is made up of multiple radio base stations, also called cells. The first generation systems utilized analogue transmission technology. They were designed mainly for national or regional use. Several standards were developed: for example, AMPS (*Advanced Mobile Phone Service*) was used in the USA and NMT (*Nordic Mobile Telephones*) in Scandinavia. Millions of mobile subscribers are still using the services of the first generation systems.

The second generation systems using digital transmission technology were introduced in the late 80s. The services offered were more advanced compared to the first generation systems: speech quality was improved and data services were introduced also to mobile communication systems. The second generation systems were intended for continent-wide usage. GSM (*Global System for Mobile Communications*) from Europe, PDC (*Personal Digital Cellular*) from Japan, and both IS-136 (also known as digital AMPS) and IS-95 from the USA are the most widespread second generation technologies. At the moment the majority of the mobile communication users are using the second generation systems.

The third generation mobile communication systems, often called *3G*, are even further improved from the second generation systems. In the following the overall description of 3G systems is presented. The key drivers for 3G are discussed first, then the high-level differences between second and third generation systems are described. Also the standardization process of 3G is presented.

2.2 The Key Drivers for 3G

The mobile communication has been an enormous success and at the moment there is no limit in sight for the increase of the number of mobile subscribers. For example in Finland, mobile subscriber penetration (the number of phones per capita) exceeded the limit of 60 percent at the beginning of April 1999. This is the highest mobile phone penetration in the world so far. Also, in December 1998 there were more mobile phone than fixed subscriptions in Finland [MTC99]. For some years Finland has pioneered and introduced the latest development trends in the telecommunications sector. If the rest of the world followed these trends, the number of mobile subscribers in the world could rise to hundreds of millions, even billions.

The International Telecommunication Union *(ITU)* has made some forecasts about mobile subscription growth worldwide. In Figure 1 ITU's forecast, ranging from the year 1990 to 2010, is presented. Note that all the mobile subscribers from the first, second, and third generation mobile communication systems are included in the figures. In ITU's forecast the growth of mobile subscriptions will continue steadily, and at the end of the year 2000 there will be 500 million mobile subscribers globally. The limit of one billion will be reached around the year 2005.

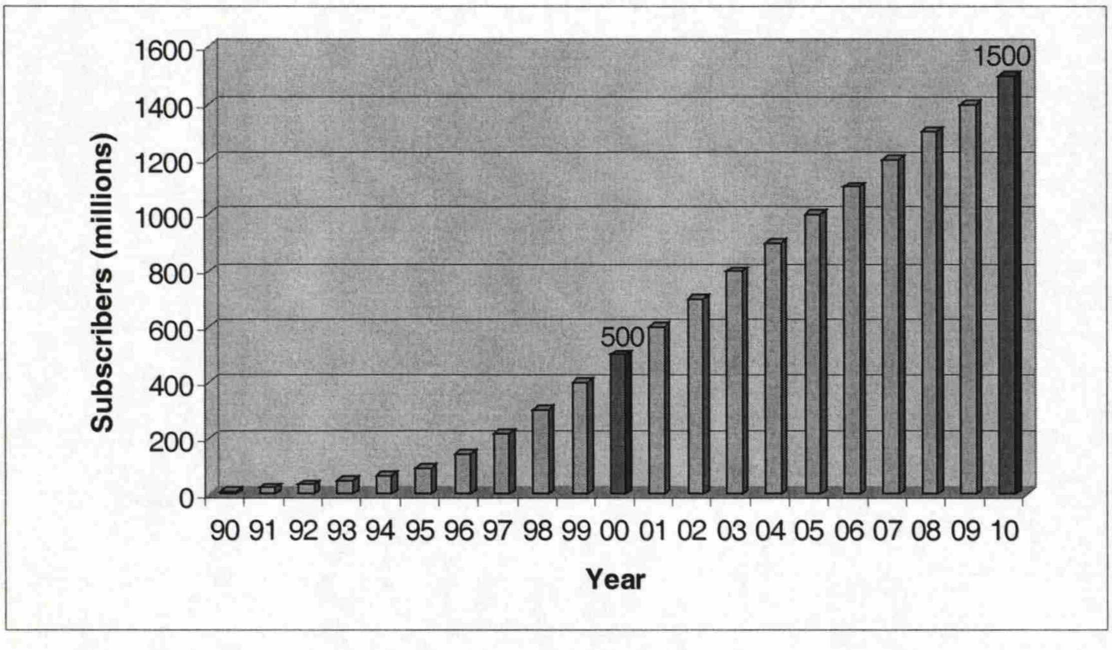

Figure 1. Cellular mobile subscribers worldwide [ITU99a].

If ITU's forecast comes true, which both mobile telecommunication operators and network and handset manufacturers hope, there will be a huge amount of traffic in mobile communication networks at every moment of time. This will cause problems: operators start to run out of network capacity, especially in terms of radio frequency in densely populated industrialized countries, and the quality of service will degrade. To overcome this operators need more capacity, in one form or another, to serve the increasing number of subscribers. When the resources of the existing first and second generation mobile communication networks are exhausted, operators will look for new solutions: the third generation mobile communication systems will offer the needed capacity.

Another key driver for the 3G systems is data. With the huge popularity of the Internet data communication has grown substantially. Especially the amount of transferred data has increased exponentially due to bandwidth-hungry applications like World Wide Web *(WWW)* and e-mail. It is even estimated that data traffic in general will exceed voice traffic volumes in the very near future [RoZd98].

Also mobile subscribers want to get the access to the data networks. The problem is that the existing solutions for mobile data access are too slow. For example, the current data transfer speed plain GSM networks offer is 9.6 kbit/s. This is quite low compared to the bit rates of fixed line services: a conventional modem can transfer data with 56 kbit/s and ISDN line with 128 kbit/s. The operators are convinced that the customers are willing to pay for faster mobile data access. In the 2nd generation systems the emphasis was in the speech services, but in 3G the emphasis will change from speech to data services.

2.3 3G Features

The 3G systems have to offer something new compared to the existing mobile communication systems in order to be successful. Improved and completely new services are required if the 3G systems are wanted to be accepted among the end-users. The following objectives have been taken as a basis for the 3G systems development: [ITUR97a], [ITUR97b], [OjPr98]

1. Higher data transfer rates than the existing systems can offer
   - Full coverage and mobility for 144 kbit/s, preferably 384 kbit/s
   - Limited coverage and mobility for 2 048 kbit/s
2. High flexibility to introduce new services
   - Multimedia
   - Video call
3. International roaming
   - Roaming between all third generation mobile communication systems worldwide
   - Interoperability with second generation networks
4. High radio spectrum efficiency to allow more users with less radio resources

2.3.1 Fast Data Transfer

The most considerable change in 3G systems is in the data transfer capacity: the users can have as high as 2 Mbit/s connections to the network. This reflects the shift from the speech services to data services in mobile communications. The second generation mobile systems were designed to carry mainly mobile voice with some data, but 3G networks are designed to carry more mobile data than voice traffic.

High data transfer rates can be used for accessing the Internet faster, but they create possibilities for new services too: with high bit rates it is possible to develop eg truly working video call. High transfer rates with flexible introduction of new services allow service providers to develop a wide variety of new services when the demand among the customers is recognized.

2.3.2 Global Roaming

3G systems will be the first mobile communication systems with global roaming. The roaming can be divided into two types, subscriber and mobile phone roaming. The mobile phone roaming means that the same piece of equipment works in the same way in all the networks. In other words a mobile phone purchased from Finland should work without any difficulties in every corner of the world.

With subscriber roaming it is possible to use the same network services in different networks by changing the SIM (*Subscriber Identity Module*) from one mobile phone to another. For example, when traveling to the USA you can take only your SIM card with you and plug it in a mobile phone rented in the USA. Both subscriber and mobile phone roaming should work in the 3G systems.

In addition to roaming in 3G networks, it should be possible to change the network from 3G to the second generation system. This should provide seamless basic service even when there is no 3G network coverage available.

2.3.3 Multicall

Multicall is a central new feature in 3G systems. Multicall means that the user can have several, simultaneous, independent calls active to one terminal. For example the user could have an active speech call while transferring file and receiving e-mail with the same mobile phone. Multicall combined with high speed data transfer offers platform for completely new types of services.

2.4 Standardization

Standardization is in central position in telecommunications. Without international standards there would not be true interoperation between the systems of different manufacturers. International, or preferably global, standards benefit everyone. The end-users and operators enjoy low prices as a result of the competition between several manufacturers. For the manufacturers bigger international markets mean a bigger group of potential customers. Also, large scale mass production lowers the costs of production.

Basically, the technical standardization of 3G systems can be divided to two main areas: the radio network technology standardization and the core network technology standardization. Even if the international standardization work might seem to concentrate more on radio network technology standardization, both areas are equally important: both radio network and core network are needed to a complete, working mobile communication system.

2.4.1 The Objective: Global 3G Standard

Already in 1985 ITU started the development work for the third generation mobile communication systems. At that time the new system was termed as Future Public Land Mobile Telephone System, FPLMTS. Afterwards the name was changed to more descriptive one, IMT-2000 (*International Mobile Telecommunications - 2000*). ITU's initial objective was to develop a single, global standard for the third generation mobile networks, but later ITU has had to reshape its objectives.

Although ITU was preparing global standards for 3G, the actual research work for the technical solutions of 3G networks was done regionally: Europe, Japan, Korea and USA were the most active, each concentrating on and promoting their own technology. The idea was that regional standardization or other active bodies would, after intensive and profound research, submit their proposal for IMT-2000 technology to ITU. After this ITU would choose the best solution, and based on that develop a global standard for the third generation mobile communication systems.

2.4.1.1 European-Japanese Co-operation

Japan has been very active in developing the third generation solutions. Japanese second generation mobile network, PDC, is deployed only in Japan. Traditionally strong Japanese consumer electronics manufacturers got only minor market shares in second generation mobile communication business. Now they did not want to miss the huge opportunities 3G global markets offer. Japanese 3G networks should be in commercial use already in 2001, almost a year before the rest of the world will gradually follow [3GM99b].

At the beginning of 1997, Japanese standardization body ARIB *(Association for Radio Industry and Business)* decided to adopt WCDMA *(Wideband Code Division Multiple Access)* for the radio technology for its 3G solution. This decision accelerated the standardization work also in Europe and USA.

In Europe ETSI (*European Telecommunication Standards Institute*) had done extensive work in standardizing European second generation mobile system, GSM. That work was taken as a basis for the new third generation system, known as UMTS. While ETSI was developing its own specifications for UMTS, European telecommunication companies were co-operating closely with the Japanese. During the fall 1997 co-operation got even closer when Japanese and European proposals for WCDMA-based 3G radio technology were harmonized. Also, a common core network solution was found: Europe's and Japan's 3G networks will be based on evolved GSM core network and WCDMA radio technology with two modes, FDD *(Frequency Division Duplex)* mode and TDD *(Time Division Duplex)* mode [MiPi99]. In essence Japan and Europe have reached mutual understanding of the 3G technology that was submitted as a proposal to ITU's IMT-2000 standardization.

2.4.1.2 USA's Approach

In USA standardization process has been traditionally more business driven than in Europe or Japan. On one hand, this leads to faster reaction to changes in the market, but on the other, it leads to a larger number of standards. The selection between standards is then left to the market forces [OjPr98]. At the moment, there are three second generation systems in use in the USA: one CDMA-based system called IS-95, and two TDMA-based *(Time Division Multiple Access)* systems, IS-136 and European GSM. American operators using GSM have been supporting the European-Japanese view of 3G (WCDMA radio technology) but both IS-95 and IS-136 camps were promoting technology of their own.

For 3G technology IS-95 camp submitted IMT-2000 proposal named cdma2000. Like UMTS, cdma2000 is based on wideband CDMA, but it is not compatible with Japanese-European WCDMA. IS-136 supporters prepared an IMT-2000 proposal called UWC-136 that is based on wideband TDMA radio technology. Despite of the fact that there were two separate radio technology proposals, the two rival American camps could agree on common core network technology. Both cdma2000 and UWC-136 will utilize core network component called IS-41 that is not compatible with GSM core network.

2.4.2 From Single Standard to Family of Standards

In addition to proposals from USA, Europe, and Japan ITU received several other proposals eg from Korea and China. These other proposals were in practice similar to the ones described above, so ITU had in its hands three proposals for radio technology and two for core network.

The operators worldwide have invested billions of dollars to the infrastructure of the mobile communication networks. The operators are willing to continue to use certain parts of existing networks also with 3G systems. That is why operators started to promote the idea of evolution from the second generation systems to the third: 3G will not be a completely new system, it will be based on existing systems.

Even ITU accepted the idea of evolution partially. Already in 1997 ITU recognized that there would be two different 3G core network solutions: GSM and IS-41. ITU still continued to promote the view of one single radio technology. But neither cdma2000, WCDMA nor UWC-136 camps wanted to adopt any other radio technology than their own. To complicate matters even further, the supporters of the two rival CDMA technologies, WCDMA and cdma2000, started to argue about the patents of CDMA technology. The patent dispute heated up: wireless industry began to talk about 3G war. The battle concentrated between the companies Qualcomm (cdma2000 supporter) and Ericsson (WCDMA supporter) while the others waited and watched with increasing impatience. The situation got serious: the patent dispute threatened the whole 3G standardization process.

Also the standardization scene reshaped. In December 1998 the standardization bodies (including ETSI and ARIB) that were backing up WCDMA radio technology, founded standardization organization called 3GPP *(3rd Generation Partnership Project)* [3GPP98]. cdma2000 camp responded quickly: a similar organization called 3GPP2 was established in January 1999 to produce the technical specifications for cdma2000 [3GPP299]. As a consequence ITU's role was changing. ITU was moving towards setting only global framework for 3G systems but it would not participate in drafting technical specifications.

Eventually things started to calm down. In March 1999 ITU had to announce that there would be no single radio standard in IMT-2000 [ITU99b]. Even Qualcomm and

Ericsson settled their dispute. As a result the overall shape of the 3G standards universe seemed more or less set.

ITU's framework for IMT-2000 was now ready. It is summarized in Figure 2. Instead of single, global solution there will be two separate CDMA-based air interfaces (WCDMA and cdma2000) and an additional TDMA interface (UWC-136) [PCSW99]. Both cdma2000 and TDMA will utilize IS-41 core network component. WCDMA will be connected to an upgraded GSM core network: this system is called UMTS. In the figure, UMTS-part of IMT-2000 family is shaded with an ellipse. This thesis will concentrate on UMTS system, the other members of IMT-2000 family are not discussed.

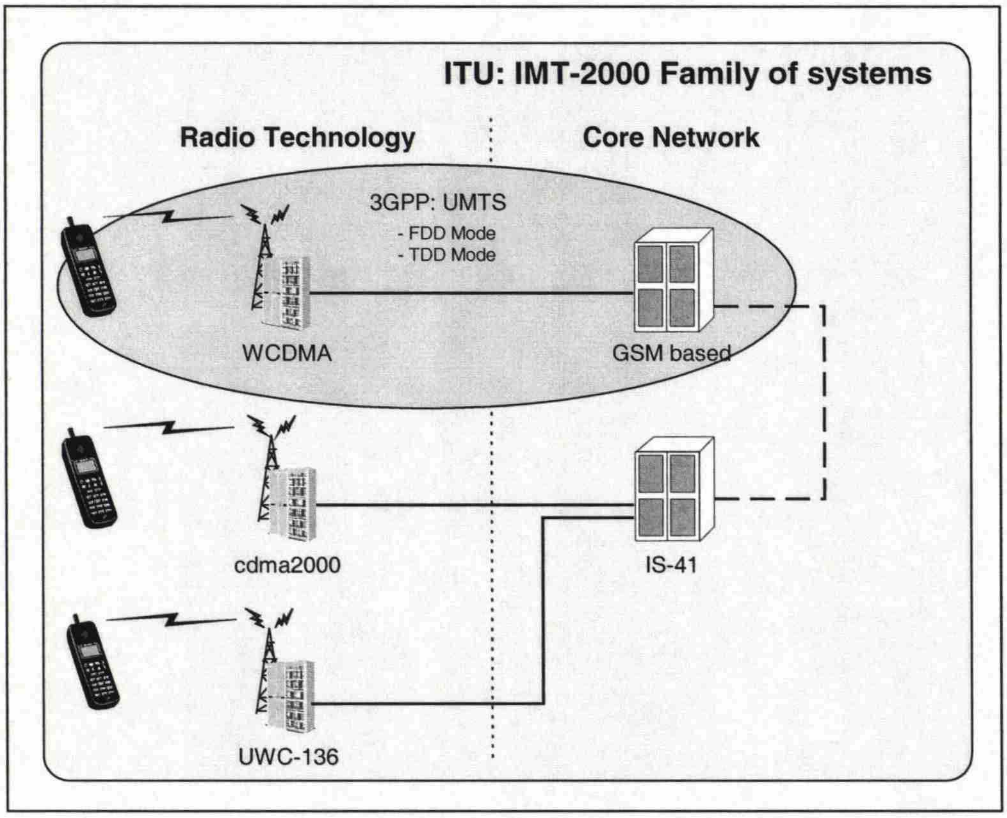

Figure 2. IMT-2000 Family of Standards framework

ITU's very active role in the beginning of 3G standardization work diminished in front of more business-driven regional bodies. ITU's original aim to prepare specifications for 3G had to be abandoned: organizations like 3GPP and 3GPP2 took the responsibility of preparing the technical specifications.

2.5 3GPP and GSM Evolution to UMTS

When it seemed clear there will not be a single global standard, standardization bodies promoting WCDMA decided to establish an organization for strengthening the position of WCDMA. The founding 3GPP partners included ETSI (Europe), ARIB (Japan), T1 (USA), TTA (*Telecommunications Technology Association*, Republic of Korea) and TTC (*Telecommunication Technology Committee*, Japan). The main objective of 3GPP is to make sure that there will be technical specifications for the UMTS with WCDMA radio and GSM-based core network technology. The first version of the specifications (so called Release 99) should be ready in December 1999.

At the moment GSM is the most widely used second generation technology in the world (45 % of total global subscribers [GM99b]). It seems obvious that the current GSM operators globally will adopt 3GPP's solution for 3G networks. The GSM community has mapped out a very clear evolutionary path to 3G. It starts with HSCSD *(High Speed Circuit Switched Data)*, moves on to GPRS *(General Packet Radio System)*, then to EDGE *(Enhanced Data Rates for GSM Evolution)* and ends up with UMTS. In Figure 3 this evolution path is presented.

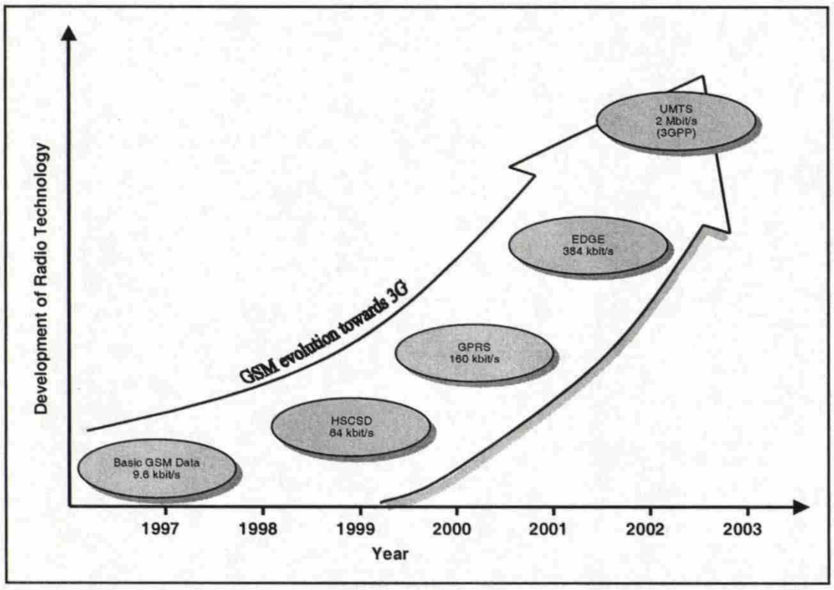

Figure 3. GSM evolution to UMTS.

HSCSD provides data transfer rates up to 64 kbit/s. Higher data transfer rates are achieved by using 14.4 kbit/s channel coding (instead of normal GSM 9.6 kbit/s) and by allocating multiple traffic channels per connection.

HSCSD offers, as its name implies, circuit-switched services. The true packet-switched services will be available in GSM when GPRS system is deployed. GPRS provides access to external packet data networks such as Internet. GPRS offers data transfer rates up to 160 kbit/s. It allocates resources dynamically in a packet-network style. The difference between packet- and circuit-switched services is explained in Chapter 3.1.2.1.

EDGE improves both HSCSD and GPRS further by using high-level modulation in the air interface. The maximum data rate EDGE can offer is 384 kbit/s.

The rest of this thesis will concentrate on the goal of GSM evolution, UMTS. All the technological solutions described in this paper are based on the specifications of 3GPP. The Release 99 of UMTS specifications should be available from 3GPP at the end of year 1999 but this thesis is written using the specification versions available in October 1999. The majority of the specifications are already in approved state (version 3.0.0, or higher) in October 1999, but some still require development. Few technical details might change from what is presented in this paper, but the main principles should remain unchanged.

2.6 CDMA Basics

Although UMTS is based on the evolution from GSM, the radio interface in UMTS is completely new. This chapter describes the basic principles of UMTS radio technology. Also the radio spectrum allocation for 3G systems is presented.

The radio interface is the most critical interface in the mobile communication systems. In other interfaces capacity is easily increased by adding more optical fiber or copper lines to the ground, but in the radio interface there is only a certain fixed amount of radio frequency band available. Furthermore, the radio band has to be divided between multiple users simultaneously. The methods used for dividing the common resources are called *multiple access methods*. The access method used in the UMTS radio interface is called Wideband Code Division Multiple Access (WCDMA). The basic difference between WCDMA and CDMA is that WCDMA uses wider frequency band as a carrier and is thus capable of carrying more user data. Otherwise the same principles apply for both WCDMA and CDMA.

2.6.1 General

CDMA is not a new technology. CDMA has been under study from 1950s but it was not accepted to cellular applications before 1980s. 3G networks will be the first commercial system with WCDMA radio technology.

The idea of CDMA differs from the other access methods. For example in FDMA (*Frequency Division Multiple Access*) the radio band is divided to several channels with unique frequency band: each user gets own channel and frequency band when transmitting data. In TDMA (*Time Division Multiple Access*) every user shares the same radio band but the radio band is allocated to each user only for a very short period of time called time slot: only one user uses the radio band at a time and the others wait for their turn. In CDMA, however, the whole radio band is allocated for every user all the time. Unique *spreading codes* are used for distinguishing different user's data from each other. In other words, in FDMA user is identified by frequency, in TDMA by time slot and in CDMA by spreading code. These basic principles of different access methods are illustrated in Figure 4.

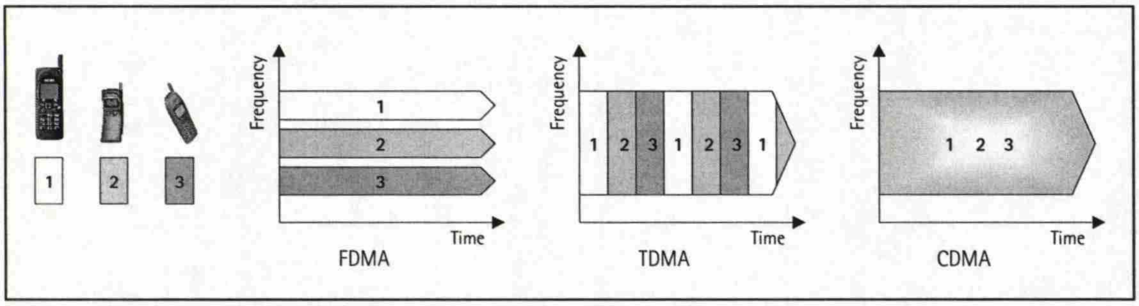

Figure 4. The basic principles of FDMA, TDMA, and CDMA.

2.6.2 Spreading Code

The spreading code is unique for every user. A new spreading code is allocated every time new radio resources are reserved. The spreading code is valid until the user releases the radio resources. The spreading code is "added" to all the information user sends. This way it is possible to distinguish the user's data in the air interface from the other user's data. The idea of spreading code usage can be compared to a restaurant full of people who are all simultaneously in conversation with different language. First everything sounds like confusing noise but after a short while you can distinguish discussion done with the language you understand.

2.6.3 Macrodiversity

In CDMA systems all base stations (BS) and mobile stations (mobile phones) are using the same frequency. That enables mobile phones to be in contact to more than one base station at the same time. This property is called *macrodiversity*. In the Figure 5 macrodiversity situation is illustrated. The mobile station is connected to two base stations, BS 1 and BS 2. Both base stations are relaying the user traffic between the mobile station and the rest of the UMTS network. The user traffic is depicted with thicker lines in the figure. Macrodiversity improves the transmission quality because the same data is transmitted via several routes. Macrodiversity has also an effect to the UMTS network architecture.

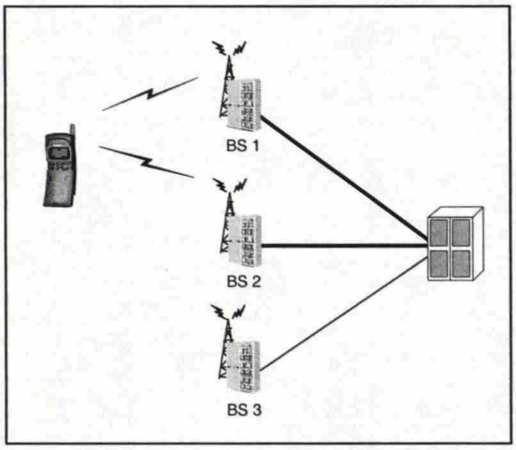

Figure 5. Macrodiversity with two base stations.

2.6.4 Radio Frequency Allocation

One of the most important issues in the third generation mobile communication system development work has been the allocation of the needed radio frequency bands. After years of negotiations, the agreement was made in WARC-92 meeting *(World Administrative Radio Conference)* in 1992. The conference identified the 1885 – 2025 MHz and 2110 – 2200 MHz bands as available on a worldwide basis for 3G, totaling 230 MHz. Within this allocation, the 1980 – 2010 MHz and 2170 – 2200 MHz bands were identified for the satellite component, leaving the remainder for terrestrial service [3GM99a].

These frequency bands will be allocated in different ways in different regions and countries. Figure 6 depicts the IMT-2000 frequency allocation in the Europe, Japan, and the USA. In the USA, a part of the IMT-2000 frequency allocation is already used for the second generation mobile PCS *(Personal Communication Services)* systems. Thus, third generation systems have to fit into the current PCS frequency allocations. In Japan and Europe the frequency allocation for 3G is almost identical.

In Finland the licenses for the UMTS operators have already been granted. The four operators – Oy Radiolinja Ab, Sonera Oy, Suomen Kolmegee Oy, and Telia Mobile – will each get one 15 MHz frequency band between 1920 – 1980 MHz (uplink) and the other 15 MHz band from 2110 – 2170 MHz (downlink). The exact frequencies for each operator will be specified later.

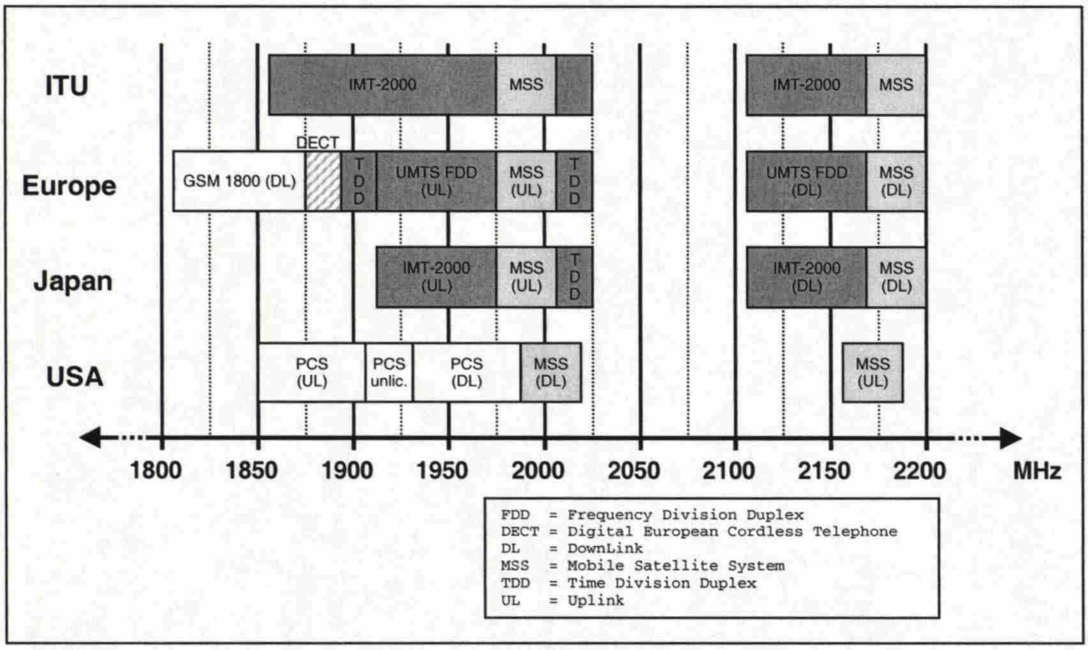

Figure 6. Frequency allocation for the third generation mobile communication.

3 UMTS SYSTEM

This chapter shortly describes the basic features of UMTS system. The services offered by UMTS network are covered first. The network architecture and protocol stacks in different interfaces are introduced.

3.1 Services

The development and standardization work of UMTS is based on GSM. The basic services standardized in GSM will be available in UMTS too. One of the main principles behind the development work of UMTS is to allow a flexible introduction of new services to the system. This gives a possibility to design new services when they are really needed. In the following the most important services are shortly presented. These services are standardized in UMTS specifications.

3.1.1 Teleservices and Supplementary Services

The most used service at the moment in cellular networks is the basic telephony. This service enables bi-directional speech calls to be established between UMTS users and subscribers in any other telephony network. The main requirement for the speech service in UMTS is that the quality of speech transferred across the network is at least as good as in existing 2G systems. Also *Short Message Service* (SMS) is included in UMTS teleservices [3GPP99a].

The *supplementary services* in UMTS are adopted directly from GSM [3GPP99a]. They modify and enrich the basic services by allowing the user to choose how his or her calls are treated by the network. The supplementary services are not specific to UMTS, most of them are directly inherited from the fixed networks and GSM. Call waiting and call forwarding can be mentioned as examples of these services. The complete list of supplementary services can be found for example in [MoPa92].

3.1.2 Data Services

Originally mobile communication systems have been designed to transfer voice. The growing popularity of the Internet increased the pressure on the development of the mobile systems towards more data-oriented solutions. For example GSM is still lacking pure packet-switched services: data services are offered to mobile users as a circuit-switched service. When GPRS will be launched in 2000, also the packet services will be available for GSM users. In UMTS both circuit- and packet-switched data services are offered from the beginning as an integral part of the system.

3.1.2.1 Definition of Circuit-Switched and Packet-Switched Services

The fundamental difference between circuit-switched (CS) and packet-switched (PS) service is that for the circuit-switched service there are always certain resources reserved for the duration of the service. Another characteristic of the circuit-switched is that the capacity of the reserved resources remains the same during the service. A normal speech call is a typical circuit-switched service: for the duration of the call there is a specific line, with certain capacity, reserved for that user only. Even if the user does not speak for hours, the resources will be kept reserved.

Packet-switched services are based on the idea that the resources are used only when there is something to send. The other users share the same network and the same resources. Sending data requires resources only when actually sending something to the network. When the data transfer has ended, the resources are free for the other users. An example of a network offering packet-switched services is the Internet.

3.1.2.2 Circuit-Switched Data Services

Circuit-switched data services offer modem-like connections. Circuit-switched services can guarantee certain bit rates for the users. That is why CS data services are used for realizing real-time services that require assured bit rates during the duration of the service. For example video call will be implemented with circuit-switched services in UMTS. In the first phase UMTS will support at least 64 kbit/s for circuit-switched data services. [3GPP99a]

3.1.2.3 Packet-Switched Data Services

Packet-switched data services are provided for interworking with packet networks such as the Internet. In addition to that, packet-switched data serves as a versatile platform for the future service development. In the following table offered packet-switched data rates are summarized. Note that the data transfer rates are depending both on the speed and the current location of the terminal (mobile phone). Higher data rates will be offered only in the area where it is economically feasible: the best services are provided in areas where most of the paying customers are. The highest bit rate of 2 Mbit/s can be accessed only in city areas with very slow moving terminal.

Table 1. Data transfer rates in UMTS. [OjPr98]

Environment	indoor, urban	urban	rural outdoor
Mobility	low	medium	high
Terminal speed	< 10 km / h	< 120 km / h	< 500 km / h
Offered bit rate	2 Mbit/s	384 kbit/s	144 kbit/s

3.2 Basic Architecture

UMTS network architecture is based on GSM design principles: GSM was designed so that the intelligence of the system can be distributed throughout the network [AhKa99]. The same idea is transferred to 3G. The decentralized intelligence of the UMTS network is implemented by dividing the network into separate entities: the UTRAN (*UMTS Terrestrial Radio Access Network*) and CN (*Core Network*). Figure 7 illustrates this functional division.

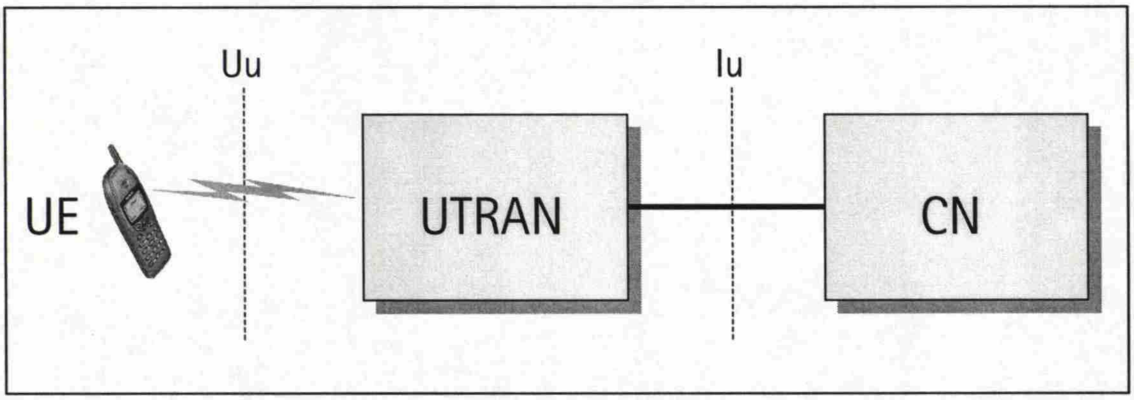

Figure 7. UMTS architecture. [3GPP99b]

The UTRAN is a network part responsible for radio path control. Mobile phone is called *User Equipment* (UE) in UMTS. The UE can use the UMTS network services by accessing the UTRAN. The CN controls the network services, like establishment and clearing of the calls.

In the figure there are also two interfaces presented, Uu and Iu. Both are standardized and fully open interfaces. Uu is the radio interface used for connecting UEs to UMTS network. Physically Uu is implemented with WCDMA technology. The UMTS radio interface basics were discussed in Chapter 2.6. The UTRAN and CN are connected to each other via the Iu interface.

One of the fundamental concepts of UMTS is the separation of the UTRAN functionality from the CN functionality. The idea is that the UTRAN provides a core-network-technology-independent access platform for UEs to all core network services. In principle, the core network technology could be changed without modifications to the UTRAN and vice versa. This allows independent development of both UTRAN and CN technology. The design of the Iu interface has to support the independent evolution of the UTRAN and CN. The Iu interface and its role in UMTS network are discussed in more detail later in this thesis.

3.3 Network Architecture

The overall architecture of UMTS in Figure 7 shows high-level functional division of UMTS network. More detailed UMTS architecture with different network elements is illustrated in Figure 8. The architecture is based on the network elements used in GSM and GPRS.

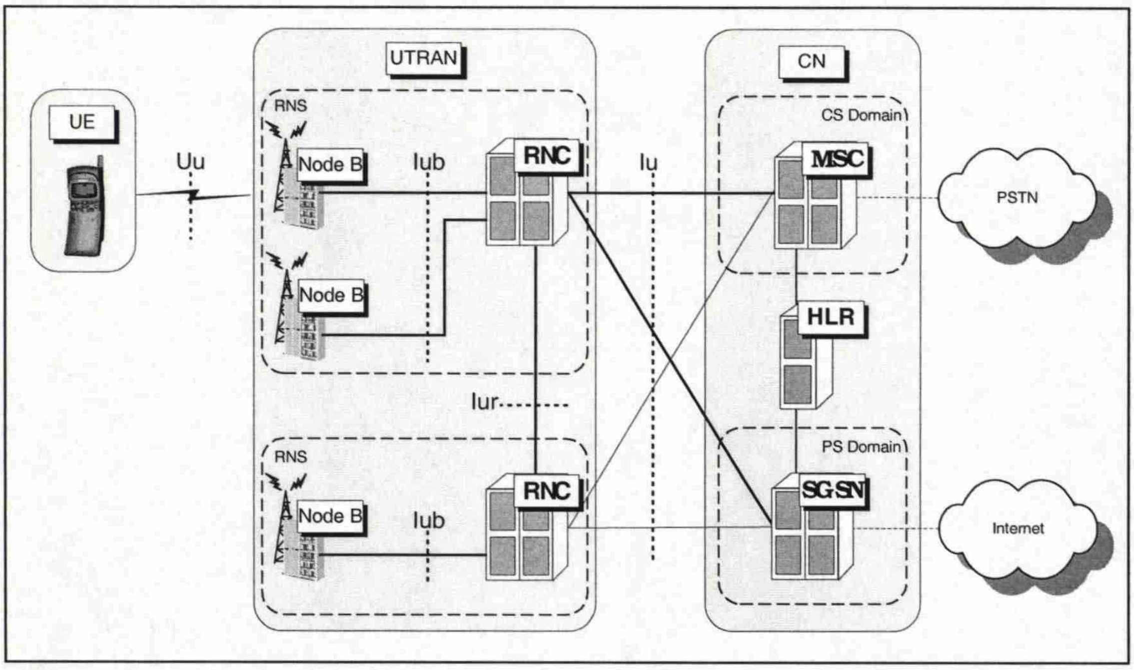

Figure 8. UMTS network elements. [3GPP99c]

3.3.1 UMTS Terrestrial Radio Access Network

The main function of the UTRAN is to provide for users access to the services of the UMTS network. The UTRAN takes care of all the radio interface specific functions. The UTRAN consists of two network elements, *Node B* and *Radio Network Controller* (RNC). In other mobile communications systems the network element corresponding to the Node B is traditionally called base station. The UTRAN is divided to one or several *Radio Network Subsystems* (RNS). The RNS is a logical entity that consists of exactly one RNC and one or several Node Bs.

UMTS is a cellular system. The geographical radio coverage is implemented with a number of Node Bs. UEs connect to the services of UMTS network through the Node B. The Node B takes care of maintaining the air interface, and relays voice and data traffic to and from the UE.

Radio Network Controller is the central network element of the UTRAN. The RNC corresponds to the BSC (*Base Station Controller*) in GSM. The RNC controls the radio access network and performs radio resource management. For example, the RNC knows the free resources of each Node B it is connected to: if a UE wants to reserve resources from the radio interface, the RNC will decide whether the UE gets the resources or not.

Inside the UTRAN there are two interfaces: Iub and Iur. The Node Bs and RNC are connected together with Iub. It will be standardized only partially: Iub will not be truly open interface. Iur is a completely new interface compared to GSM. It connects two RNCs together. The Iur interface is introduced to UMTS network to support macrodiversity between two or more RNCs (see Chapter 2.6 for CDMA basics). In the case of macrodiversity with two RNCs the user data is relayed between the two RNCs via the Iur interface. Both Iur and Iub interfaces are out of the scope of this thesis, they are discussed only generally.

3.3.2 Core Network

The core network (CN) is responsible for connecting, controlling and releasing the services that user requests. The CN provides also the means for interworking with external networks such as the Internet and PSTN (*Public Switched Telephone Network*). The design of the UMTS core network is based on GSM. The UMTS CN network elements are derived directly from GSM and their functionality is basically the same in both systems.

The UMTS core network is divided into two logical, independent domains:

- the Circuit Switched Service Domain (CS domain)
- the Packet Switched Service Domain (PS domain)

CS domain offers the basic circuit-switched services (speech and circuit-switched data) and PS domain provides the packet-switched data services.

The split of the CN to two logical domains is reflected in the CN network elements too: *Mobile Switching Center* (MSC) implements the CS domain services and *Serving GPRS Support Node* (SGSN) PS domain services. The MSC and SGSN are the most complex network entities in the UMTS. Their main function is to coordinate the setting-up of calls and packet data contexts to and from UMTS users. The specification of UMTS does not require that the MSC and SGSN are implemented as a separate, physical devices. The functionality of the MSC and SGSN can be combined to a single unit, if that is required.

Home Location Register (HLR) is the place where all the subscriber permanent information is stored. Both the MSC and SGSN use the information of the HLR, for example, to keep track of the location of the subscribers. Also a list of the services user is allowed to use is stored in HLR.

A similar register, called *Visitor Location Register* (VLR), is integrated to the MSC. VLR is not shown in Figure 8, but it is located inside the MSC. VLR maintains same kind of information as HLR, but the information in VLR is always related only to subscribers that are currently in the service area of the MSC. When the subscriber leaves the service area of the MSC, subscriber related information is removed from VLR. The SGSN does not use the services of the VLR.

There are also other CN network elements that are not showed in Figure 8: the network elements are omitted because they are not essential when discussing UMTS system from the Iu interface and bearer management perspective.

Both the MSC and SGSN are connected to the UTRAN via the Iu interface. The same interface should support as well circuit- as packet-switched traffic, which are very different types of traffic by nature. Each RNC is connected to exactly one MSC and/or SGSN, but MSCs and SGSNs can be connected to several RNCs.

3.3.3 User Equipment

User Equipment (UE) is the terminal equipment the subscriber uses to connect to the services of the UMTS network. The UE can move freely in the area of the UTRAN radio coverage without break in the services. The UMTS UE will offer for the users the basic speech services and wide variety of data services. The most visible change in the UMTS UE compared to the existing 2G mobile phones will be the bigger screen: the new UMTS services will rely more on vision than hearing.

3.4 Protocol Stacks in UMTS

Protocols are a set of definitions that specify how different network elements communicate in a well-defined manner over an interface. The functions of a network element are realized by dividing the functions to smaller sub-functions called layers: each layer takes care of a certain task in that network element. In addition to that lower layers provide services for upper layers. When the protocol layers are combined they form protocol stacks. The whole protocol stack is needed to implement the required functions of the network element. UMTS control plane protocol stacks in Iu, Iub and Uu interfaces are presented below.

3.4.1 Control Plane and User Plane

In UMTS protocols are divided to two different classes: to control plane protocols and user plane protocols. The division is made to distinguish different functionality from each other. The control plane is used for transferring control information between network elements and to control the establishment, modification and release of user plane connections. The user plane is used for carrying the actual user data, such as speech in the case of voice call.

3.4.2 Protocol Stacks between UE and CN

The control plane protocol stacks in Iu, Iub and Uu interfaces are illustrated in Figure 9. Note that the control plane is the same for both circuit- and packet-switched services. The protocol layers (marked as L1 to L3) relation to OSI (*Open System Interconnection*) reference model is also shown on right side of the figure. Each protocol is described briefly and the main functions of the protocols are presented. Transport layers shown in the figure provide reliable transmission path for the upper layer protocols. The characteristics of the radio interface (WCDMA L1) were discussed in Chapter 2.6.

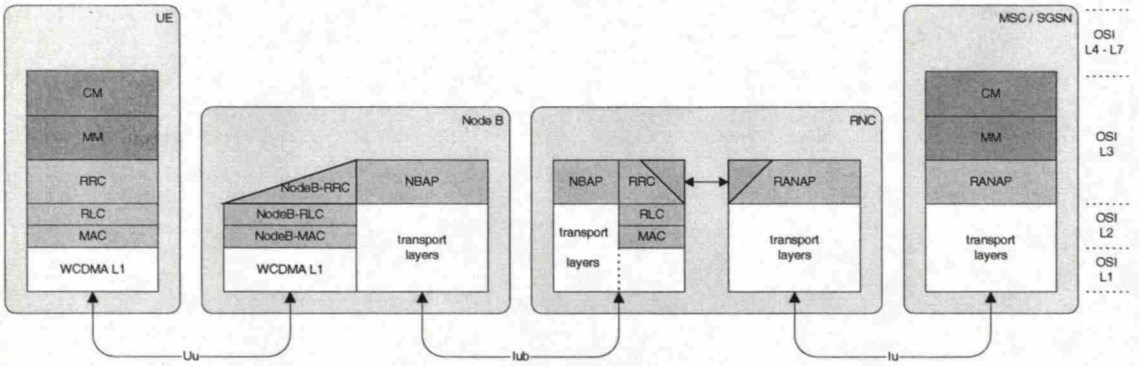

Figure 9. Control plane protocol stacks between UE and CN.

3.4.2.1 CM

CM layer (*Connection Management*) is responsible for initiating connection establishment and release between the UE and CN. Connection management controls the use of transmission resources and it participates in bearer management procedures too. CM layer consists of a few protocols: for example *Call Control* (CC) protocol is responsible for establishing, and releasing the calls in the circuit-switched domain while *Session Management* (SM) protocol takes care of corresponding responsibilities in the packet-switched domain.

3.4.2.2 MM

Mobility Management (MM) protocol provides the means to manage the mobility of the subscribers in both circuit- and packet-switched domains. The main function of MM protocol is to keep track of the location of the subscriber. Mobility Management also takes care of the identification of the user equipment and the subscriber. Another important responsibility of MM protocol is to perform the security functions between the UE and CN.

3.4.2.3 RANAP

Radio Access Network Application Part (RANAP) is responsible for managing transport resources in the Iu interface. RANAP setups and maintains the control plane connection from the CN to the RNC and provides the service to transport upper layer (MM and

CM) messages over the Iu interface. RANAP takes care of establishment and release of the user plane connections over the Iu interface upon request from CM. The main task of RANAP is to provide the means for bearer management: RANAP transfers bearer establishment, modification and release commands from the CN to the RNC and initiates bearer management procedures concerning the Iu interface. The RANAP protocol is described in detail in Chapter 5.

3.4.2.4 RRC

Radio Resource Control (RRC) protocol is in logical level similar to RANAP: RRC takes care of establishing and maintaining control plane connections (called RRC connections) between the RNC and UE and setups user plane connections when CM layer requests it. The difference to RANAP is, however, that RRC works on the radio interface between the UE and RNC. RRC participates bearer management by controlling the radio interface bearers. Together RANAP and RRC provide end-to-end control plane connection for the upper layers (MM and CM): RRC connection transports upper layer messages from the UE to the RNC and RANAP relays the messages from the RNC to the CN over the Iu interface.

3.4.2.5 NBAP

Node B Application Part (NBAP) protocol is used for controlling the Node B. NBAP transfers the radio interface management information from the RNC to the Node B. NBAP also establishes and releases both control and user plane connections over the Iub interface.

3.4.2.6 RLC and MAC

Radio Link Control (RLC) and *Medium Access Control* (MAC) are both layer two protocols. Together they provide reliable data transfer from the RNC to the UE. RLC takes care of flow control and asks re-transmission if a data packet is not received correctly. The main responsibility of MAC is to handle the access to the physical layer. RLC and MAC differ from the other control plane protocols, RLC and MAC are used in both control and user plane.

4 BEARER MANAGEMENT IN UMTS

UMTS offers wide variety of services. Each service has different kind of characteristics and requirements concerning the transmission path. In GSM all the services were realized by using the same type of transmission channel capable of transferring 9.6 kbit/s of user data. This kind of approach is not feasible in the UMTS. The data speeds that the UMTS services need can vary drastically: for example speech can be transported in a transmission path capable of transferring 10 kbit/s but data services can require up to 2 Mbit/s of transmission capacity. To support the different requirements of the services a bearer concept was introduced to UMTS.

In UMTS bearers are used for providing the needed data transport capacity and to fulfill the *Quality of Service* (QoS) requirements of each service. Every connection between UMTS network elements is implemented with bearers, data cannot be sent across the network without the services of the bearers. Basically, bearer is a logical connection that represents data transfer capabilities through the network. These capabilities are characterized with a set of attributes.

Bearers can be flexibly set up and modified on demand. The functions that control the establishment and releasing of the bearers are called bearer management. The bearer management functionality is distributed to different UMTS network elements. This chapter describes how they interact with each other to offer flexible and optimized way of data transfer across the UMTS network.

First the user requirements and the QoS in UMTS are discussed. After introducing the basic ideas of the bearers and the bearer service architecture, bearer management functions are covered. In the last part of the chapter some example bearer management procedures are presented. This chapter is based on mainly 3GPP specification [3GPP99d], but some aspects are taken from the specifications [3GPP99e], [3GPP99f], [3GPP99g], [3GPP99h], and [3GPP99i].

4.1 UMTS Quality of Service

All the different kinds of services UMTS offers have one commonality: the quality of each service has to be good enough in order that the users are willing to pay for them. It is the end-user who decides whether he is satisfied with the QoS or not. The UMTS system has to be designed so that the system fulfills the requirements of the users. In Figure 10 the principle of providing QoS to UMTS users is illustrated.

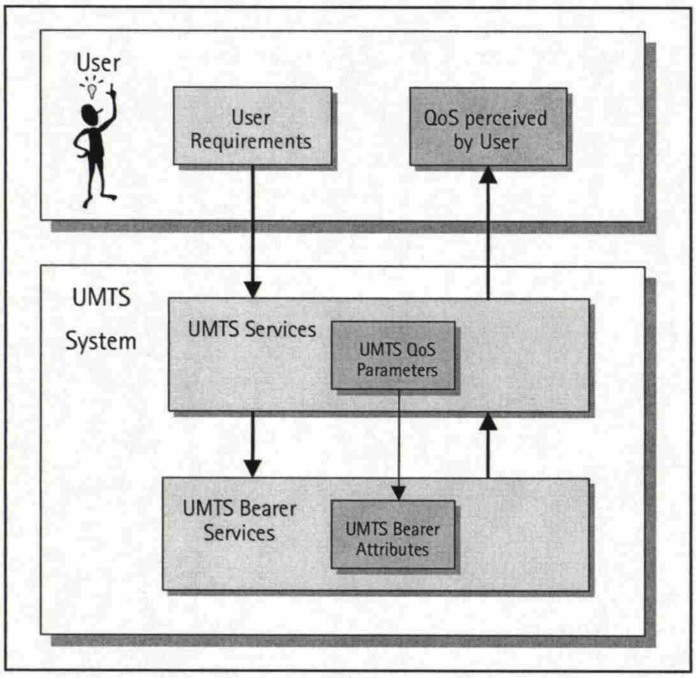

Figure 10. Providing UMTS Quality of Service to the user.

The user requirements are the basis for the design of UMTS services. Each UMTS service is realized by using the UMTS bearer services. To meet the user requirements UMTS services have requirements of their own for the UMTS bearer services. Ultimately it is the bearer services that fulfill the user requirements by providing efficient means for transferring data across the UMTS network.

The end-users' sense of the QoS is based on the personal opinion and feeling. However, inside UMTS system QoS has to be defined with an exact set of parameters, such as bitrate and transfer delay. These parameters have to correspond the end-user's expectations of the QoS as well as possible. Each UMTS service has its own characteristics: these characteristics are mapped to certain values of UMTS QoS parameters. From the user's viewpoint the mapping of QoS parameters has to be automatic, the user should only choose the service and leave the rest to the UMTS system. The UMTS QoS parameters are further mapped to the actual values of UMTS bearer attributes, as depicted in the figure. The bearer attributes define the properties of the bearer exactly. The bearer together with the implementation of the service realizes the QoS that the user perceives.

4.1.1 UMTS Quality of Service Classes

In UMTS there are four different QoS classes, also often called traffic classes. The QoS classes are used for optimizing the usage of the resources. Each UMTS service is categorized into one of the four classes depending on the type of the traffic the service creates. In the Table 2 UMTS QoS classes are presented.

Table 2. UMTS QoS classes [3GPP99d].

QoS Class	Characteristics	Example service
Conversational class	• Preserve time relation between information entities of the stream • Conversational pattern	Voice
Streaming class	• Preserve time relation between information entities of the stream	Streaming video
Interactive class	• Request-response pattern • Preserve payload content	WWW browsing
Background class	• Destination is not expecting the data within a certain time • Preserve payload content	Background download of E-mails

The main difference between the classes is how delay-sensitive the traffic is: Conversational class is meant for the traffic which is very delay sensitive while Background class is the most delay-insensitive traffic class.

Conversational and Streaming classes are mainly intended to be used for carrying real-time traffic flows. Conversational real-time services, like video or speech telephony, are the most delay-sensitive applications and those data streams should be transported in the most reliable traffic class. Streaming class is used with the services that produce steady flow of data in one direction. For example real-time video or audio could be sent from the service provider to the subscriber using Streaming class.

Interactive and Background classes are designed to be used by traditional Internet applications like WWW browsing, E-mail, Telnet, and file transfer. Interactive class is used by interactive applications, such as WWW browsing, that need to react to user input within a reasonable time. Background class is meant for background-type of traffic, eg for downloading E-mails from the mail server or for uploading files. The traffic of Background class has lower priority than the other classes' traffic and the background traffic is sent across the network only when the other classes do not need the transmission resources.

4.2 Bearer Service architecture

Bearer services are provided between UMTS network elements. Each interface in the network has its own characteristics. The bearer services on an interface have to take account the specialties of that specific interface. However, these specialties are not wanted to be seen to the users of the bearer services, like UMTS services (refer to Figure 10). The bearer service architecture is based on the idea that the bearer services on each interface are encapsulated in a higher layer bearer service that hides the details of the lower layers. The UMTS bearer service users then utilize only the services of the higher layer bearer service. The hierarchical bearer service architecture of UMTS is depicted in Figure 11.

In the figure there are few new entities: CN Gateway, TE, and MT. *CN gateway* is an UMTS network element that provides access to external networks such as PSTN and the Internet. *Mobile Termination* (MT) is a part of the User Equipment (UE): MT provides for the UE access to the UMTS network and terminates the radio path in the UE. *Terminal Equipment* (TE) refers to the device that implements the actual user application. TE can be, for example, a laptop connected to a UE.

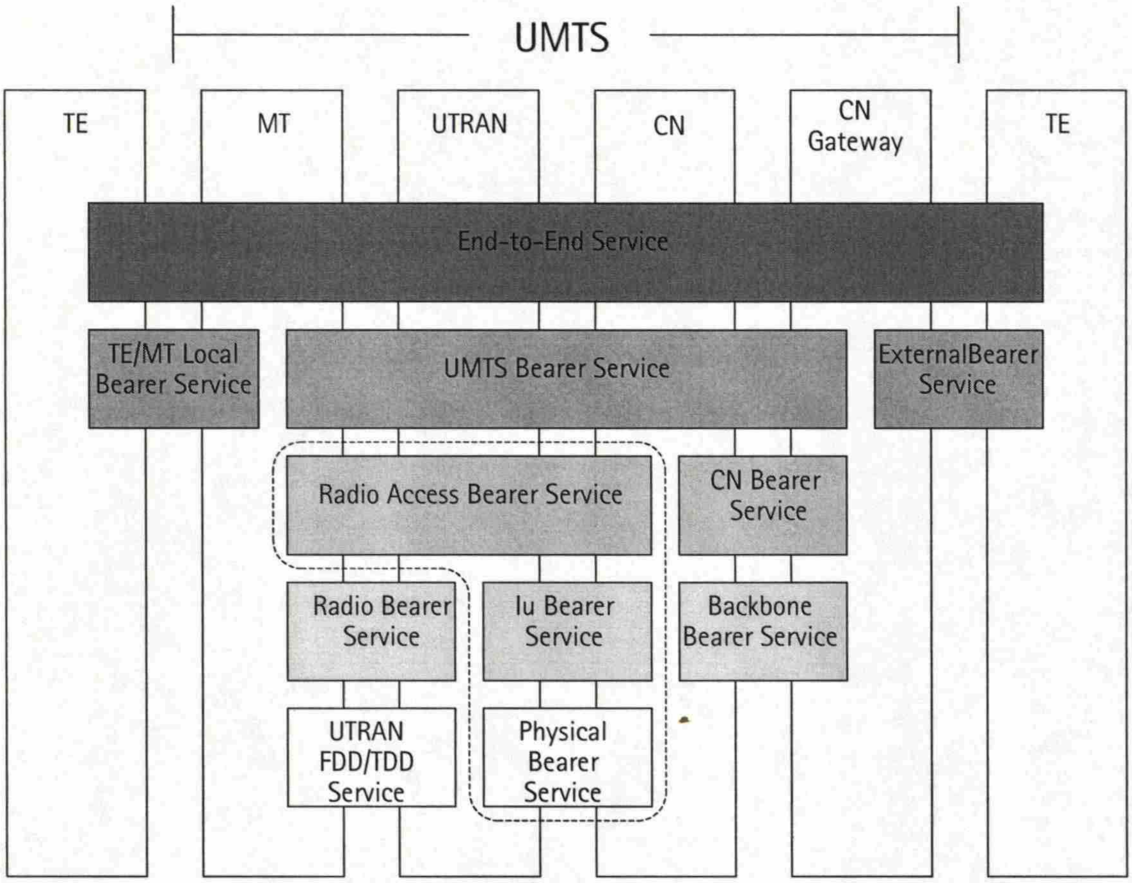

Figure 11. UMTS layered bearer service architecture.

The *End-to-End Service* is the highest layer in UMTS bearer service architecture. The End-to-End Service is the service the end-user sees. However, End-to-End Service is only a logical connection: the physical connection from TE to TE is implemented with the services of the lower layers. End-to-End Service used by the TE is realized by using a *TE/MT Local Bearer Service*, a *UMTS Bearer Service* and an *External Bearer Service.* The TE/MT Local Bearer Service is used for connecting, for example, a laptop to UE. The External Bearer Service is needed to provide the connection from UMTS to external networks, such as the Internet. Both the TE/MT Local Bearer Service and the External Bearer Service are not truly UMTS services and they are not discussed further.

The UMTS Bearer Service is the service UMTS network provides for the user applications. Like End-to-End Service also UMTS Bearer Service has only logical meaning: it hides the details of the physical implementation of the bearers in different interfaces. UMTS Bearer Service consists of two parts, the *Core Network Bearer Service* and the *Radio Access Bearer Service*. The Core Network Bearer Service is used for connecting different UMTS core network elements to each other, and it is physically implemented with the *Backbone Bearer Service*.

The Radio Access Bearer Service provides reliable and confidential information transfer between the CN and UE. This service is based on the characteristics of the radio interface and it is maintained for a moving UE. The Radio Access Bearer Service is realized by a *Radio Bearer Service* and an *Iu Bearer Service*.

Just like bearer services also bearers have hierarchical layered structure. A connection between the CN and UE, or a *Radio Access Bearer* (RAB), consists of exactly one *Radio Bearer* and one *Iu Bearer*: when establishing a connection from the UE to the CN both Radio Bearer and Iu Bearer have to be setup before the Radio Access Bearer is ready to be used.

The *UTRAN FDD/TDD Service* and *Physical Bearer Service* present how the bearer services are physically implemented in each interface. The UTRAN FDD/TDD Service is implemented with WCDMA technology either in FDD or in TDD mode. The Physical Bearer Service uses ATM (*Asynchronous Transfer Mode*) as a transportation medium.

The area encircled with the dashed line in Figure 11 will be under discussion for the rest of this thesis: Radio Access Bearer Service and especially Iu Bearer Service with its physical implementation are covered in the most detail.

4.3 Bearer Types

Generally the bearers UMTS network offers are generic: the bearers are used for carrying any type of information between network elements. However, bearers can be categorized according to their purpose of use. In UMTS there are two high-level types of bearers, *signaling bearer* and *data bearer*. Both types of bearers are implemented using the UMTS bearer services (see Figure 11). The type of the bearer is determined when the bearer is established. The type remains the same throughout the lifetime of the bearer until the bearer is released.

4.3.1 Signaling Bearer

Signaling bearers are used for carrying *signaling* information between the UE and CN. Generally, signaling means all the control information relayed between the network elements. Signaling bearers are a part of the *signaling connection*. A signaling connection is established between the UE and CN when the UE needs to send some control information to the CN or vice versa. The signaling connection is a logical connection, it is realized by signaling bearers in each interface from the UE to the CN: in the Uu interface with Radio Signaling Bearer and in the Iu interface with Iu Signaling Bearer. The signaling bearers are special types of bearers, they always have the same fixed parameters.

Signaling connection and signaling bearers are specific for each user: no one else is using the same signaling connection or signaling bearers to transport the control information between the UE and CN. Users cannot have more than one signaling connection, all the signaling information related to the user is transferred with the same signaling connection. The signaling connection is also called control plane connection. The functions of the control plane were described in Chapter 3.4.1.

4.3.2 Data Bearer

Data bearers are used for transferring the actual user data between the UE and CN. Data bearers are the bearers that provide QoS for the end-users. Data bearers can be further divided to two sub-types: a *circuit-switched data bearer* (CS data bearer) and a *packet-switched data bearer* (PS data bearer). The CS data bearer is used for carrying CS type user data, like voice or video call, and the PS data bearer carries PS user data, such as WWW browser data. UMTS services are normally implemented with one data bearer. Unlike signaling bearers, data bearers can have wide range of attribute values. The bearer attributes are discussed in Chapter 4.3.4.

4.3.3 Signaling and Data Bearers in an Active Call

Both signaling and data bearers are closely related to the different protocol planes in UMTS: signaling bearers are used for carrying the messages of the control plane protocols and data bearers are a part of the user plane that transports the user data across the network. In Figure 12 this relationship is depicted. The figure shows the Iu interface where RANAP is the control plane protocol. The bearers in the figure are provided by the Iu Bearer Service (refer to Figure 11). The same principles apply to every interface, only the name of the protocol and the bearer services change.

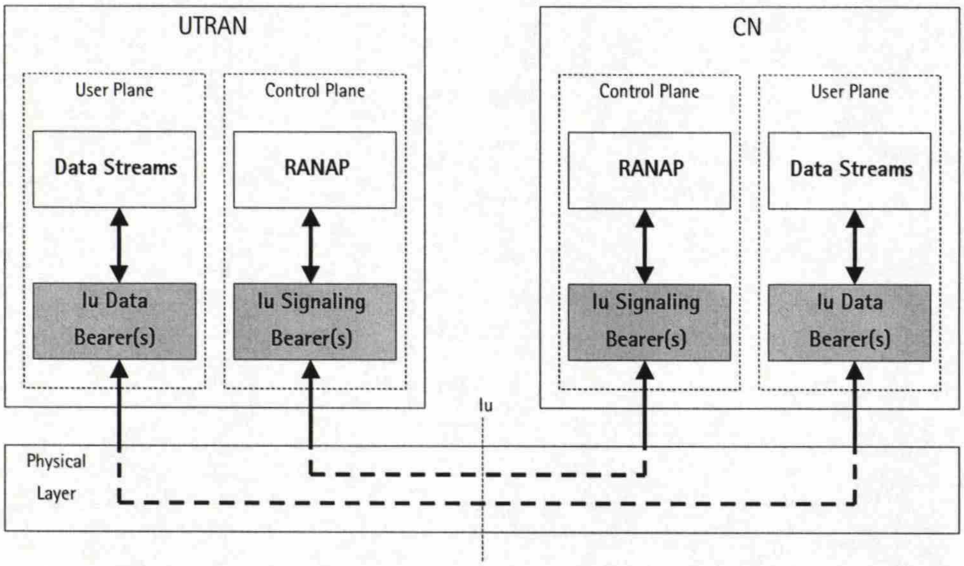

Figure 12. Signaling and data bearers' relation to control and user plane.

The signaling connection (control plane connection) can be stand-alone but the user plane connection is always associated to a signaling connection. When a subscriber wants to make a call, either a CS or PS type of call, a signaling connection with related signaling bearers in each interface is established first. The established signaling connection is used to transmit information about the type of the desired call to all relevant network elements. When the network elements have the required information, a user plane connection can be established between the UE and CN. The setup of a user plane connection means essentially establishing data bearers in every affected interface.

An active call includes one signaling connection and one user plane connection. In the case of multicall (see Chapter 2.3.3) each active call can have a data bearer of its own. So the user might have at the same one signaling and several user plane connections established with the associated bearers. However, in a multicall data bearers can be shared too: for example, two simultaneous services could use the same data bearers for the user data transportation.

When the subscriber decides to disconnect the call, signaling connection is used for transferring the control messages to release the data bearers reserved for the user plane. When the user plane and the data bearers are cleared, also the signaling connection with the signaling bearers is released.

4.3.4 Bearer Attributes

The bearer attributes define the bearer characteristics exactly. The bearer attributes are derived from the QoS parameters of the UMTS service. First UMTS QoS parameters are mapped to UMTS Bearer attributes and then, according to the layered bearer service architecture, further to Radio Access Bearer attributes and finally to Radio Bearer and Iu Bearer attributes. Basically the set of attributes is the same for every different layer bearer: the attributes of UMTS Bearer can be mapped quite straightforwardly to Radio and Iu Bearer attributes. However, there are some attributes that cannot be mapped directly between the different layers. If this is the case it is mentioned in the description of the bearer attribute.

The bearer attributes are defined when the bearer is established. The values of the bearer attributes can be modified also during the lifetime of the bearer. The list of bearer attributes is quite long, only the most important attributes are presented here. The complete list can be found from [3GPP99d].

Traffic class tells the type of application for which the bearer will be established. Traffic class is derived directly from the UMTS QoS Class of that service (see Chapter 4.1.1). By including the traffic class itself as an attribute UMTS can make assumptions about the traffic source and optimize the transport for that traffic type. For example, if the traffic class has the value 'Conversational class' the service could be implemented with CS bearers.

Maximum bitrate tells how much transport capacity the service requires from the bearer. The value is given as kbit/s and the highest value for the attribute is 2 048 kbit/s.

Guaranteed bitrate is the lowest bitrate when the service is still applicable. If the bitrate falls under the level of guaranteed bitrate at some point of the duration of the service, the QoS of the service will start to degrade. Guaranteed bitrate attribute can be used when the bitrate is allowed to change but never fall under the value of the attribute during the service. When used together with maximum bitrate attribute, maximum bitrate can define the upper limit and guaranteed bitrate the lower limit for the bearer.

Bit error rate (BER) indicates how many bits can be transferred erroneously compared to non-erroneously transferred bits. Typical value is, for example, 10^{-4} which means that every 10 000th bit is allowed to be sent erroneously. If the BER is much higher during the service than the requested value, QoS will not be what the user asked. The BER attribute cannot be mapped directly between different layers: the value has to be smaller in the Iu Bearer than in the Radio Access Bearer since the Radio Access Bearer BER includes also the BER of the Radio Bearer.

Transfer delay defines the total transfer delay of the bearer. This attribute specifies the delay tolerated by the application. The transfer delay is expressed in milliseconds. The value of the transfer delay attribute has to be larger over several interfaces than over only one interface: Iu Bearer transfer delay is smaller than Radio Access Bearer transfer delay.

Note that all the parameters are not applicable for every traffic class. Maximum bitrate and bit error rate are used with all four traffic classes but guaranteed bitrate and transfer delay have only meaning with the Conversational and Streaming class. In the following table there are two examples of mapping bearer attributes from service the QoS parameters.

Table 3. Example bearer attributes for speech and video stream services.

Bearer attribute	**Speech service**	**MPEG-4 Video stream**
Traffic class	Conversational	Streaming
Maximum bitrate	12.2 kbit/s	128 kbit/s
Guaranteed bitrate	-	24 kbit/s
Bit error rate	10^{-4}	10^{-6} no visible degradation, 10^{-5} little visible degradation, 10^{-4} some visible artifacts
Transfer delay	100 ms	150…400 ms

4.4 UMTS Bearer Management Functions

UMTS bearer management has two main functions: control the establishment, modification and release of the bearers dynamically when needed and maintain the QoS of the active bearers. The first function is the responsibility of the control plane but the second one is implemented in the user plane.

4.4.1 Bearer Management Functions in Control Plane

The control plane bearer management functions are distributed to different network elements to support the layered bearer service architecture. The basic idea is that the CN controls the bearer management. The UE requests the CN to provide the needed UMTS Bearer services. The UTRAN is responsible for providing Radio Access Bearer services when the CN asks it. Figure 13 illustrates the bearer management function distribution. The figure presents the establishment of data bearer, so the signaling connection with signaling bearers is already setup.

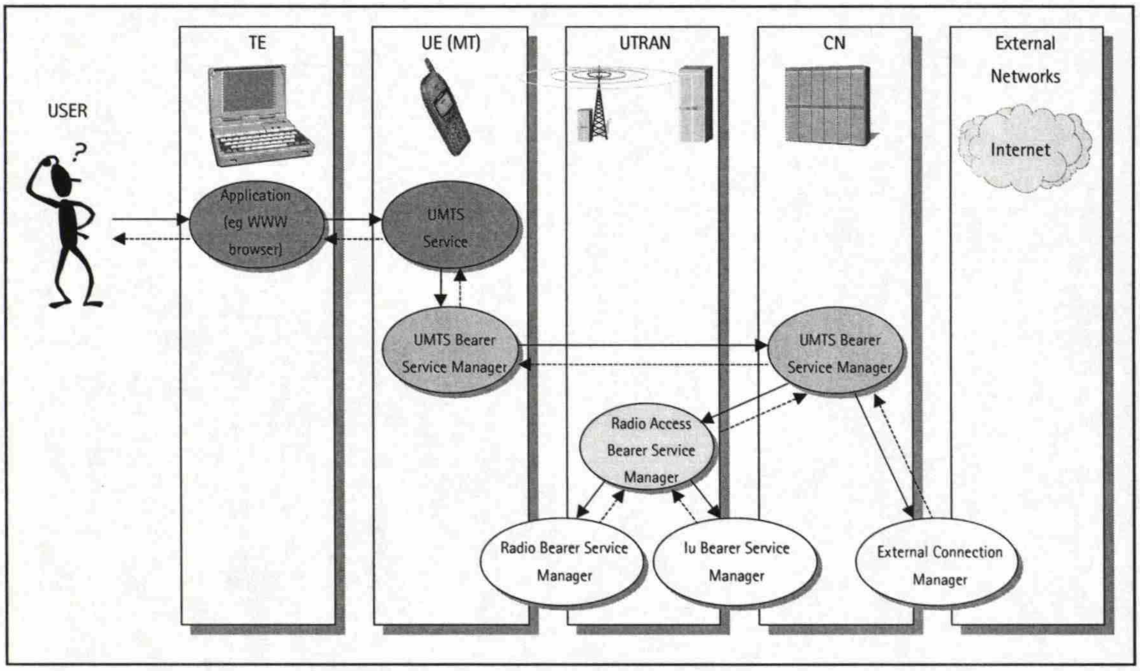

Figure 13. Bearer management functions in control plane.

In Figure 13 the user wants to connect to the Internet using the UMTS services. First the user application, eg WWW browser, requests the UE and UMTS services to provide a connection to the Internet. The user application has QoS requirements and they are mapped in the UE to the corresponding UMTS bearer attributes. The UMTS Bearer Service Manager in the UE compares the given UMTS bearer attributes to the capabilities of the UE. If the UE can support the requested service with the given attributes, the UMTS Bearer Service Manager in the UE will relay the request to the corresponding Bearer Service Manager in the CN. But if the UE cannot provide the requested service with given QoS values, the UE will inform the user application and ask it to adjust the requirements down before retry.

When the UMTS Bearer Service Manager receives the request to establish a UMTS Bearer, it first checks whether the subscriber is allowed to use the service requested in the current CN. If this admission control check is successful, the bearer establishment can proceed. Otherwise the CN informs the UE about the rejection of the bearer request. In the successful case UMTS Bearer Service Manager maps the UMTS bearer attributes to the Radio Access Bearer attributes and asks the UTRAN to start establishing the Radio Access Bearer. Also a connection to the external network (to the Internet in this case) has to be established.

UTRAN provides Radio Access Bearer Services for the CN. When RAB Service Manager in the UTRAN has received the RAB establishment request from the CN, it checks the available resources in both the radio and Iu interface. Only the UTRAN has the knowledge of the free radio resources, that is why the resource check is made in the UTRAN and not, for example, in the CN. If the requested resources are available, the RAB Service Manager maps RAB attributes to the corresponding Radio and Iu Bearer attributes and asks both Radio and Iu Bearer Service Managers to setup the actual bearers in the radio and Iu interface.

When Radio and Iu Bearer Service Managers have established the actual bearers in both interfaces, they send an acknowledgement to the RAB Service Manager. This is illustrated with dashed arrows in the figure. Now RAB Service Manager knows the requested RAB has been successfully setup and it can send the acknowledgement further to the UMTS Bearer Service Manager. The confirmation of the bearer establishment is passed through the UMTS network up to the user application. The user application can now inform the end-user about the established connection by giving the user a notification message.

The data transfer between the bearer management functions (depicted with the arrows in Figure 13) is actually implemented with the protocol messages: protocols carry the information between the Bearer Service Managers. In Chapter 4.5 it is described what information is sent and how it is handled by the protocols.

4.4.2 Bearer Management Functions in User Plane

While the control plane bearer management functions concentrate on controlling the establishment, modification, and release of the bearers, the user plane bearer management takes care of maintaining the bearers during their lifetime. The main task is to guarantee that the data transfer characteristics are in accordance with the bearer attributes agreed on the bearer establishment. The user plane bearer management functions have to maintain the QoS also for the moving subscriber.

The bearer management functions in the user plane are logically simpler than in the control plane: in each interface there are separate functions that take care of the resource allocation in that interface. The bearer management functions in the user plane are illustrated in Figure 14.

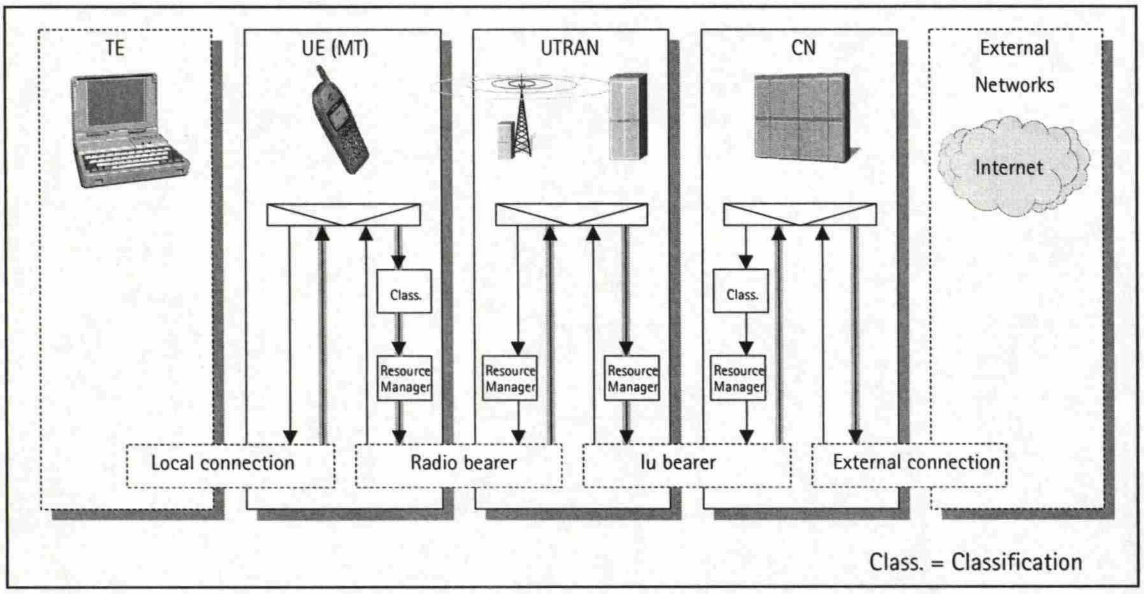

Figure 14. Bearer management functions in user plane.

The data flow from the UE to the CN is shown with the shaded arrows. The data traffic from the UE to CN and from the CN to UE follows the same principles. When the user data units come to the UE from the TE they are first classified: the Classification function categorizes the data units to be carried in the correct bearer. The Resource Manager is responsible for distributing its resources in an interface between all the established bearers requesting data transfer. The Resource Manager guarantees that each bearer will get enough transport capacity to provide promised QoS to the bearer's user.

4.5 Bearer Management Examples

The purpose of the following examples is to give an overview of what protocols are involved in bearer management. The interface-specific details are omitted. More detailed description of the bearer management functions for the Iu interface and RANAP protocol is presented in Chapter 5.

This chapter describes with signaling examples what kind of information protocols exchange when establishing bearers. A set of identifiers in different protocol layers is used to identify the resources needed to establish bearers between network elements. The identifiers, like *Radio Access Bearer Identifier* (RAB ID), distinguish the specific resource from the others. The following examples do not cover all the possible situations, only the successful establishment of a signaling, CS, and PS bearers are presented. All the examples are *mobile-originated* which means that the mobile is the initiator of the procedure. The differences to the other possibility, *mobile-terminated*, are minimal since it is always the CN that provides the bearer services to the UE.

4.5.1 Signaling bearer

The signaling connection comprises of a RRC connection and an Iu signaling connection. RRC connection is physically realized by Radio Signaling Bearer and Iu signaling connection with Iu Signaling Bearer. In Figure 15 the establishment of a signaling bearer between the UE and CN is illustrated. The name of the protocol sending the message is presented after the number of the event, and the parameters of the message are in the parenthesis under the arrow.

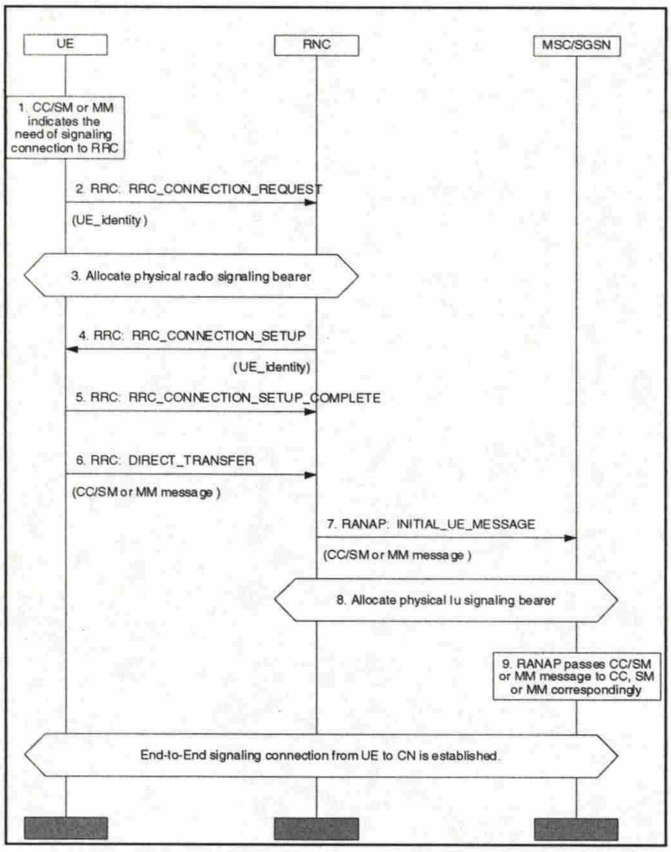

Figure 15. The establishment of a signaling bearer.

The signaling bearer establishment starts when one of the layer 3 protocols – MM, CC or SM – requests a connection from the UE to CN. The RRC protocol in the UE transmits the message RRC_CONNECTION_REQUEST to the RNC. The RNC establishes the physical signaling bearer between the UE and RNC and sends RRC_ CONNECTION_SETUP back to the UE. The UE finishes RRC connection establishment by sending the message RRC_CONNECTION_SETUP_COMPLETE to the RNC. The initiating protocol, either MM, CC or SM, sends its message from the UE to the RNC using the just established RRC connection. RRC transfers this message as a parameter in DIRECT_TRANSFER RRC message.

In the RNC, RANAP establishes an Iu signaling bearer and sends the message INITIAL_UE_MESSAGE to the CN using the new signaling bearer. When RANAP message reaches the CN, either the MSC or the SGSN, the layer 3 message is delivered to the corresponding receiver: CC message to CC protocol etc. After this a signaling connection between the UE and CN has been successfully established. The radio and Iu signaling bearers will be reserved till the signaling connection is explicitly released.

4.5.2 CS bearer

A circuit-switched bearer is always related to a circuit-switched call. In Figure 16 the establishment of a CS bearer between the UE and MSC is illustrated.

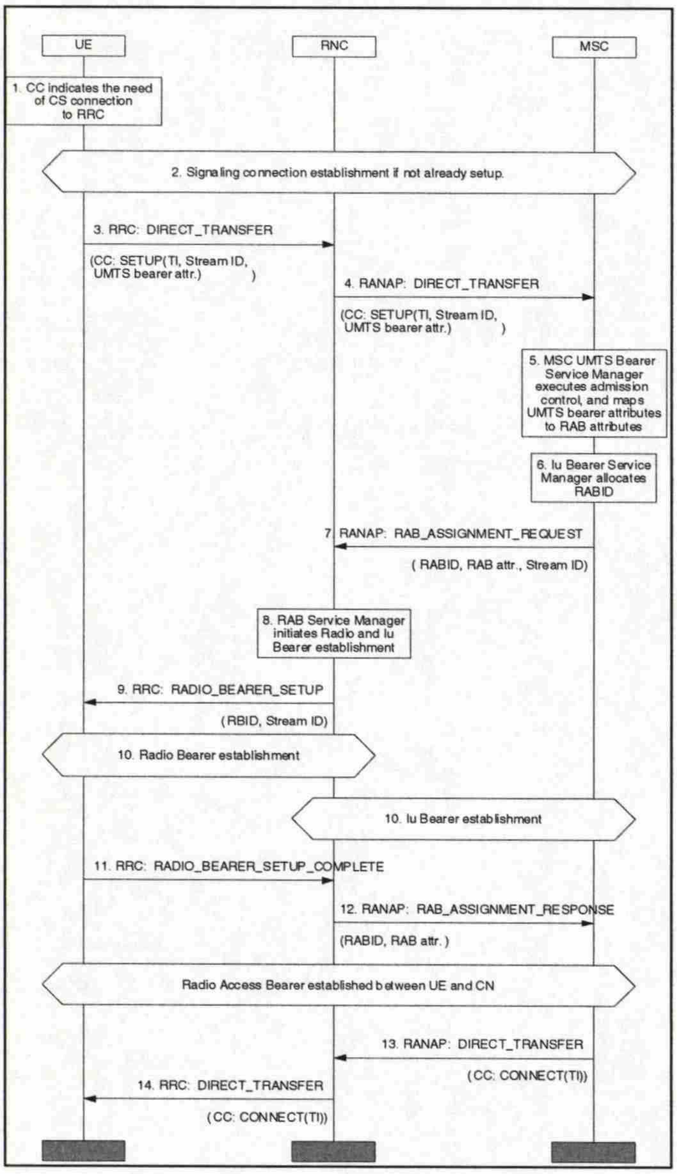

Figure 16. The establishment of a CS bearer.

The setup of a CS RAB is initiated by the CC protocol. When the end-user requests a CS service, the CC protocol in the UE converts the QoS requirements to UMTS bearer attributes. CC notifies RRC that a signaling connection is required. If the signaling connection for the UE does not exist, it will have to be established.

CC message SETUP is then sent from the UE to the CN using the services of RRC and RANAP protocols. SETUP message includes several parameters that are all UE specific. *Transaction Identifier* (TI) distinguishes the call from the others: each call in the UE has unique TI value. *Stream Identifier* identifies the UMTS bearer to be used with the call. If the value of the Stream ID is not in use with the UE at the moment, it means that CC wants a new CS bearer to be established for the new call. In the case of multicall the Stream ID value can be the same as the Stream ID of an already established CS UMTS bearer. In this case CC indicates that no new bearer will be established: the old bearer (identified with the Stream ID) will be used also with the new call. The UMTS bearer attributes are also included in the SETUP message.

In the MSC the reception of the CC SETUP message triggers several procedures. The MSC has to check that the UE has sufficient rights for the requested service. The UMTS bearer attributes are mapped to RAB attributes. Also a Radio Access Bearer Identifier is allocated for the new RAB. The RAB ID will remain the same during the lifetime of the bearer. After that the MSC requests the RNC to provide a RAB by sending RANAP message RAB_ASSIGNMENT_REQUEST to the RNC.

Upon reception of the message 7 the RNC will initiate the establishment of both Radio and Iu Bearer simultaneously. First the RNC checks that there are available resource for the requested bearers. If there is not, the RNC can still start the setup but with the reduced bearer attribute values: for example, maximum bit rate can be lowered to match the available resources.

RRC message RADIO_BEARER_ SETUP informs the UE about the new Radio Bearer. After receiving the message the UE has the mapping between the UMTS Bearer identifier, Stream ID, and *Radio Bearer Identifier* (RB ID) so the UE can route the user data traffic from a service to the correct UMTS Bearer and further to Radio Bearer and vice versa. When the UE can receive data from the new physical Radio Bearer, the UE acknowledges the bearer establishment by sending RRC message RADIO_BEARER_ SETUP_COMPLETE to the RNC.

The setup of Iu Bearer is executed in parallel with the Radio Bearer. When both Radio and Iu Bearer are established, the RNC can inform the MSC about the successful completion of the RAB assignment by sending the message RAB_ASSIGNMENT_ RESPONSE. The RAB attributes are included in the message only if the RNC was not able to establish the RAB with the attributes the CN asked. In this case RAB attributes tell the new, lowered values. In the MSC RANAP indicates to CC about the RAB assignment and CC sends the message CONNECT to the UE to indicate the successful establishment of UMTS bearer. If there are several call setups ongoing in the UE, TI value indicates for which call the bearer was established. After this the service can start using the bearer connection from the UE to UMTS network.

4.5.3 PS bearer

In UMTS packet-switched call is termed *PDP Context* (Packet Data Protocol Context). PDP Context is a connection to packet data network. Basically, the packet data network could be of any type but the most common packet data network to connect is the Internet. The establishment of a PS call is called PDP Context activation. When the PDP Context is active, the UE has an *IP address* (Internet Protocol address) so the data can be routed from the Internet to the UE. PDP Context is realized by the PS bearers. In the CN, the PS bearer terminates in the SGSN. In Figure 17 the establishment of a PS bearer between the UE and CN is illustrated.

In the PS domain the connection management protocol is the SM protocol. SM initiates the establishment of PS bearers when a user application requests UMTS PS services. Just like in the CS case, a signaling connection is required first. Application QoS requirements are mapped to UMTS bearer attributes. SM allocates *NSAPI* (Network Service Access Point Identifier) to be used during the time PDP Context is active. NSAPI corresponds the Stream ID in CS domain, but the usage of NSAPI is a little bit different: NSAPI is always related to exactly one PDP Context. In other words NSAPI identifies the PDP Context in the UE and CN. Also TI is used to identify PDP Context although NSAPI would have been sufficient. TI is included in PS domain because it was wanted that both CC and SM protocol messages have a common identifier for calls.

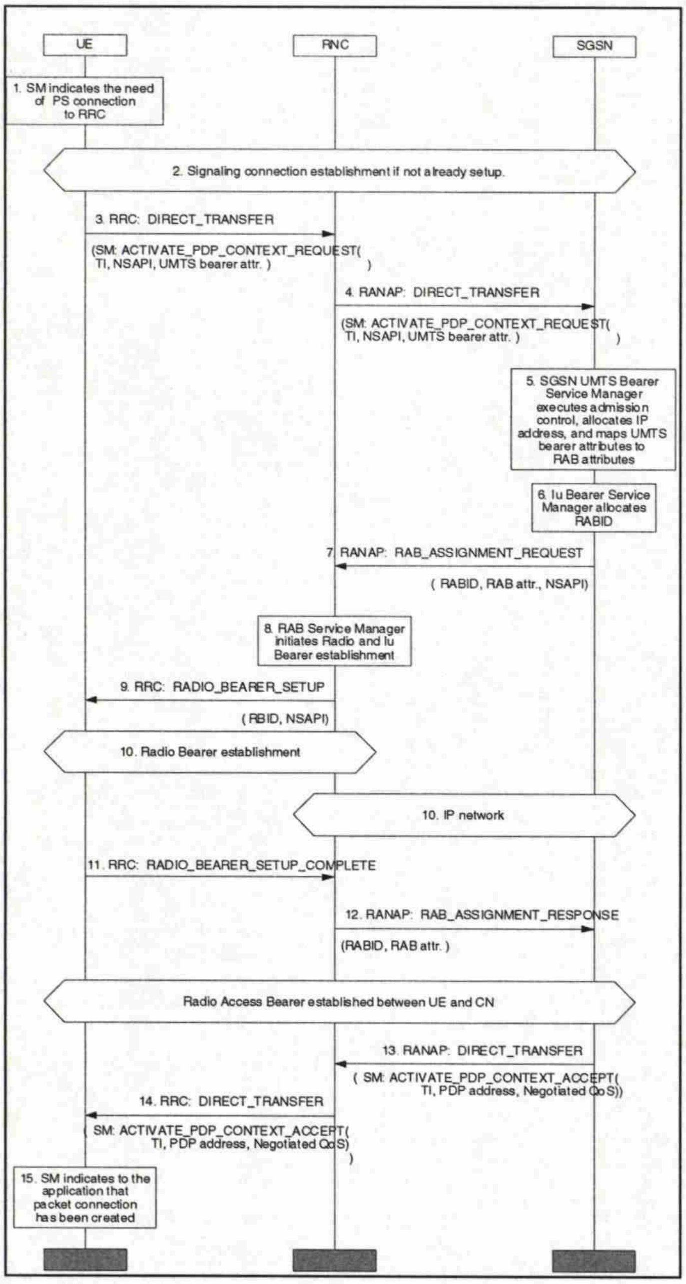

Figure 17. The establishment of a PS bearer.

SM sends the message ACTIVATE_PDP_CONTEXT_REQUEST to the SGSN that executes similar procedures as the MSC in the case of CS bearer setup. The SGSN allocates a new IP address for the UE for the duration of the active PDP Context. The RAB assignment is initiated in the same way as in the CS case: the SGSN sends the RANAP message RAB_ASSIGNMENT_REQUEST to the RNC with the assigned RAB ID and RAB attributes.

RNC establishes PS Radio Bearer in an analogous way to CS Radio Bearer, physical radio resources are reserved for the Radio Bearer. However, in the Iu interface PS bearer establishment is a little different. No actual resources are reserved because the Iu data bearer towards the PS domain is implemented with IP-based connection.

The successful completion of the RAB assignment is informed to the SGSN by sending the RANAP message RAB_ASSIGNMENT_RESPONSE. If the RNC had to reduce the RAB attributes because of the lack of the resources in the radio interface, it is indicated with the new set of RAB attributes in the message. In the SGSN these values are given to SM that sends them to the UE in SM message ACTIVATE_PDP_CONTEXT_ACCEPT as Negotiated QoS parameter. The message includes also the assigned IP address for the UE. When the UE receives the message, it can inform the user application about the successful establishment of the packet connection from the UE to the target network.

In UMTS the PS connection has some CS type characteristics. The resources in the radio interface have to be explicitly reserved for every connection and it is to some extent against the basic idea of the packet data. The nature of the radio interface forces to reserve PS radio bearers in the same way as CS radio bearers. However, if the amount of the user traffic decreases during the PS connection, it is possible to do some optimization in the radio interface: although the PS Radio bearer is reserved the actual physical radio channels that realize the PS Radio Bearer can be freed until the amount of the user data increases again. This way the basic principles of the packet data are utilized also in the radio interface.

5 IU INTERFACE AND RANAP PROTOCOL

This chapter describes in detail the functions of the Iu interface and RANAP protocol. The specification of RANAP is covered by presenting the services RANAP offers and by describing the RANAP procedures. At the end of this chapter an example of RANAP signaling flow is illustrated to show how RANAP procedures are used together when establishing a call.

5.1 Iu Interface Principles

The Iu interface connects the UMTS Radio Access Network and the Core Network together. The Iu interface will be standardized completely as an open multi-vendor interface. This allows operators to purchase the UTRAN and CN network elements from different manufacturers. The interoperability between different manufacturers' devices is guaranteed with a set of Iu specifications. 3GPP technical specifications [3GPP99e], and [3GPP99j] define the overall requirements for the Iu interface. The specification [3GPP99k] specifies the signaling bearer and [3GPP99l] the data bearers used in the Iu interface.

The same Iu interface is used to connect both CS Service and PS Service domains to the UTRAN (see Figure 8 in Chapter 3.3). The Iu interface has to be designed so that it supports the independent evolution of the UTRAN and CN technologies. In addition to that, it should be possible to develop independently CS and PS Service domains inside the CN and still use the same Iu interface to connect the domains to the UTRAN.

One of the most important tasks of the Iu interface is to offer Radio Access Bearer Service for the CN. A common set of various types of RABs is provided for both CS and PS Service domains. As a part of the RAB, the Iu interface has to support procedures to establish, maintain and release of the Iu Bearers.

From a UTRAN perspective it is desirable that the connections to the CS and PS Service domains are as similar as possible. That is why a single signaling protocol between the UTRAN and CN is defined. This protocol is called Radio Access Network

Application Part (RANAP) and it is defined in the 3GPP technical specification [3GPP99m]. Both CS and PS domains use the RANAP protocol to access the services provided by the UTRAN.

5.2 Protocol Stacks in the Iu Interface

The control plane protocol stacks in the Iu interface are presented in Figure 18. RANAP is the protocol that controls the resources in the Iu interface. One RANAP peer entity resides in the RNC and the other in the MSC or the SGSN. RANAP offers services to the protocol layers above it and uses the services of the layers below it.

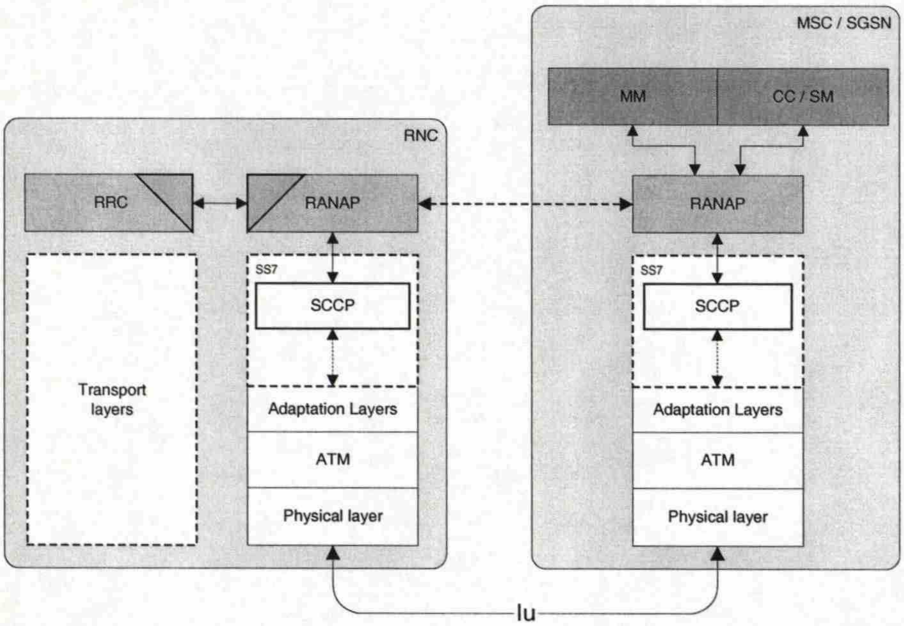

Figure 18. Control plane protocol stacks in the Iu interface.

RANAP is located on top of the Iu transport layers. RANAP uses the services of the transport layers to transfer RANAP messages over the Iu interface. At the moment 3GPP has defined that the transport layers in the Iu interface are comprised of SS7 (*Signaling System Number 7*) protocol stack over ATM. SS7 stack with ATM realize the Iu signaling bearer.

SS7 protocol stack is derived directly from the fixed networks. SS7 stack consists of several protocol layers but only the topmost protocol, *Signaling Connection Control*

Part (SCCP), is shown in the figure. SCCP offers RANAP the interface to the services of SS7 stack. The services RANAP requires from the layers below are discussed in Chapter 5.2.2.

The *Internet Engineering Task Force* (IETF) and 3GPP have specified an alternative protocol stack structure towards the PS Service domain (between the RNC and SGSN). Instead of SS7 protocol stack, signaling transport would be based on UDP/IP (*User Datagram Protocol / Internet Protocol*). However, also with this alternative SCCP will be used as an interface between the RANAP protocol and signaling transport. Since the two alternatives are included in the specifications, the manufacturers have to implement them both.

In the Iu interface ATM is used as a transport technology. ATM was chosen because it is suitable for carrying different types of user traffic: both CS and PS type of traffic as well as signaling is efficiently carried over ATM. However, ATM Adaptation Layers are needed to adapt the data traffic to be transported on ATM. In UMTS there are two types of adaptation layers in use: AAL2 and AAL5. AAL2 is used with CS user traffic and AAL5 with other types of traffic, such as signaling and PS traffic.

One of the basis of the whole Iu interface design has been the independence between RANAP and the Iu interface transport layers. RANAP is designed so that SS7 stack and ATM can be changed without any modifications to the RANAP protocol itself. Although SS7 and ATM are included in the specifications at the moment, the situation might be completely different in five years. If some future solution is found superior to the existing one, the new solution can be taken in use with RANAP easily. This way RANAP and the Iu interface is tried to make as future proof as possible.

In Figure 18 there are shown all the interfaces RANAP has with the other protocols. RANAP offers services to the upper layer protocols: in the MSC, MM and CC use RANAP services and in the SGSN the corresponding protocols are MM and SM. The services RANAP offers are described in Chapter 5.2.1. Together with RRC, RANAP transfers the messages of the upper layer protocols between the UE and CN.

The interface between RANAP peer entities is depicted with a horizontal dashed arrow in the figure. This interface presents the exchange of the RANAP messages. The protocol messages peer entities send to each other are called *Protocol Data Units* (PDUs). RANAP specification [3GPP99m] defines all the RANAP PDUs. In the specification the PDUs are collected to a groups of couple of PDUs. These groups are called RANAP procedures. Each procedure implements a service or a part of the service RANAP offers to the upper layers. RANAP procedures are covered in Chapter 5.3.

5.2.1 The Services RANAP Offers

RANAP services are divided to two groups, general control services and dedicated control services. General control services are related to the whole Iu interface between the RNC and CN. Dedicated control services support the separation of each UE in the Iu interface: they are always related to a single UE. The majority of RANAP services are dedicated services. All the RANAP messages of the dedicated control services are sent on a dedicated connection. This connection is called signaling connection. Signaling connection and its role was discussed in Chapters 4.3.1 and 4.3.3.

The overall RAB management is one of the main services RANAP offers. RANAP provides the means for the CN to control the establishment, modification, and release of the RABs between the UE and CN. Related to that, RANAP supports UE mobility by transferring RAB to a new RNS when the UE moves from the area of the serving RNS to another. This service is called RNS relocation. RANAP is also responsible for transporting upper protocol layers' messages between the UE and CN. Controlling the security mode in the UTRAN is one of the RANAP services as well. All the above mentioned services are dedicated control services.

The general control services are needed only in exceptional situations. RANAP provides, for example, means to control the overload in the Iu interface if the amount of the user traffic grows too high. In the case of a fatal failure in either end of the Iu interface RANAP offers reset service. That initializes the whole Iu interface and clears all the active connections.

5.2.2 The Services RANAP Uses

In order to offer the services of its own RANAP has certain requirements for the transport layers below. Basically it is assumed that each message (PDU) RANAP sends to the peer entity will reach the destination without any errors. In other words it is the responsibility of SS7 protocol stack to provide reliable data transfer across the Iu interface for the RANAP PDUs.

Each PDU of the dedicated control services should be sent on a unique signaling connection. In the Iu interface, signaling connection is realized by an Iu signaling bearer. The transport layers shall provide RANAP the means to dynamically establish and release signaling bearers for the Iu interface when RANAP requests it. Each active UE shall have its own Iu signaling bearer. It is the responsibility of the transport layers to maintain the bearers. If the Iu signaling bearer connection breaks for some reason, SCCP informs RANAP about it.

5.3 RANAP Procedures

This chapter describes RANAP procedures. Main emphasis is on the procedures that participate in the RAB management in one way or another. Some minor RANAP procedures are omitted since they do not handle any information related to the RABs.

Each RANAP procedure is started with a trigger from some other protocol, for example from MM, CC, SM, or RRC. RANAP do not start any process by itself. Basically RANAP procedures consist of two parts: first initiating peer entity sends a request message and then the other responses with a response message. The response message can indicate either a successful or unsuccessful outcome of the procedure, depending on the execution of the procedure in the other peer entity. The initiating entity forwards the result to the protocol that triggered the procedure. However, some procedures do not have the second part, they consist of only one message without a response.

In the following RANAP procedures are described in the order they are used when establishing a connection and bearers between the UE and CN. If not otherwise mentioned all the PDUs of the procedures are sent across the Iu interface using a dedicated signaling connection. In Chapter 5.4 there is an example of how different procedures are used together.

5.3.1 Initial UE Message

The Initial UE Message procedure is used to establish Iu signaling connection between the RNC and CN. The procedure consists of only one PDU, INITIAL_UE_ MESSAGE, that carries MM, CC, or SM PDU from the RNC to the CN. The procedure is started when RANAP receives via RRC the first upper layer PDU from the UE. RANAP requests SCCP to provide an Iu signaling bearer between the RNC and CN. This Iu signaling bearer is used to transmit INITIAL_UE_MESSAGE PDU to the RANAP peer entity in the CN. There RANAP entity forwards the upper layer PDU to the correct receiver. In Chapter 4.5.1 there is a description how the end-to-end signaling connection from the UE to the CN is established.

The Initial UE Message procedure is used only when there is no signaling connection for the UE. When the signaling connection is established the upper layer protocols' PDUs are sent over the Iu interface with Direct Transfer procedure. After executing Initial UE message procedure all the PDUs that concern the specific UE are sent on this just established signaling connection.

5.3.2 Direct Transfer

The Direct Transfer procedure is used to carry the upper layer protocols' PDUs over the Iu interface. RANAP does basically nothing with the PDUs, it only passes them over the Iu interface. There is only one PDU in the procedure, DIRECT_TRANSFER, but it can be sent to both directions. In Figure 19 the procedure is shown. Direct transfer procedure together with Initial UE message procedure is used to carry all the upper layer PDUs over the Iu interface.

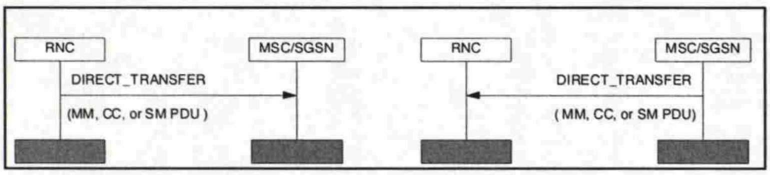

Figure 19. Direct transfer procedure, from RNC to CN and from CN to RNC.

5.3.3 Security Mode Control

The security Mode Control procedure is used to manage the encryption of the signaling and data bearers of the user. The actual encryption algorithms are implemented in the UTRAN, so all the data and signaling bearers are encrypted only between the RNC and UE. RANAP participates in encryption only by passing the encryption information from the CN to the UTRAN.

MM protocol in the CN initiates the procedure by giving the encryption information to RANAP. RANAP sends the information as a parameter in SECURITY_MODE_ COMMAND PDU to the UTRAN. The UTRAN uses the information to select the appropriate encryption device and algorithm for the user. When the encryption with a specified algorithm can be used between the UE and RNC, the RNC will ask RANAP to send SECURITY_MODE_COMMAND PDU to the CN. This indicates that the encryption was taken in use successfully.

The execution of the encryption procedures might also fail either in the UTRAN or the UE. In this case the UTRAN rejects SECURITY_MODE_COMMAND PDU and RANAP sends SECURITY_MODE_REJECT PDU with the cause of the rejection to the CN. In Figure 20 both successful and unsuccessful cases of the procedure are presented.

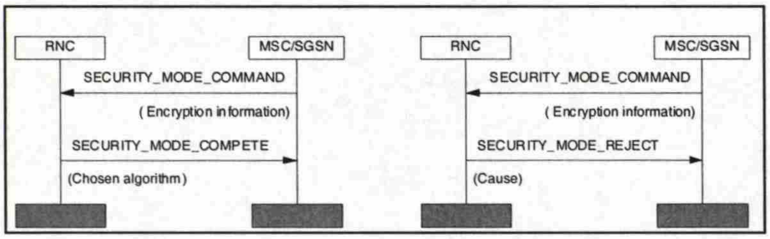

Figure 20. Successful and unsuccessful Security Mode Control procedures.

5.3.4 Common ID

The Common ID procedure is executed after the establishment of the signaling connection. The purpose of the procedure is to inform the RNC about the identity of the subscriber (the UE) that the just created signaling connection was established for. The CN sends the ID of the subscriber to the RNC in RANAP PDU COMMON_ID.

The reason for using the procedure is clarified in the Figure 21. The signaling connection from the UE to the CN consists of the RRC connection and Iu signaling connection. The just established signaling connection between the UE and MSC is denoted by number 1. If the UE will need a signaling connection also to the SGSN, RANAP has to establish Iu signaling connection between the RNC and SGSN. It is denoted by number 2 in the figure. But between the UE and RNC the same RRC connection can be used for both Iu signaling connections. The Common ID procedure is needed to inform the RNC the subscriber identity so that the RNC can multiplex the two Iu signaling connections to one RRC connection and vice versa.

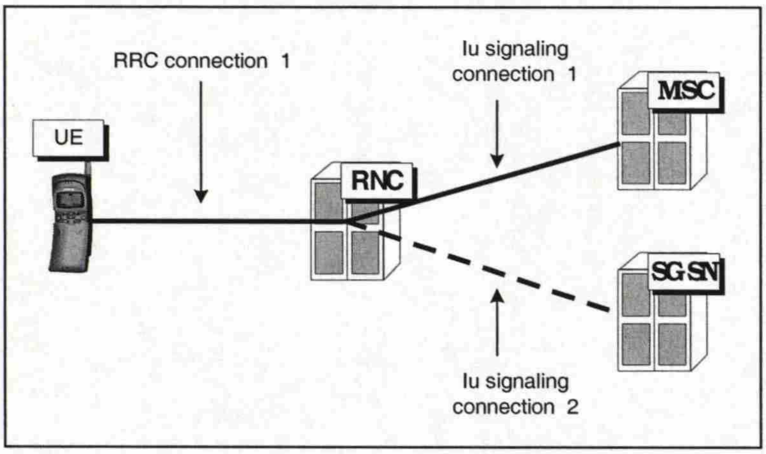

Figure 21. The relation between Iu signaling connections and RRC connection.

5.3.5 Radio Access Bearer Assignment

This procedure is used for managing all the data bearers between the UE and CN. The same procedure is responsible for establishing the new RABs and for modifying or releasing already established RABs.

The procedure is initiated by CC or SM protocol in the CN. RANAP sends RAB_ASSIGNMENT_REQUEST PDU to the RNC. One or several RABs can be established, modified, and released with the same PDU. The type of the assignment procedure depends on the parameters of RAB_ASSIGNMENT_REQUEST PDU. The PDU contains the following information:

- A list of RABs to establish with their bearer attributes and the corresponding Iu bearers establishment information

- A list of RABs to modify with their bearer attributes and the corresponding Iu bearers modification information
- A list of RABs to release

All the RABs are identified with a unique RAB identifier (RAB ID). Each of the lists can be empty. However, at least one of the lists has to contain at least one RAB. The result of the procedure in the UTRAN is reported back to the CN with the PDU RAB_ASSIGNMENT_RESPONSE.

The following signaling examples describe all the possible outcomes of the RAB Assignment procedure for the CS data bearers. The management of the PS data bearers is almost identical, only few parameters are changed in the PDUs.

5.3.5.1 Establishment

In the Figure 22, a simultaneous establishment of two RABs is illustrated. After sending the PDU RAB_ASSIGNMENT_REQUEST RANAP entity in the CN starts a timer called T(RABAssgt) to guard the arrival of the response message. When the RNC receives the PDU, it will initiate the actual establishment of the bearers in each interface. In the figure, the first bearer identified with RABID_1 is successfully established with the bearer attributes RAB_attr_1.

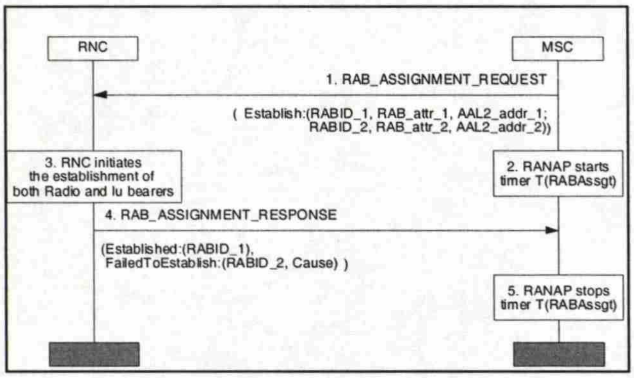

Figure 22. Successful and unsuccessful establishment of bearers.

However, it was not possible to establish the second RAB for some reason. The outcome of both establishments is sent to the CN in RAB_ASSIGNMENT_RESPONSE PDU. The failure cause for the second RAB is also included. The cause could be, for example, insufficient resource in the Iu interface. RANAP in the CN forwards the results to the initiator of the procedure and stops the timer T(RABAssgt).

If the timer expires before the reception of the PDU RAB_ASSIGNMENT_ RESPONSE, the establishment of all the RABs identified in the request message is considered as a failure. In this case RANAP in the CN aborts the RAB Assignment procedure and informs the initiator about the failure.

5.3.5.2 Modification

The characteristics of the bearers can be modified. For example, if a video call is dropped to a basic speech call, also the bearer has to be modified accordingly. For RANAP, RAB modification is very similar to RAB establishment, the only difference is that instead of an establishment list there is a list of RABs to be modified in RAB_ ASSIGNMENT_REQUEST PDU. In the Figure 23 the modification of a RAB is presented. Also modification can result a failure. In that case the original, unmodified RAB will remain in use.

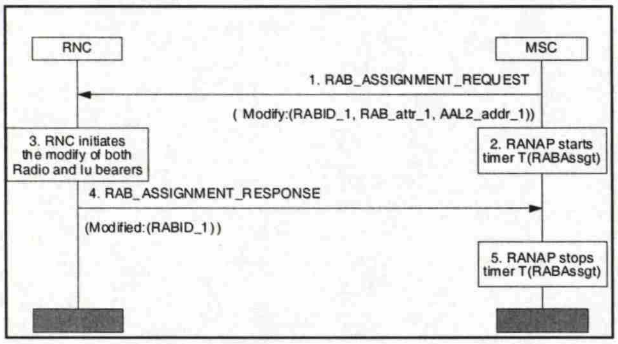

Figure 23. The modification of an established RAB.

5.3.5.3 Queuing

If there are no available resources when the RNC receives PDU RAB_ASSIGNMENT_ REQUEST, the establishment requests can be queued. RANAP in the RNC informs the CN about the queuing and starts a timer T(Queuing). The timer specifies the time period in which the requested resources have to be freed. When there are available resources in every interface and the corresponding bearer are established successfully, RANAP informs the CN about the successful RAB assignment. The procedure is finished and all the timers are stopped. The queuing is illustrated in the Figure 24.

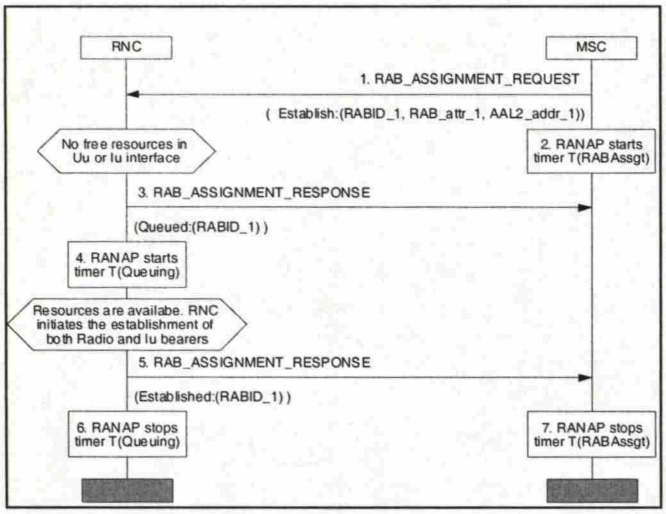

Figure 24. Queuing of a RAB.

The queuing has to be allowed for each bearer separately in the bearer attributes. When there are no resources available, the establishment will result in failure if the queuing is not allowed for the bearer. If the resources are not freed before the timer T(Queuing) expires, RANAP aborts the procedure and informs the initiator about the failure.

5.3.5.4 Release

When the user decides that he does not need the connection from his UE to the CN anymore, the corresponding RABs have to be released. RABs are released just like established: in groups or one by one. The difference to the establishment is that the release can be requested also from the RNC. In Figure 25 the release of a RAB is shown.

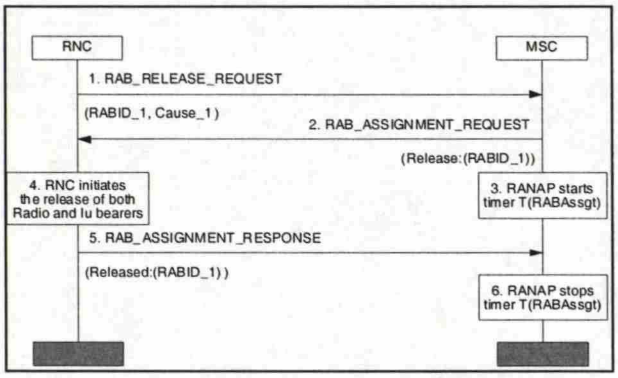

Figure 25. Release of a RAB.

Normally the RAB release is initiated from the CN, so the procedure is started from the PDU 2. But in some exceptional situations, like in case of an equipment failure, the UTRAN can request the CN to start the release procedure. In this case, the PDU RAB_ RELEASE_REQUEST is sent to the CN to trigger the actual procedure.

5.3.6 Iu Release

The Iu Release procedure is used to release the signaling connection between the UE and CN. When executed the procedure also has to take care of releasing all the active RABs, if any, of the UE. So the RABs can either be released with the RAB Assignment or the Iu Release procedure. The difference is that the Iu Release procedure releases all the physical connections of the UE, including the signaling connection. In Figure 26 the Iu Release procedure is illustrated. Just like the RAB release, also Iu Release procedure can be initiated from the UTRAN in an exceptional situation with the PDU IU_RELEASE_REQUEST. However, normally the first PDU is omitted and the procedure is started from the CN with the PDU IU_RELEASE_COMMAND.

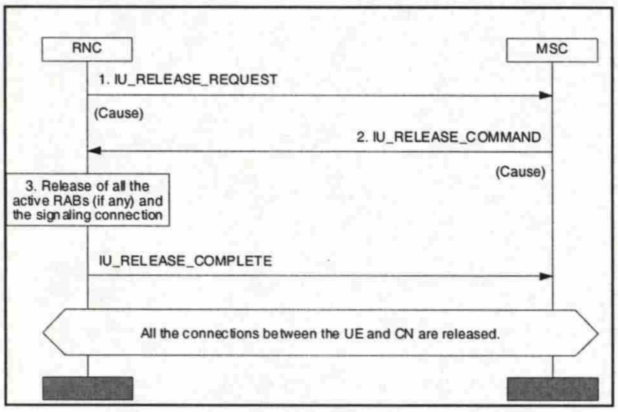

Figure 26. The Iu Release procedure.

5.3.7 Serving RNS Relocation

The Serving RNS Relocation is the most complex RANAP procedure. Actually the procedure consists of several smaller procedures that together implement the needed functionality. The purpose of the procedure is to transfer the Iu signaling and data bearers of the UE from one RNC to another. The procedure is used when the UE moves away from the area of the current RNS (called serving RNS). The logical level functionality of the Serving RNS Relocation procedure is shown in Figure 27.

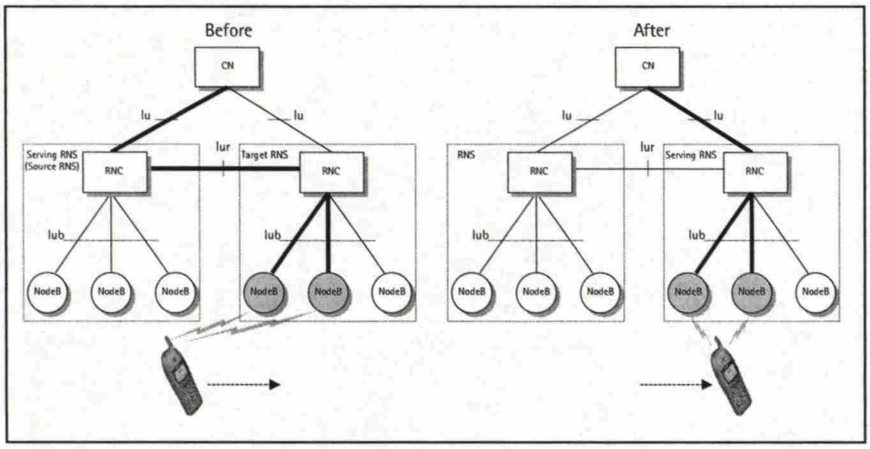

Figure 27. Serving RNS Relocation.

RANAP signaling of the Serving RNS Relocation procedure is illustrated in Figure 28. During the procedure the old serving RNS is called source RNS and the new serving RNS is termed target RNS. The RNC of the source RNS initiates the procedure by triggering the sending of RANAP PDU RELOCATION_REQUIRED. The PDU contains the identifier of the target RNC, so that the CN can direct RELOCATION_ REQUEST PDU to the correct RNC. Before the PDU can be sent, a new Iu signaling connection between the RNC of the target RNS and CN has to be established. It is done in the same way as in the Initial UE Message procedure, this time from the CN to the RNC: RANAP requests SCCP to provide an Iu signaling bearer.

The handling of the two PDUs RELOCATION_REQUEST and RELOCATION_ REQUEST_ACKNOWLEDGE corresponds the execution of the RAB Assignment procedure. However, the RABs identified in the RELOCATION_REQUEST are already established by the source RNS between the UE and CN. Now the RABs should be transferred under the control of the target RNS's RNC. Since the RABs are already set up, the corresponding radio bearers are also established. So the only thing that is needed is to allocate the Iu bearers between the target RNS's RNC and CN. The timer T(RELOCalloc) guards the allocation of the Iu bearers in the same way as T(RABAssgt) guards the RAB allocation in the RAB Assignment.

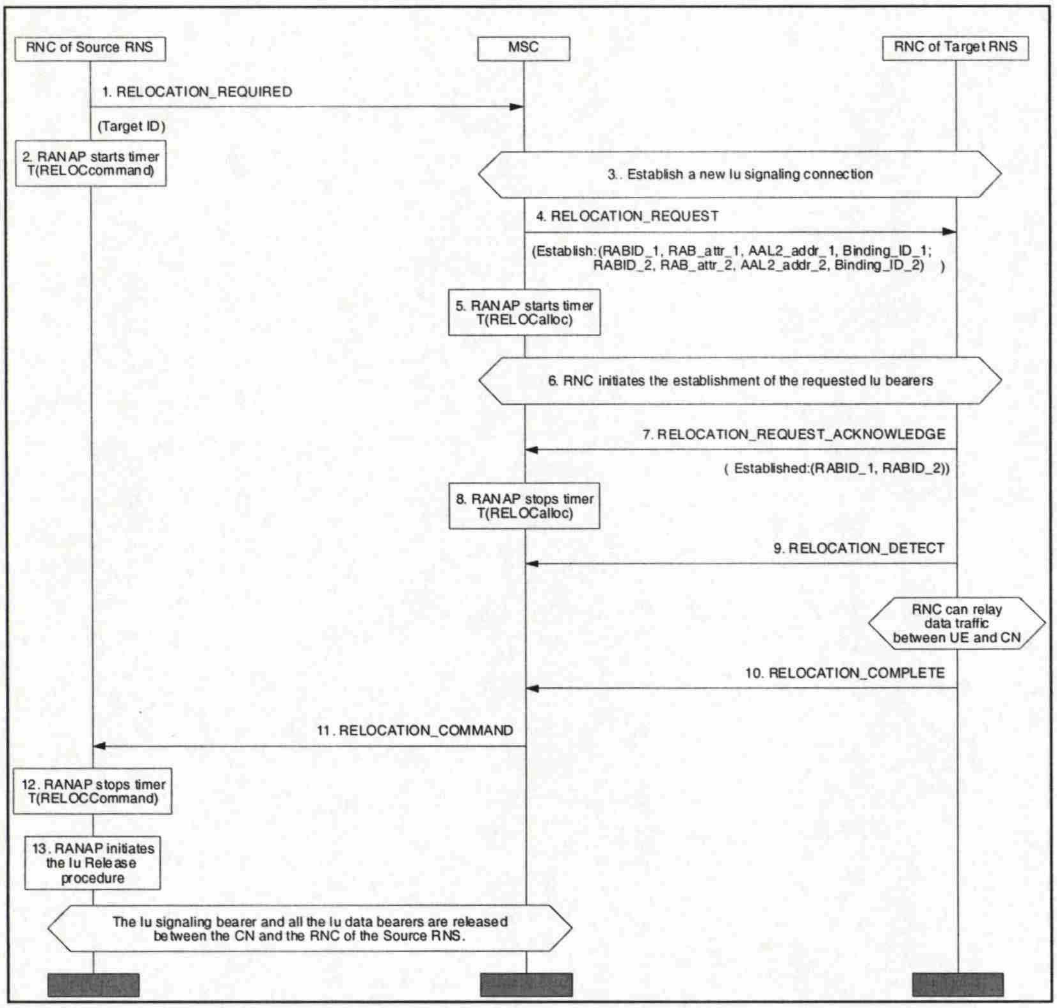

Figure 28. RANAP signaling in Serving RNS Relocation.

When the Iu bearer establishment is completed and when the target RNS's RNC has received relocation execution trigger, RELOCATION_DETECT PDU is sent to the CN. The relocation execution trigger can be received either from the RNC of the source RNS (via the Iur interface) or from the UE. After sending the PDU the new RNS can take the role of the serving RNS. When the new serving RNS's RNC can receive data traffic from the UE, RELOCATION_COMPLETE PDU is sent. This ends the Serving RNS relocation procedure for the new serving RNS.

In the CN the PDU RELOCATION_COMMAND is sent to the RNC of the source RNS. The reception of the PDU terminates the procedure in the RNC. However, the release of the old signaling connection and Iu data bearers is still required. This is done with the Iu Release procedure.

5.3.8 Paging

The Paging procedure is used to contact an idle UE for a UE terminating service request. For example when establishing a normal speech call to a UE, the first RANAP procedure to execute is Paging. The procedure consists of only one PDU, PAGING. CC or SM protocol in the CN triggers RANAP to send the PDU to the RNC that forwards it to be sent in the radio interface. The PDU contains the UE identity so the correct UE can respond to the paging when it realizes that it is searched. PAGING PDU is sent over the Iu interface using the connectionless SCCP services. SCCP offers them to transfer separate messages not related to any signaling connections.

5.3.9 Reset

The purpose of the Reset procedure is to initialize the RNC and CN in the event of a failure. The procedure applies to the whole Iu interface. The Reset procedure can be initiated either from the CN or the RNC. Both PDUs of the procedure are transmitted using the connectionless SCCP services. When the initiating entity has suffered a fatal failure (eg all memory references are lost), it is indicated to the other end by sending RESET PDU. Upon reception of the PDU all the RABs and signaling connections should be released. After this RESET_ACKNOWLEDGE PDU is sent to the initiating end. Because the Reset procedure disconnects all the active connections, it is the last measure to take in fault situations.

5.3.10 Overload Control

This procedure is used to control the amount of the signaling traffic in the Iu interface. The Overload Control procedure is executed when either Iu signaling bearers or processes handling the signaling messages are overloaded. Both the RNC and CN can initiate the procedure by sending the PDU OVERLOAD to the other. The receiving end should reduce the signaling traffic with a certain amount. After a time period specified with a timer the receiving end can increase the signaling traffic to the normal level. The PDU of the procedure is transmitted using the connectionless SCCP services.

5.4 Usage of RANAP Procedures

This chapter describes how RANAP procedures are used together when establishing *mobile terminated call* (MTC) of CS type. MTC means that the call is connected to a UE. The RANAP signaling in *mobile originated call* (MOC, the call is connected from a UE) is almost the same: the only difference is that the Paging procedure is omitted.

5.4.1 Mobile Terminated Call

In Figure 29 the establishment of MTC is illustrated. First the CN initiates the Paging procedure. The UE responds to the paging by sending an MM message in the INITIAL_UE_MESSAGE. Using the just established signaling connection MM executes few procedures to authenticate the subscriber. Then MM triggers the Security Mode Control procedure to encrypt the signaling and data bearers. After the Common ID procedure, call establishment proceeds in the way described in Chapter 4.5.2.

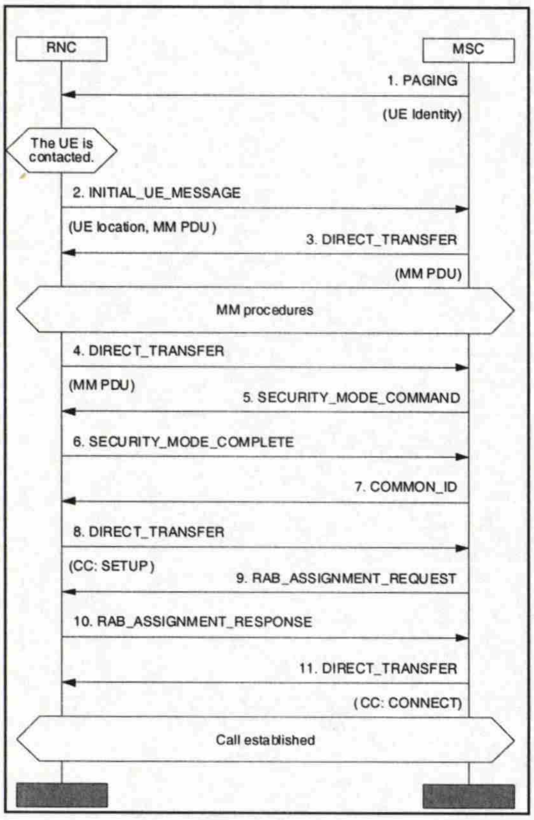

Figure 29. RANAP signaling in mobile terminated call.

6 IMPLEMENTATION OF RANAP PROTOCOL

This chapter describes how the RANAP protocol was implemented as a part of a Nokia Research Center project. First some general background information about the project is given. The tools that were used in the project are described. The overview of the RANAP implementation and a few experiences from the project are presented.

6.1 General

The implementation of RANAP described in this chapter is a part of software package that models the functions of the UMTS protocols. The software package is developed to be used as a library of protocol implementations that other projects can reuse. All the control plane protocols of each UMTS network element are implemented in the software package.

The protocol implementations are tested in a test system consisting of one MSC connected to two RNCs and a few base stations. The configuration of two RNCs are needed to test eg Serving RNS relocation. The test system also includes a couple of UEs.

The project responsible for the software package was started in Nokia Research Center in the summer of 1998. The initial functional version of the project was finished in the end of 1998. In the beginning the interfaces between the network elements were only modeled as internal interfaces inside the network model. During 1999 the implementation has been updated and improved due to constantly changing draft specifications. The coding functions were also added for every protocol. As a consequence all the interfaces between network elements have the same functionality as real-life implementations. In principle, the model could be connected to real network elements.

6.2 Tools and Languages

Complex telecommunications systems cannot be implemented without powerful tools. Tools should relieve the designer from taking care of every small detail to concentrate on higher-level logic. In the project SDL (*Specification and Description Language*) was used as an implementation language. All the PDUs and primitives were defined using ASN.1 (*Abstract Syntax Notation One*). A commercial toolset SDT (*SDL Design Tool*) offers several tools for SDL-based design of real-time systems. In the following the characteristics of SDL, ASN.1 and SDT are briefly described.

6.2.1 SDL Language

SDL was initially developed, as the name implies, for specification purposes. SDL was used as a part of the specifications to clarify the functions of the protocols. However, nowadays SDL is used like any other high-level programming language. The development work of SDL started already in 1972. The first standard defining SDL was published in 1988 by ITU-T [ITUT94]. Since then some additions, like object-oriented features and combined usage of SDL and ASN.1, have been included to the standard. A comprehensive description of SDL language can be found from [ElHoSa97].

In SDL there are two representation notations, textual and graphical. Graphical representation is easier to understand, textual format is seldom used. The structure of SDL code is hierarchical. The topmost level is the *system* level. SDL system consists of one or several *blocks*. Each block can contain one or several *processes*. Processes define the behavior of SDL code. In other words, processes contain the state machine of the protocol to be implemented. Processes communicate with each other by sending messages on signal routes that connect the processes together.

6.2.2 ASN.1

ASN.1 is a generic notation for the specification of data types and values. ASN.1 is standardized by ITU-T in the recommendation X.680 [ITUT97a]. The standardization work started already in the early 1980's. The general concept behind ASN.1 is to describe information independently from the transfer format. ASN.1 is widely used in standardizing telecommunications protocols, for example PDUs of RANAP are defined with ASN.1.

Associated with ASN.1 there are encoding rules that determine the transfer format for every ASN.1 value. Conversion of ASN.1 values to the transfer format is called *encoding*. The reverse operation is *decoding*. The operations of encoding and decoding are commonly called simply coding. It is defined in 3GPP that PER (*Packed Encoding Rules*) encoding rules are used in the coding of RANAP PDUs. PER is standardized in ITU-T recommendation X.691 [ITUT97b]. In the project, an in-house developed tool was used to create the coding functions.

6.2.3 SDT

SDL Design Tool is a commercial tool developed by Swedish company Telelogic. SDT is used widely within the telecommunication companies. The current version of the tool is 3.6. A new version is released two times per year, so the tool is continiously developed.

SDT consists of several tools. SDL Editor is a graphical tool used to write and edit SDL code. SDL Analyzer checks the correctness of the written code. MSC (*Message Sequence Chart*) Editor can be used both in analysis and in design of the SDL model. Simulator is ment for executing the SDL code. Simulator also creates MSC trace from the executable SDL code. The MSC trace can be used to check that the system behaves as it should. Validator is a helpful tool in testing: Validator checks the SDL model formally and detects for example all the deadlocks and never-executed parts of the code.

6.3 SDL Implementation of RANAP

The SDL implementation of RANAP is made in accordance with the 3GPP specification [3GPP99m]. The implementation realizes the specification for the most part, although some minor functionality is not included. For example some messages are left out since it is still unclear are the PDUs really needed in the real implementation. Also some information elements (parameters) of the PDUs are not implemented since they are not essential for the operation of the whole UMTS model.

RANAP is an asymmetric protocol, which means that the RANAP peer entities (one resides in the RNC and the other in the CN) are not alike. In practice this shows in the handling of the PDUs: normally each PDU is sent only to one direction, either from the RNC to the CN or vice versa. Both peer entities were implemented in the project. The following examples are taken from the CN side.

The structure of project's SDL model is very logical, each network element is implemented in its own SDL block. In Figure 30, the SDL system level of the system UMTS is presented. The system consists of eight blocks: two UEs, three Node Bs, two RNCs and one MSC. On the right side of the figure, the structure of the block MSC is illustrated. Inside the MSC, the functionality of each protocol is divided into a single SDL block. As the figure shows, SDL blocks can be further divided to several sub-blocks. RANAP_CN stands for the CN peer entity of the RANAP protocol implementation.

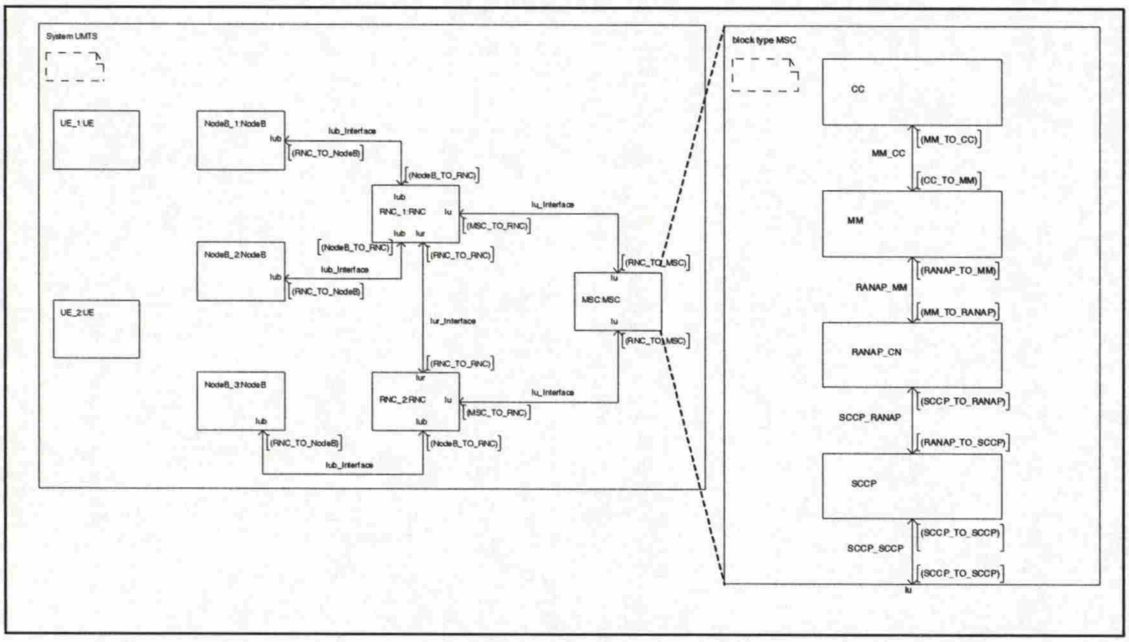

Figure 30. SDL system level of the project and the MSC block level structure.

6.3.1 Phases

It requires a lot of work before a specification is realized as an implementation with a programming language. The job cannot be done in an unorganized manner. That is why the project was conducted in six phases: analysis, design, implementation, (module) testing and validation, integration, and integration tests. The phases are not simply executed one after another. Instead, the process is more like an iterative one: for example, if a bug is found during the module testing, it is perhaps necessary to revisit the analysis and design phases before removing the bug from the code and then test it again.

The specifications are normally written in such a manner that there is some room for the interpretation. In addition to that, the specifications describe the functionality of the protocols with quite a narrow view. It is hard to get the big picture and see how different things in different protocols are actually related to each other. The analysis phase was meant for creating an understanding of the protocol and its relations to the whole UMTS system.

The purpose of the design phase was to prepare unambiguous plans for the SDL implementation of each protocol. Another important goal was to define the primitive interfaces between different protocol layers. Normally the protocol specifications only specify how the protocol peer entities change information with each other. In other words the specifications define the PDU or peer interface of each protocol. However, protocol layers need to communicate inside network elements too. These "vertical" interfaces (eg between RANAP and MM) are called primitive interfaces. They belong to the group of so called implementation-specific matters: every manufacturer can decide on his own how to implement the internal primitive interfaces inside a network element.

When all the interfaces are defined and the plans for the SDL model are ready, the implementation phase can be started. The implementation and testing/validation phases are actually conducted in parallel. When some feature of the protocol is implemented, the feature should be immediately tested and validated.

During the integration phase all the different SDL implementations of the protocols are connected together. First the protocols of each network element are combined and then the network elements are connected forming the UMTS system shown in Figure 30. The aim is that the integration and integration test phases would not require any modifications to the RANAP or any other protocol implementation. If the design phase with the definition of the primitive interfaces has been conducted properly, each protocol should work with its neighbors without problems. However, it is quite normal that few details are not thoroughly considered during the design or analysis phase and these details cause problems during the integration. So it is possible that some modifications are needed to protocol implementation also in the integration phase.

6.3.2 Structure

This chapter describes the internal structure of the RANAP_CN SDL block from Figure 30. The structure of the peer entity in the RNC is almost identical. The block RANAP_CN consists of three SDL processes: RANAP_CN_Conn, RANAP_CN_Mgr and RANAP_ CN_RED. The structure of the RANAP_CN block is shown in Figure 31.

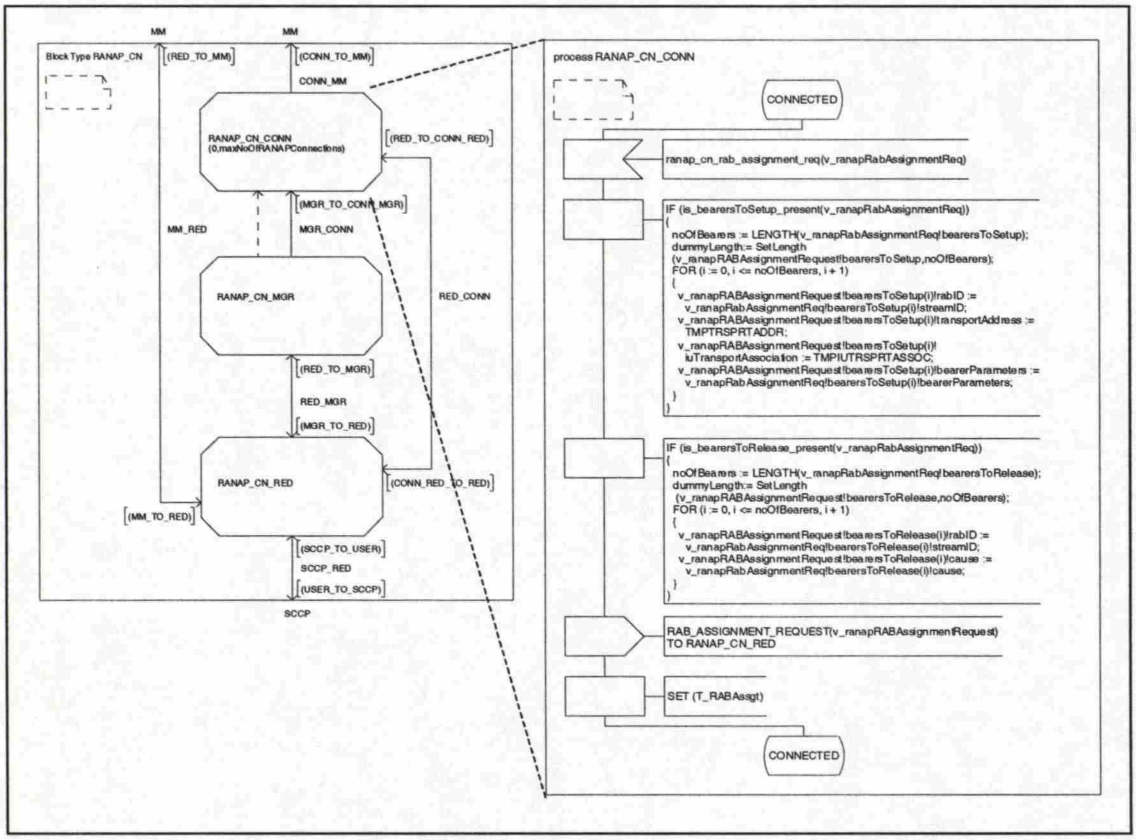

Figure 31. The structure of the RANAP_CN block with a clip of SDL code from RANAP_CN_CONN process.

RANAP_CN_Mgr is the manager process of the whole RANAP_CN block. RANAP_ CN_Mgr is responsible for creating a new user specific RANAP connection when a signaling connection from the UE to the CN is established. In practice this means that

RANAP_CN_Mgr creates a dynamic process RANAP_CN_Conn to take care of all the signaling traffic of that signaling connection. The creation of the dynamic processes is denoted with a dashed arrow in the SDL. In addition to that, RANAP_CN_Mgr handles also non-user-specific PDUs, like PAGING and OVERLOAD. When executing the SDL model, there is always exactly one instance of the RANAP_CN_Mgr process.

RANAP_CN_Conn process is responsible for handling all the user-specific PDUs. There is one instance of the RANAP_CN_Conn process for each user (UE) who has an active signaling connection. Further, each RANAP_CN_Conn process is associated with a single SCCP connection (Iu signaling bearer).

RANAP_CN_RED process is responsible for routing, encoding and decoding functions. All those functions are not directly related to the functional logic of the protocol and they are gathered under a distinct SDL process to clarify the internal structure. In SDL there is no in-built system for routing the messages to different processes. That is why special hand-written routing routines are needed. The routing is based on different identifiers associated with PIDs (*Process Identifier*).

In Figure 31 there is also a clip of SDL code from the process RANAP_CN_Conn. By going through it, the function of the RANAP_CN block can be clarified. In the example ranap_cn_rab_assignment_req primitive is received from CC protocol. RANAP_CN block has only one interface towards upper layer protocols; CC messages are relayed transparently through the MM layer. All the incoming messages to RANAP_CN block go first to RANAP_CN_RED process that routes the primitive to the correct dynamic RANAP_CN_Conn process. In the figure the primitive is received in the Conn process. First the parameters received from CC are copied to the data structures of the RANAP PDUs. Then the PDU corresponding the primitive is sent to the RANAP_CN_ RED process for encoding. A timer T_RABAssgt is set to guard the reception of the response PDU from the RNC. When the RED process has encoded the PDU, it can be sent to the RNC using the services of the SCCP protocol.

6.3.3 State Machine

A state machine defines the behavior of the telecommunications protocol. RANAP specification does not specify any state machine; it is left for the manufacturers to decide the structure of the RANAP state machine. Due to the nature of the RANAP protocol the handling of all the PDUs take place in a single state. In the implementation this state is called CONNECTED in RANAP_CN_Conn process and the state is also shown in Figure 31. For the sake of clarity two other states were added to the implementation, CONNECTION_PENDING and CONNECTION_RELEASE. The state transition to CONNECTION_PENDING happens when signaling connection establishment is started. Upon reception of the notification of the successful Iu signaling bearer establishment a state transition to the state CONNECTED will take place. The state CONNECTION_RELEASE is visited when sending the PDU IU_RELEASE_ COMMAND. After releasing the signaling connection the process RANAP_CN_Conn is terminated.

6.4 Experiences

The project was started before there were any official specifications available. Today some of the UMTS protocol specifications are still in draft stage and under modifications. However, the project's SDL model should reflect the current status of the specifications. This very dynamic environment sets tough requirements: the design of the protocol implementation should be good enough to cope with all the possible future changes.

The interfaces are never defined too well. All the interfaces between the protocol entities and inside the protocol implementation have to be defined as completely and as early as possible. This will reduce effort needed to rewrite codes due to changing interfaces. Of course, unambiguously defined interfaces are essential also in the integration.

At the moment, validation is a hot topic in the protocol engineering. One of the project's goal was to test the usage of SDT Validator. Disappointingly, Validator was found incapable of handling real-life protocols, only small models with few implemented

PDUs could be validated. The whole UMTS SDL model was never successfully validated due to deficiencies of the tool. It has to be said that, after all, validating is useful. During the project some bugs undetected in testing were found with validating.

SDT requires strict rules for how it should be used. At the moment SDT does not support the distributed implementation process well. The success of the project depends on the professional skills of the programmers. For example, the file structure of the SDL model has to be clearly defined. Otherwise the integration will be nightmare if everyone has used their own ad hoc methods to divide the SDL model to the files. SDT is a powerful tool but a lot of responsibility is left to the programmer.

Overall, SDT is fairly easy to use. The graphical user interface of the tool and the graphical SDL are fast to learn. SDT is especially suitable for rapid prototyping: it is easy to make small SDL models with SDT to verify some new solutions or ideas. SDT is also very handy when testing SDL code. With SDT Simulator it is easy to control the execution of the code and see graphically is the code behaving as it should.

7 CONCLUSIONS

The development work for third generation mobile communication systems is going on around the world. Initially the objective was to develop a single global system, but currently various standardization efforts exist. This thesis concentrated on UMTS, the third generation system standardized by 3GPP.

The UMTS system is based on the evolution of the GSM platform. The network architecture of both systems is almost similar. However, the services UMTS offers are much more diverse.

The objective of this thesis was to describe the bearer management in UMTS. The bearer concept with the hierarchical, layered bearer service architecture was clarified. It was shown that the bearer concept is required to fulfill the different transmission requirements of each UMTS service. It was also presented how the different UMTS services are realized by using appropriate type of bearer.

The characteristics of the bearers are defined with a set of attributes. It was described how the UMTS system treats the end-users' requirements and how the requirements are converted to the bearer parameters. In UMTS the functionality of the bearer management is distributed among the network elements. Each network element participates by taking care of some small task. Successful establishment of an end-to-end connection requires that all bearer management functions and procedures in every network element are successfully completed.

One of the protocols taking part in the bearer management was described in detail, namely RANAP. The functions of RANAP related to bearer establishment, modification, and release were presented. Also an example implementation of RANAP modeled with SDL was given. RANAP together with the rest of the implemented UMTS protocols showed that the overall bearer management is working as specified in the current versions of the specifications.

3GPP is actively working on the UMTS standardization. Still even the first release of the specifications is not yet finished. There may be some minor modifications in the details of the bearer management functions in future releases. However, the distributed approach hinders from making any major changes to the basic principles of bearer management. It is probable that the guidelines specified today will be valid also in the future. The design and implementation of the bearer management should be done carefully. The efficiency of the bearer management functions is essential for the successful operation of the UMTS network.

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