UMTS
The Fundamentals

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Contents

Preface

UMTS is a so-called Third Generation (3G) mobile radio system and is seen as the successor to Second Generation (2G) systems such as GSM and to evolved 2G systems such as the General Packet Radio Service (GPRS). It has a completely different air interface that is based on Code Division Multiple Access (CDMA), whereas most of the 2G and evolved 2G systems in use in most parts of the world use Time Division Multiple Access (TDMA). The expert knowledge on the functioning and behaviour of 2G networks can only be of limited use in 3G systems. As a consequence, people working in UMTS development, marketing, operation and teaching have to update the knowledge to be able to fulfil their duties. The introduction of UMTS in the field as the next generation technology requires knowledge of its concepts, architecture, procedures and techniques as a prerequisite for all those involved in the introduction of UMTS in one way or another.

This book presents the valuable experience gained by the authors from teaching university courses on UMTS graduate students and teaching continuing education courses to engineers and management personnel in industrial companies. The material contained is based on the authors' research work on UMTS and the implementation and traffic performance evaluation of the complete UMTS protocol stack [35]. In presenting the course in form of a book we are acceding to the requests of companies and professional teaching organizations to make the material available to the public.

The material has not been selected with the intention of providing developers of UMTS with the detailed knowledge necessary to design and improve a real system but to enable those working with UMTS to be able to understand the relevant concepts and their impact on the roll-out, operation, usability and capabilities of the system. The comprehensive introduction to UMTS is aimed at teaching the basics, functions and ways of operation of UMTS to those working in development departments and to operators of UMTS in an easy-to-follow manner. Since it is planned to introduce two versions of UMTS, namely one frequency and one time division duplexing based system, both are covered here.

To ease the study of the material and to allow for a common basis of understanding, we open the book with two chapters on the basic functioning of cellular mobile radio systems and digital transmission of information via radio channels. After that, chapters on the transmission technique and the protocols of the UMTS air interface follow. Later sections of the book are devoted to the system architecture, the various network elements and the protocols used in the UMTS fixed core network.

The keys to the commercial success of UMTS are new services that are not available with the existing mobile radio systems. This is why we introduce future service architectures and services for UMTS that have already been experimented with in the GPRS. Further, we describe the development paths to evolved 3G systems and as well as discussing spectrum availability, we evaluate the suitability of Wireless Broadband Systems based on Local Area Networks (LAN) to supplement 3G mobile radio systems.

UMTS: The Fundamentals is primarily aimed as a course book for self-study and as background material for course teaching. Beyond what is available from the textbook we offer additional teaching material that can be ordered using the URL http://www.umts-thefundamentals.com. Based on their knowledge of GSM, GPRS and UMTS, the authors have started a consulting company called P3 Solutions, which offers courses, consulting services and testing in the field of 2G and 3G (http://www.p3-solutions.com).

Our warm thanks go to Ingo Forkel, PhD student at the chair for Communication Networks at Aachen University of Technology (RWTH) for his valuable input and his assistance in the completion of the book. The text has been gradually expanded from a first version published in German. Our thanks go also to Hedwig Jourdan von Schmoeger for the careful translation into English. Thanks are also due to Mark Hammond of Wiley & Sons for his excellent co-operation during the preparation of this book.

Aachen, March 2003

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1 Digital Data Transmission

This course unit briefly summarizes some of the basic concepts of digital message transmission that are important for understanding data transmission in UMTS. The particular topics covered are digital modulation, the spectral characteristics of signals, the problematic aspects of error-prone transmission and the throughput achievable in digital wireless communication systems.

1.1 Digital modulation

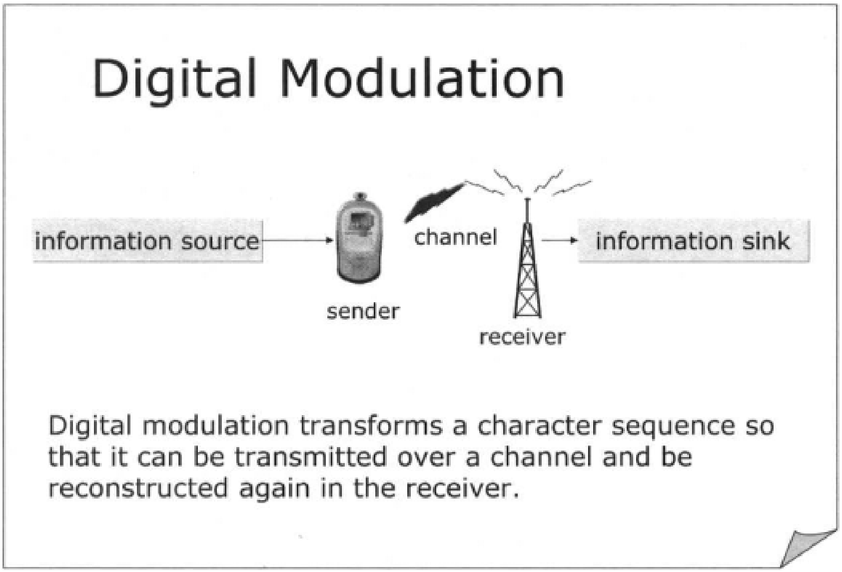

Figure 1.1: Digital modulation

A message transmission system generally consists of a message source, a transmitter, a channel, a receiver and an information sink. Digital modulation is the modulation of messages represented by characters that takes place in the transmitter (see Figure 1.1). This means that by digital modulation a character sequence supplied by an information source is transformed so that it can be transmitted over a channel and be reconstructed again in the receiver. In

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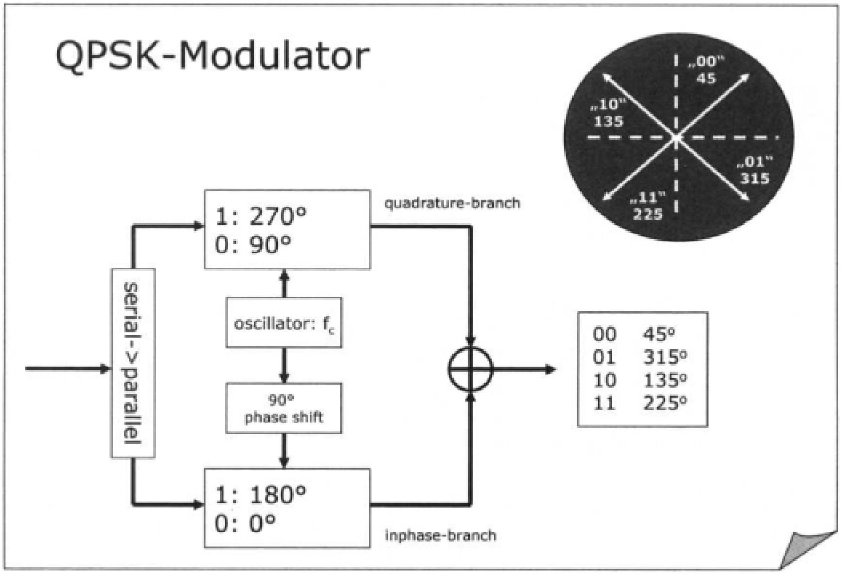

Figure 1.2: QPSK modulation

the case of mobile radio systems, a channel is the mobile radio channel with its typical characteristics, such as multipath propagation, time dispersion and Doppler distortion.

1.2 QPSK modulation

The digital modulation that occurs in the UMTS application is called *Quaternary Phase Shift Keying* (QPSK). As the name of the modulation technique implies, QPSK maps the bit sequence being transmitted to a symbol sequence, the elements of which consist of an alphabet of four different symbols. In the signal transmission a modulation symbol corresponds to exactly one of the four possible phase positions of the carrier wave. This is also referred to as four-ary modulation [19].

At the modulator entrance two successive bits are combined into a bit pair; thus the serial bit stream is converted into two parallel bit streams. The branches resulting from the division into two bit streams are called inphase and quadrature branches. Depending on the value of the respective bits, a sine oscillator with a specific phase position is produced in each branch. The addition of the two oscillators that originate in this way in turn produces an-

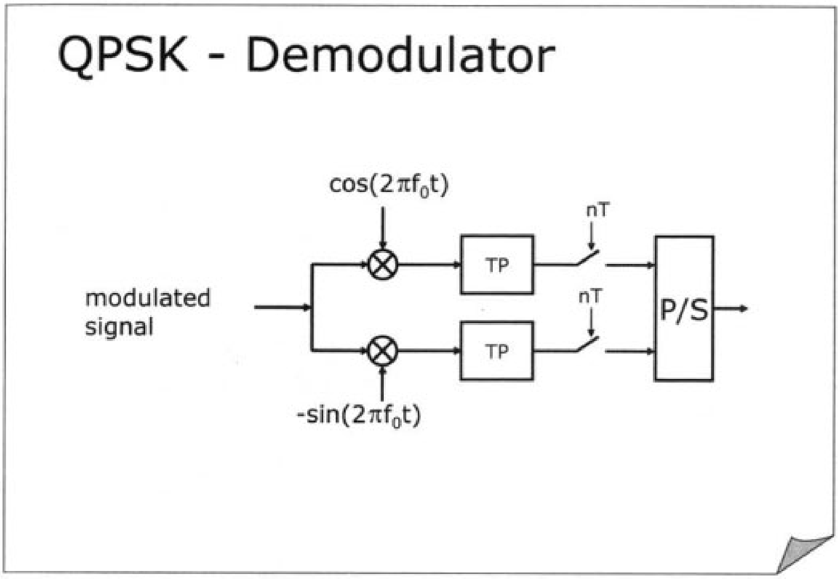

Figure 1.3: QPSK demodulator

other sine oscillation, the phase of which now depends on the bit pair mapped in it. The carrier wave experiences a symbol-dependent phase shift as a result.

Figure 1.2 shows the 4-level complex signal space constellation with an arrow representing an oscillation with a particular phase. The phase corresponds to the angle between the abscissa and the respective arrow. One can see, for example, that the symbol representing the two bits 00 is implemented with a 45 degree shift phase with regard to the reference phase.

A look at only one branch of the modulator shows that each branch alone implements two-phase shift keying, with the phase of the fundamental wave of the two branches shifted against each other by 90 degrees.

There are different ways in which QPSK-modulated signals can be received and the transmitted bit sequence recovered. A simple example is shown in Figure 1.3. The modulated signal is multiplied into two separate branches, once with a sine and once with a cosine signal. Both signals have the same frequency as the oscillation of the QPSK modulator and run cophasal with regard to the reference phase. This multiplication produces higher frequency parts that can be eliminated through a lowpass filter. The signal is then sampled in each branch, and the sampled values are used to reach a decision on the value of the transmitted bits. Finally, the two bit streams created this way are joined again into a serial bit stream.

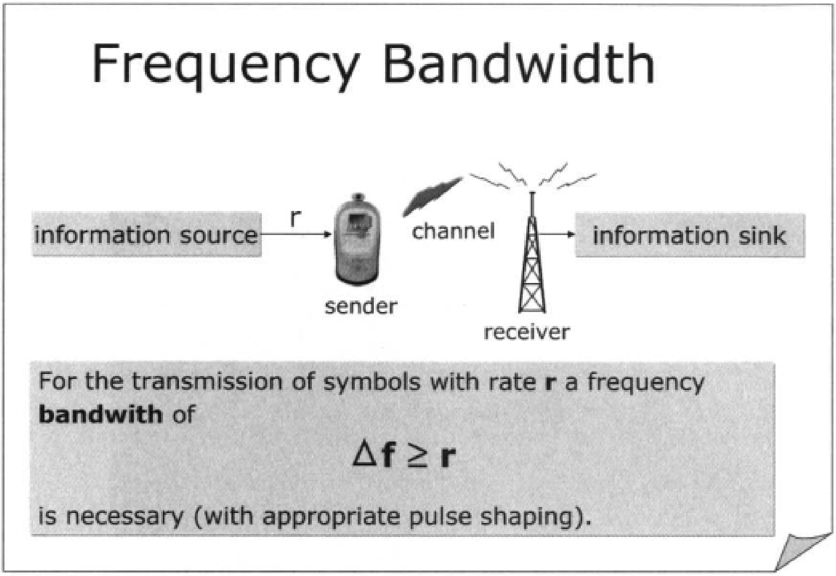

Figure 1.4: Bandwidth requirement for digital modulation

1.3 Spectral characteristics of modulated signals

With digital message transmission the need for frequency bandwidth depends on the rate of the transmitted modulation symbols. Data transmission over a mobile radio channel generally requires a frequency bandwidth that is higher or the same as the symbol rate (see Figure 1.4). In practice, however, the requirement is for frequency bandwidths that are higher than the symbol rate. This is essentially due to technological limitations that prevent the generation of ideal signal forms. For restriction of the transmitted signal to a given frequency bandwidth it is necessary that the transmitted bit pulses are formed in such a way that the spectrum required to transmit these formed pulses is as small as possible [19].

If a data signal is to be transmitted over a radio channel with limited bandwidth, the rectangular bit signal forms with their finite length have to be remade through pulse forming into signal forms with a theoretically infinite length. The fact is that in principle the time duration of a signal is reciprocally proportional to the frequency bandwidth required to transmit the signal. Theoretically, a signal of finite length requires a frequency spectrum of infinite width (see Figure 1.5).

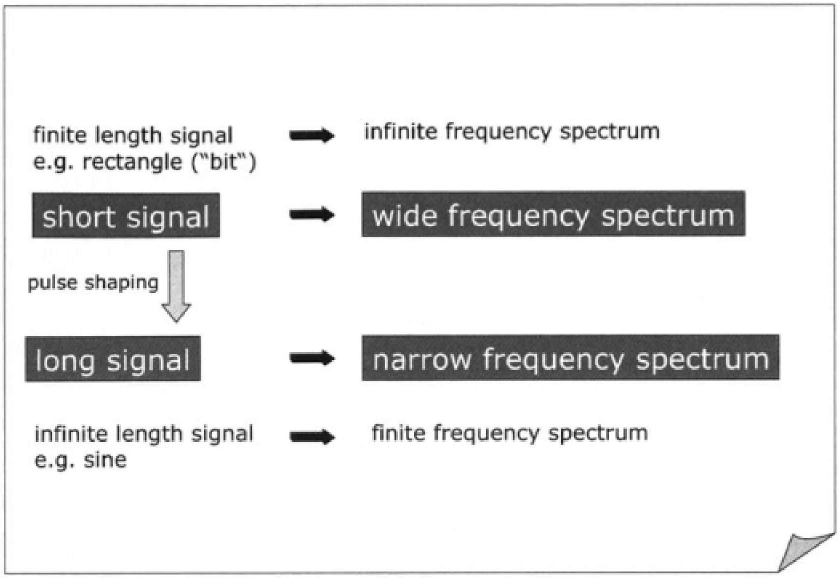

Figure 1.5: Characteristics of pulse forming

For example, operating a light switch generates an abrupt change in the voltage signal on a circuit. Due to this abrupt change, a theoretically infinite broad spectrum is needed to transmit this signal. In fact, the usable frequency bandwidth of a power supply circuit for signal transmission is limited so that a measurement of the voltage at the end of the circuit shows that changes are slower rather than abrupt. The circuit then has the effect of a pulse former, the signal is changed when it is transmitted over the circuit. If this change is to be prevented, then only those signals that require a restricted frequency bandwidth should be transmitted.

As already explained above, the transmission of a character stream requires at least a frequency bandwidth corresponding to the rate of the character stream. The required frequency bandwidth cannot be made arbitrarily small through pulse forming.

Because bits can only have the values zero or one, the signal form specified by a bit sequence is rectangular. The signal increases and decreases abruptly, which results in an abrupt change to the phase of the carrier wave in the QPSK modulator. The aim is to reduce the frequency bandwidth needed to transmit such signals. Consequently, in the modulator the rectangular signal forms of the bit sequences are transformed into slow changing signal forms in so called pulse formers. This prevents an abrupt change to the phase of the modulated signal (see Figure 1.6).

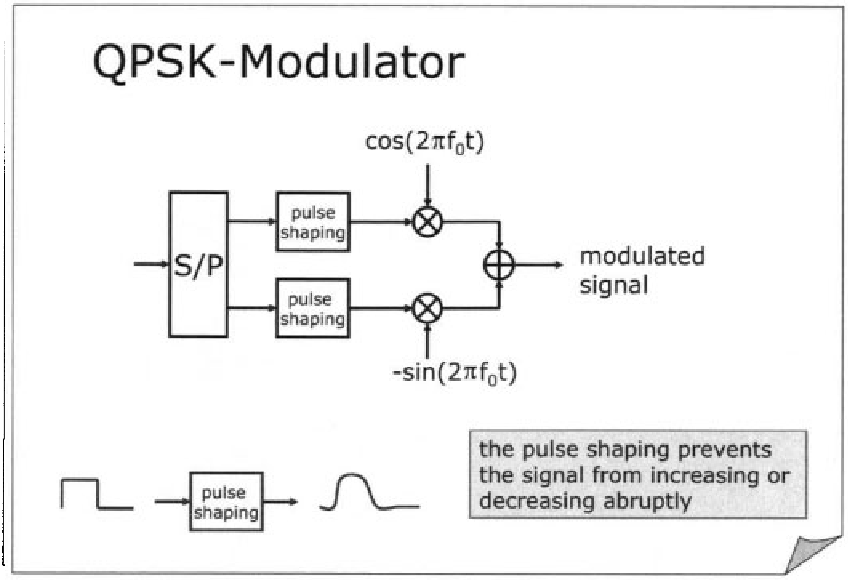

Figure 1.6: Pulse forming in QPSK modulator

A theoretically infinitely wide frequency spectrum is required for the transmission of rectangular signals. Only very few finite bandwidths are available for mobile radio telephone services. Therefore, the generated and formed pulses must have a theoretically infinite time spread.

Signals that are not time-limited inevitably overlay each other. As Figure 1.7 shows, interference-free data transmission can also occur with such infinitely long signals if the signals have zero values at intervals of one symbol duration. In this case, only one signal contributes to the sample result at each sampling instant because all the others have a zero value there.

It is clear from Figure 1.7 that the sampling instant for the represented signal form must be adhered to precisely if values from neighbouring symbols are to be avoided. Because such perfect synchronisation is practically impossible to achieve, signal forms are used that fade more quickly timewise and overshoot to a lesser degree. Since sampling at the wrong time means sampling the overshoots of the neighbouring signals this measure allows to tolerate errors in the sampling time without loosing much of the orthogonality of the signals. However, the quicker fading inevitably results in a widening of the signal spectrum.

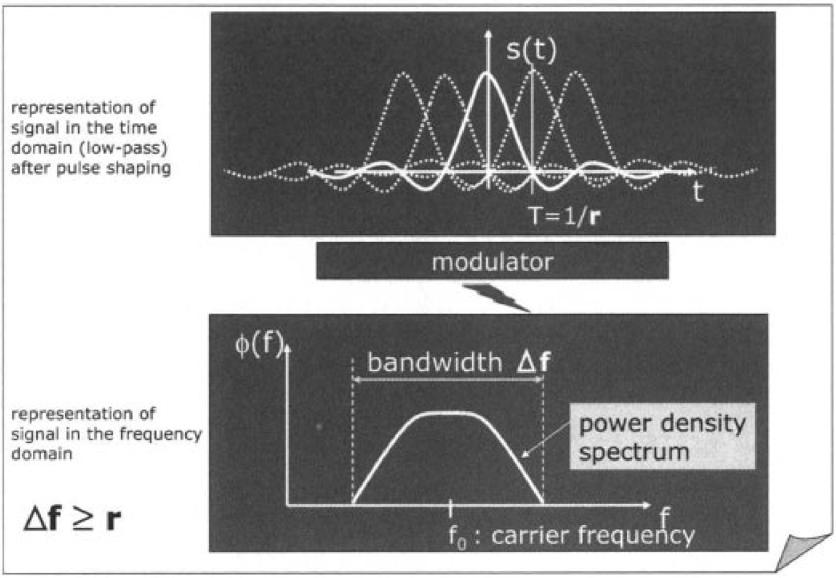

Figure 1.7: Signal representation of a band-limited signal

1.4 Noisy transmission

In practice, message transmission is never without interference. There are many reasons for such interference. It can either be due to natural causes (such as thermal noise in the receiver) or be system-related (such as co-channel interference in mobile radio systems, see Figure 1.8). The transmitted signal is thus overlaid by interference, and in some circumstances this can result in a false interpretation of the received signal and faulty recognition of the transmitted information in the receiver.

As protection from such transmission errors, channel coding procedures that add redundancy systematically to the information being transmitted are used. Stated in simplistic terms, more information than needed is transmitted systematically so that enough information remains despite transmission errors to enable the transmitted data sequence to be reconstructed in the receiver.

For an understanding of noisy transmission we consider the simple case of interference caused by additive white Gaussian noise. This sort of interference originates, for example, through the noise of an amplifier in the receiver. Figure 1.9 shows the receiver lowpass filter for a branch of the QPSK demodulator and subsequent sampler that samples the filtered signal for the period of a symbol duration.

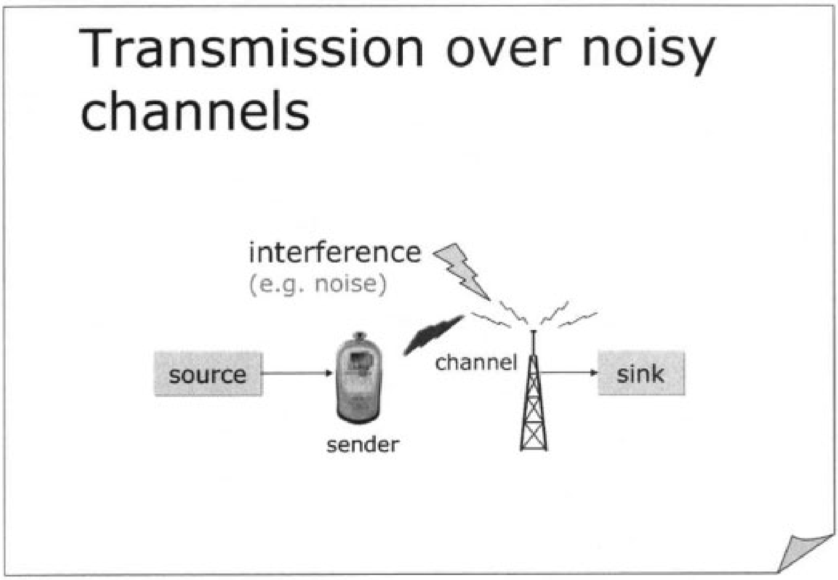

Figure 1.8: Transmission system with noisy transmission

For the bit transmission case described, which is noisy due to additive Gaussian noise, the probability density of the sampling values can be represented as shown in Figure 1.9. The sampling values for a transmitted "one" produce the value of a one, albeit coincidentally corrupted by the noise. In this case, the probability density therefore takes on the form of a Gaussian distribution with the medium value one. In the same way, the probability density for the sampling value for a transmitted zero produces a Gaussian distribution with the medium value zero.

In the receiver the sampler is followed by a decision stage that decides on a transmitted one or a transmitted zero based on the sampling value. In the simplest case scenario, this includes comparing the sampling value with the value of a decision threshold. If the sampling value is higher, it is assumed that a one has been sent; otherwise the decision is that it was a zero.

A transmission error occurs when either a one was sent but the sampling value is smaller than the decision threshold or, vice versa, if a zero was sent and the sampling value is higher than the decision threshold. Mathematically, the error probability for a non-recognised one corresponds to the area below the probability density for the sampling value of a sent one left of the decision threshold. The larger this area is, the greater the probability that a sent one is interpreted as a zero [17].

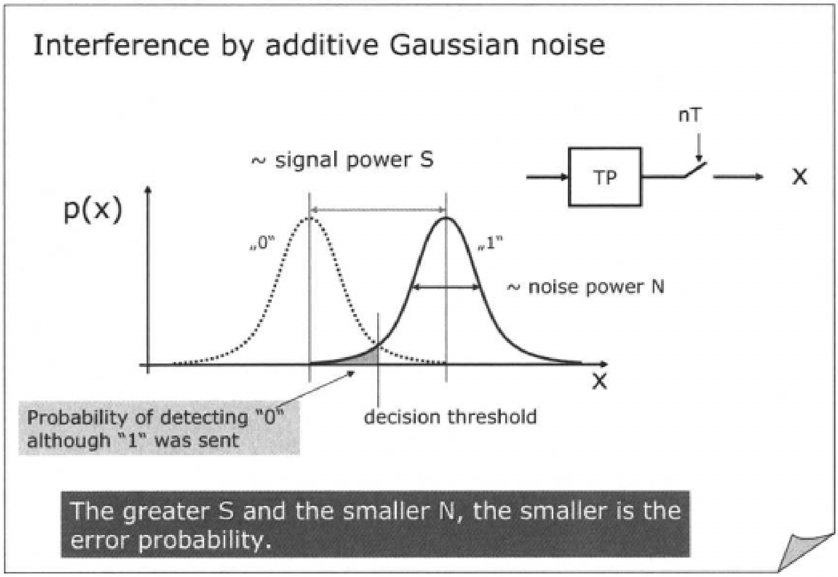

Figure 1.9: Origin of decision errors in the receiver

The distance between the two Gaussian distributions is proportional to the power of the received signal. This means that the greater the received signal strength, the smaller the chance of error frequency. The width of the Gaussian distribution is proportional to the noise power. This means that the greater the noise power, the higher is the error probability and vice versa.

Thus, the error probability increases proportional to the noise power and decreases proportional to the signal power. One measurement of the quality of a message transmission, i.e., for the error frequency, is therefore the ratio of the received signal strength to the noise power, also referred to as S/N ratio. The greater this ratio, the smaller the probability of transmission error; thus achievable throughput is proportional to the S/N ratio.

In network engeneering, the ratio between the received wanted carrier signal power and the sum of all received interference power is an indicator of the received signals quality. For the so-called carrier-to-interference ratio C/I applies the same as for the signal-to-noise ratio: the higher the wanted signal's power (carrier) and the lower the interference power the lower is the bit error probability, see Chapter 2.

If data transmission is only noisy due to Gaussian noise, then the transmission capacity of such a channel is dependent on the frequency bandwidth and the signal-to-noise ratio. The formula by Shannon for the calculation of channel

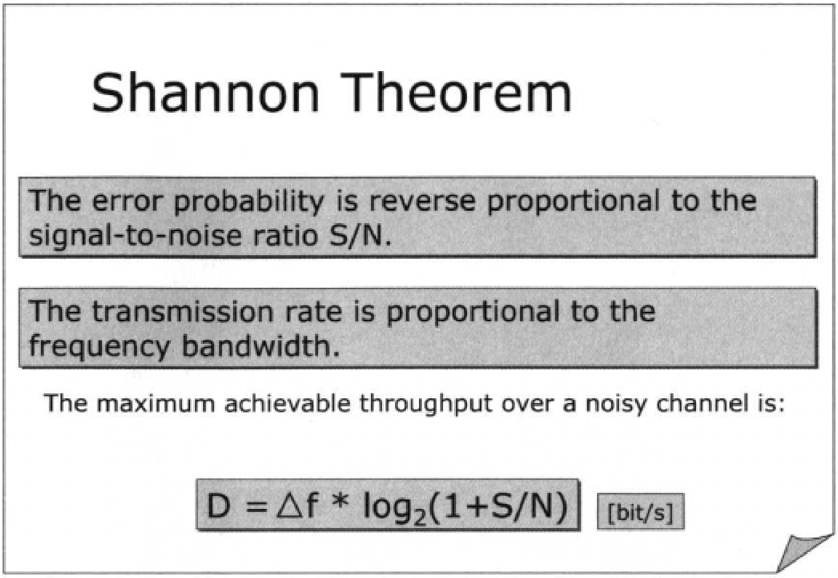

Figure 1.10: The Shannon theorem

capacity presented in Figure 1.10 shows that a lower S/N ratio can be permitted for a given capacity if there is an increase in the frequency bandwidth.

Since noise power increases with the bandwidth when constant noise power density exists at the output of a channel, channel capacity is also limited for an infinitely wide frequency spectrum. If the signal-to-noise power ratio is very low, i.e. if the noise power is much greater than the signal power, the channel capacity is zero.

2 Cellular Mobile Radio Networks

2.1 First-generation mobile radio systems

The first mobile radio systems were designed to accommodate only very few users. In 1946 in St. Louis, Missouri, USA, the first *Mobile Telephone Service* (MTS) was installed in a car. It used half-duplex and had only a very limited range [2]. MTS operated at 150 MHz using 6 channels. The downlink transmit power was around 250 W. In the 1960s, the system was upgraded to the *Improved Mobile Telephone Service* (IMTS), but the basic limitations remained the same.

In Europe, the situation was similar. The German A-Network was operational from 1958 to 1977 and was able to manage up to approx. 10,000 users. Calls were still being switched manually. The system operated in the frequency range between 154 MHz and 177 MHz, using frequency modulation with a channel grid of 50 kHz.

Later, the first real cellular systems were implemented, such as the analogue *Advanced Mobile Phone Service* (AMPS) system in the US. For the first time, frequencies were reused resulting in the interference inherent to cellular networks (see below). AMPS uses tone signalling and operates between 825-845 MHz and 870-890 MHz. The last AMPS networks are scheduled to be shut-down within the next 5 years [6].

The German B-Network started operating in 1972. This network also lacked the ability to automatically locate its users, although users were able to bypass the indirect mechanism of manual switching and make outgoing calls using self-dialling procedures similar to the ones used in AMPS. When the A-Network was shut down, the B-network was expanded through the inclusion of its former frequencies but in 1994 it also ceased operating. At its peak phase 25,000 subscribers were using the B-Network.

What was common to these systems is that they were able to provide coverage to a very large area using only one transmitter mast. In rural areas the radius of a coverage area supplied by a base station was up to 150 km, the transmitter power per channel was 20 W and higher. Due to the large radii of the cells, large areas could be supplied with mobile radio services despite minimal infrastructure (see Figure 2.1). Because of the limited number of available frequency channels, this kind of system could only serve a small number of subscribers. Mobile stations as well as base stations had to transmit simultaneously at high power in order to bridge the large distances. Therefore,

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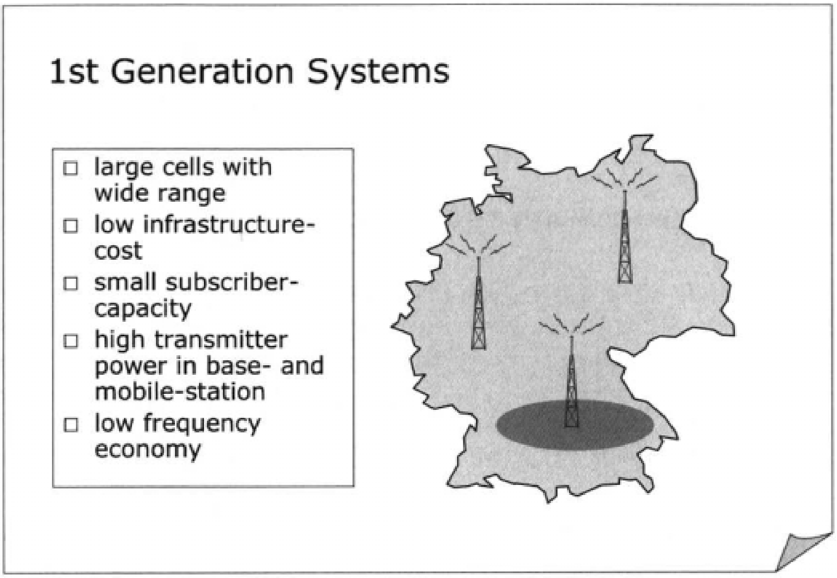

Figure 2.1: 1st generation mobile radio systems

handsets could not be implemented and terminals had to be built into the boot of vehicles. It was a real luxury to be able to make a mobile phone call [3].

2.2 The cellular concept

In May 1972 Bell Labs (today Lucent Technologies), the research subsidiary of the US telephone giant AT&T, registered a patent that laid the foundations for today's second and third generation mobile radio systems.

The idea is simple: instead of a single base station illuminating as large an area as possible, each base station should only cover a small area. In this case the antennas would not have to be erected as high as possible; consequently, the same frequency could be reused over relatively small distances. The spectrum can then be used many times in a given area, thus enabling coverage for a greater number of subscribers. However, a mechanism is needed that switches a user's connection from cell to cell. In addition, many more masts have to be erected than before, massively increasing the infrastructure cost (see Figure 2.2).

The cellular concept offers some important advantages: on the one hand, the transmitter power can be lower and this in turn results in smaller terminals

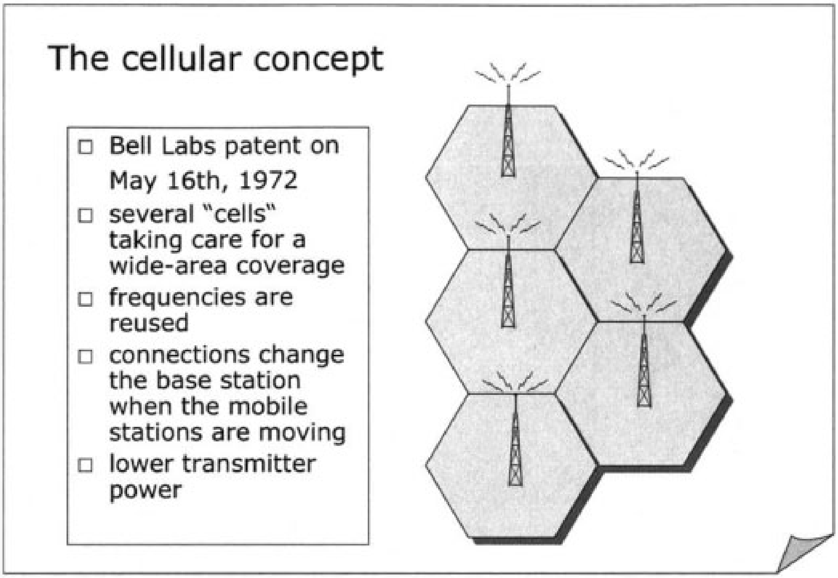

Figure 2.2: The cellular concept

and longer operating times. On the other hand, the subscriber capacity of such a network, i.e. the maximum number of active users per area element, is considerably higher due to the reuse of frequency channels. All modern mobile radio systems are based on this approach.

2.3 Frequency reuse and cluster formation

Reuse of frequency channels produces the following problem: if a frequency is used by more than one transmitter, interference occurs. This topic will be dealt with in detail below.

The group of cells that the spectrum allocated to a system makes full use of is called a *cluster* (the boxed-in area in Figure 2.3).

An example: a mobile radio network operator is allocated twelve frequency channels. The operator can distribute this spectrum among three cells. Each cell is then allocated four frequency channels. The next three cells also have to use the same spectrum. In this case, the cluster size would be 3. Alternatively, the network operator could distribute the spectrum among four cells, in which case only three frequency channels would be available to each cell.

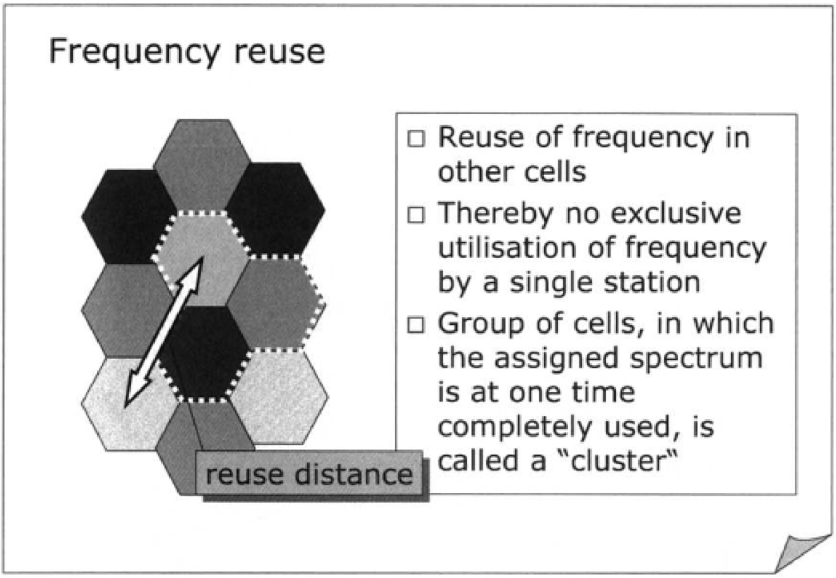

Figure 2.3: Frequency reuse and cluster formation

The distance between two cells that are allocated the same frequency is called the ***reuse distance***. The smaller the reuse distance, the closer the cell in which the frequency is reused. This cell is also called *co-channel cell.* The closer co-channel cells are located to a cell, the more they will cause interference to communication in the cell.

Network operators therefore have a considerable interest in making clusters as small as possible. The reason: the smaller the cluster is, the more frequencies are available per cell and the higher the capacity of the cell. On the other hand, interference increases as clusters become smaller and the quality deteriorates. Network operators therefore always have to find the right trade-off between capacity (clusters as small as possible) and quality (clusters as large as possible).

In reality, base stations are not equally distributed and therefore cells have different sizes. Furthermore, the traffic is higher in some cells than in others; consequently, the same number of frequency channels is not selected for all cells but capacity is allocated depending on subscriber density and topology of the area.

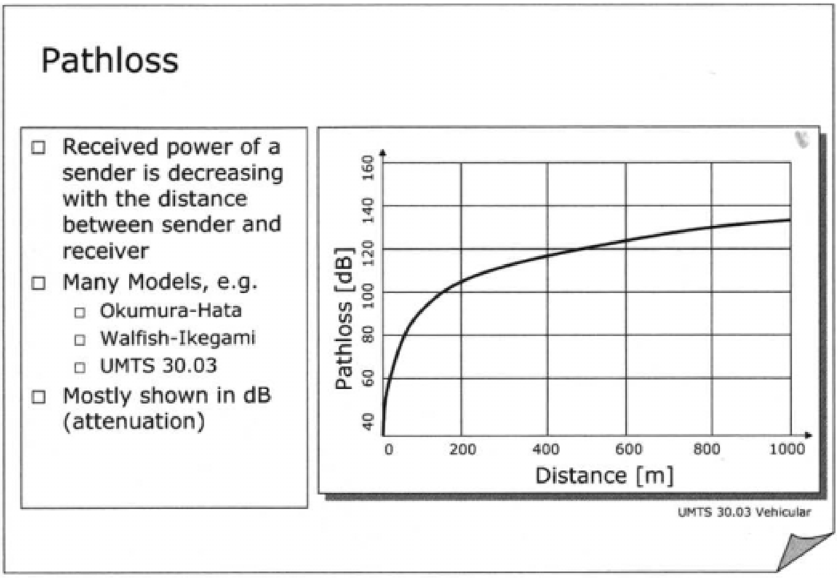

Figure 2.4: Propagation attenuation

2.4 Propagation attenuation

For the discussion that follows it is necessary to understand the relationship between the distance and the propagation attenuation of radio waves. The further away a receiver is located from a transmitter, the smaller the portion of the emitted power that arrives at the receiver (see Figure 2.4).

A number of mathematical models describe this dependency. The exact relationship depends on the frequency range, the type of antennas used, the condition of the environment, and so forth. The best-known model is the one by Okumura-Hata, which is valid for a frequency range of 500 MHz up to 1,5 GHz. The model is based on measurements and differentiates between two types of terrain and three types of buildings. Similar propagation models also exist for the frequency range at 2 GHz, which is relevant for UMTS.

Since propagation can vary in a wide range, the propagation factor is often represented logarithmically in decibels (dB).

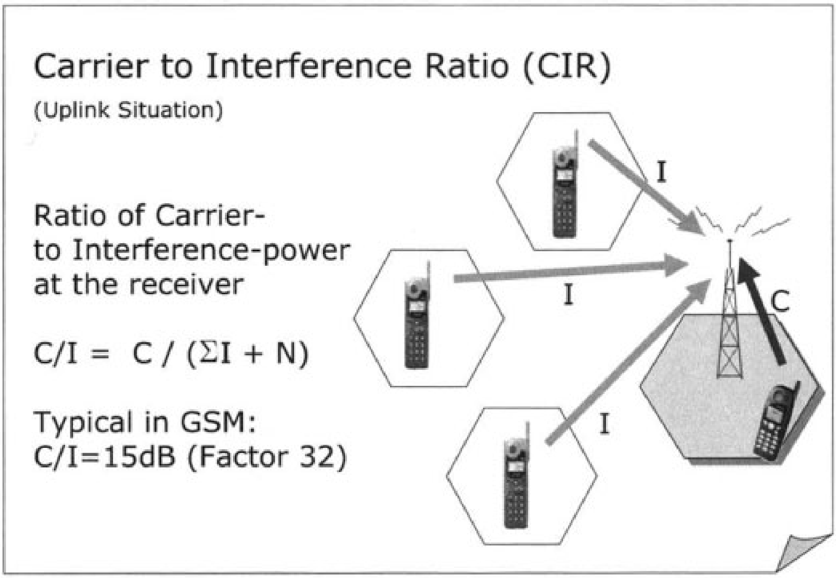

Figure 2.5: Signal-to-interference ratio

2.5 Interference and co-channel interference

By interference, we mean noise in a receiver caused because other senders other than the user that are also emitting energy in the same frequency band. Since co-channel cells always exist in cellular radio networks, interference is inherent to these networks.

The level of interference depends, among other things, on the distance between the receiver and the jamming transmitters. This is directly dependent on the reuse distance. If all subscribers use the same transmitter power, the level of interference only depends on the geometric constellation between the subscribers.

Interference is an unwanted contribution to the received power. In mobile radio systems power is likewise represented logarithmically, with the unit dBm, a logarithmic representation of the power related to 1 mW.

The interference in neighbouring co-channel cells increases as more users become active simultaneously in a network. Consequently, it is not only the number of channels available but possibly also the interference aspect that limits the number of concurrently active users in a mobile radio network.

The *Carrier to Interference Ratio* (CIR) is one of the most important dimensions in mobile radio networks. In Section 1.4 this dimension was also referred to as the S/N ratio. It is normally also indicated in decibels (dB) because in this representation a division becomes a simple subtraction.

The CIR is calculated as follows:

$$\frac{C}{I} = \frac{C}{(\sum_n I_n + N)}$$

The value C in the equation represents the carrier power occurring in the receiver. For example, a typical value for C in *Global System for Mobile Communications* (GSM) is in an average coverage situation about -78 dBm or converted as $\approx 1.5 \cdot 10^{-8}$ mW. This example illustrates that the received signal strengths in mobile radio networks of the second and third generations are very low and that receivers have to be appropriately sensitive.

The I in the denominator of the equation is the total interference I_n that occurs in the receiver from other stations. If one considers the uplink (mobile station sends, base station receives), the sources of interference are mobile stations that are active in other cells and transmit from there. The radio waves transmitted there are also received as interference at base stations outside the cell (see Figure 2.5).

On the downlink (base station transmits, mobile station receives) the sources of interference are other base stations that are also transmitting on the same frequency and the radio waves of which are being received by the considered mobile station. This shows that the CIR for the uplink of a connection can be different from the one for the downlink.

The CIR is limited even if no sources of interference exist. The term N represents the thermal noise in the receiver that practically always exists. In GSM thermal noise is typically below -115 dBm and therefore negligible in our example.

As a rule of thumb, receivers in GSM can receive with sufficient strength up to CIR from 8 dB. If the CIR values are any lower for a longer time, the connection is cut off. The 8 dB approximately correspond to a factor 6 in the linear representation. This means that the carrier has to be received at six times the strength of the aggregate of the interference signals. The minimum values for the CIR are dependent on many factors, such as the type of receiver used, the modulation method, and the channel coding.

The aggregate of the carrier signal of power C and the interference signal of power I arrive at the receiver. The higher the interference power, the more errors the receiver makes. So it is not the sum of C and I but the ratio between carrier and interference power that makes the difference, as already explained in Section 1.4. Therefore, the aggregate receive signal strength as indicated on the display by almost all 2G telephones used today is not necessarily a guarantee of interference-free reception.

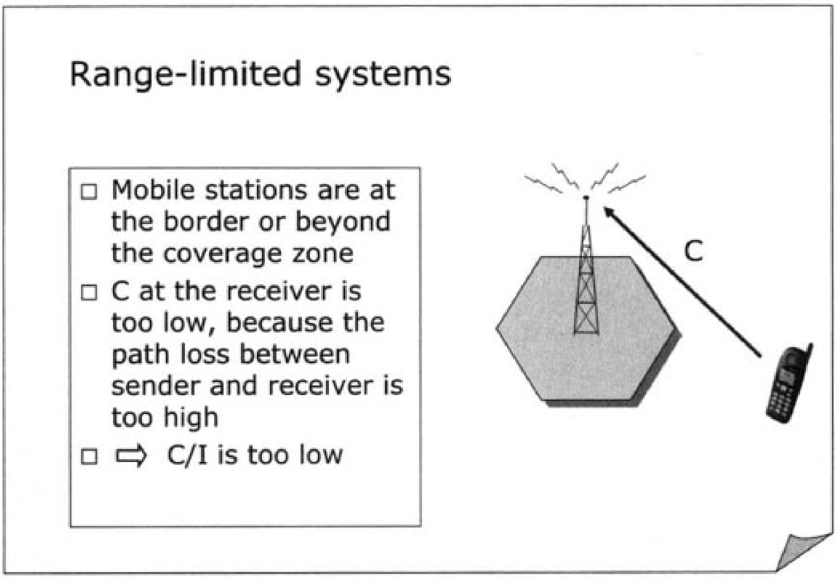

Figure 2.6: Range-limited systems

2.6 Range, interference and capacity-limited systems

When the points raised in the last section are applied to real systems, it is easy to find different reasons that can account for a poor radio connection.

The first case arises when a mobile station operates beyond the range of a cell (see Figure 2.6). This can occur on the cell boundary or, for example, if a mobile station is affected by shadowing. The received carrier power C is too low in this case and consequently the overall CIR is too low. The frequency of transmission errors becomes too high, and in turn, the connection gets noisy or even cut off. This situation can even occur if there is no interference at all, because the coverage areas of the cells are limited and the thermal noise in the receiver limits the range of the cells. This kind of system is called *range limited.*

The second case occurs when the received carrier C is sufficient but too much interference power is being received from other stations. In this situation the CIR again drops below a lower threshold and communication becomes disturbed. If the capacity of a network is determined by interference, the network is called *interference limited.*

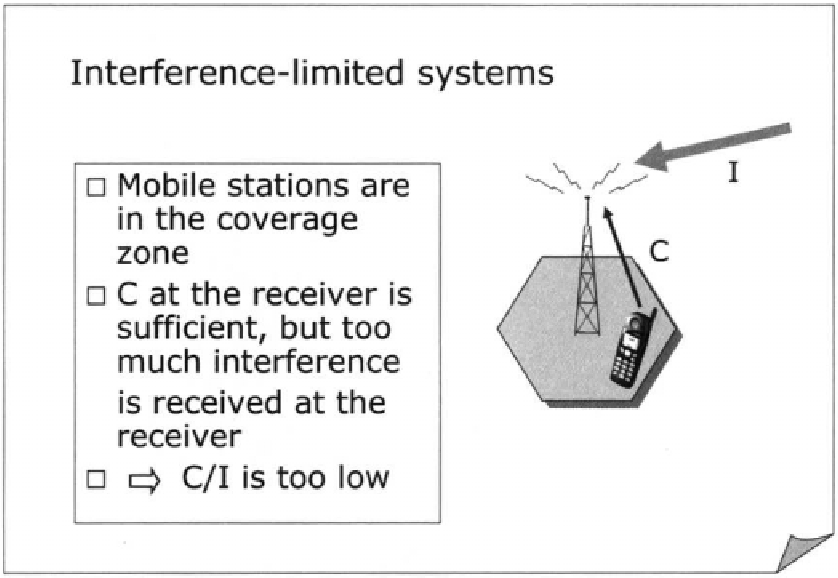

Figure 2.7: Interference-limited systems

Depending on the geometric constellation, the mobile and base stations can have a different CIR. Anyone who has used a mobile telephone will recognize the effect: one can hear the person on the other end, but one's own words are not reaching the partner. This situation is illustrated in Figure 2.7. Interference occurring at the base station is the reason why data transmitted by the mobile stations is not reaching the called party. In this example, the downlink could be totally free of interference.

The third situation in which communication is not possible is when all resources (e.g., channels) in a cell are in use. In this case, a connection is possible from the CIR point of view. However, the connection is rejected because no unused channels are available (Figure 2.8).

When a new connection is originating within a fully loaded cell, this connection is usually blocked. When this happens, the user usually makes another attempt to make the call a short time later. The typical dimensioning threshold of blocked calls tolerated in cellular networks is a 1–2% blocking probability.

A less favourable situation is one in which active users from neighbouring cells move into the respective cell. If no radio resources are available in this cell, the call is maintained as long as possible in the old cell. The reason why this is possible is because cells partially overlap each other. If the mobile station continues operating in the interior of the respective cell and no channel

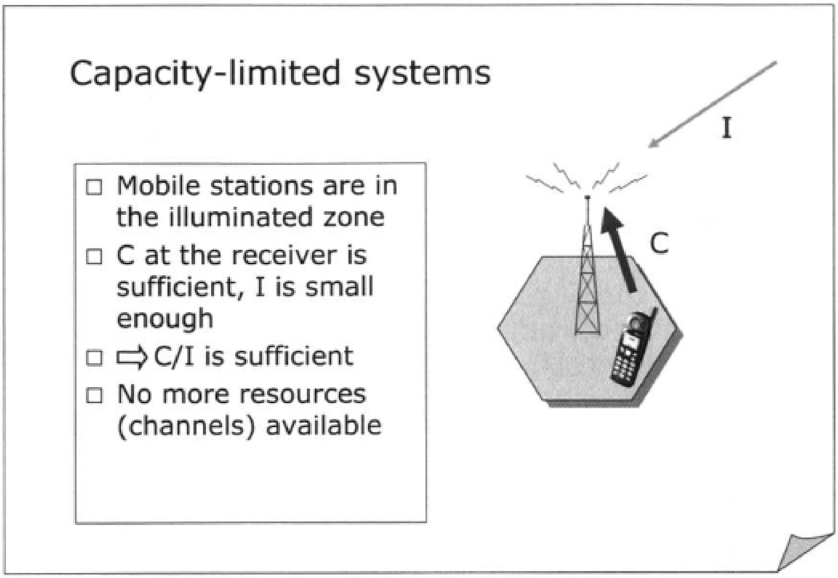

Figure 2.8: Capacity-limited systems

is available for a handover, the connection will disconnect. This is called a *dropped call.*

Because customers are more negatively affected by dropped calls than by blocked calls, network operators often reserve some channels in each cell for handover purposes.

When GSM was introduced in 1992, network operators focused on metropolitan centres. No radio coverage was provided outside these areas. Because so few users were active, the system did not experience any capacity bottlenecks and interference also was minimal. These were clearly ***range-limited*** systems.

Today, the GSM900 networks are often interference-limited. Although the networks provide good radio coverage, the high number of users create a high level of interference. Methods such as power control and frequency hopping can lower interference, thereby creating more capacity in a system.

The GSM1800 systems are often allocated considerably larger frequency spectrums and thus a larger number of frequency channels. At the same time the user numbers for these newer networks is lower. This enables the network operators to implement large clusters in order to maximise the signal-to-interference ratio and to minimise transmission errors. Interference in these networks is therefore not a capacity-limiting factor. The capacity limits are

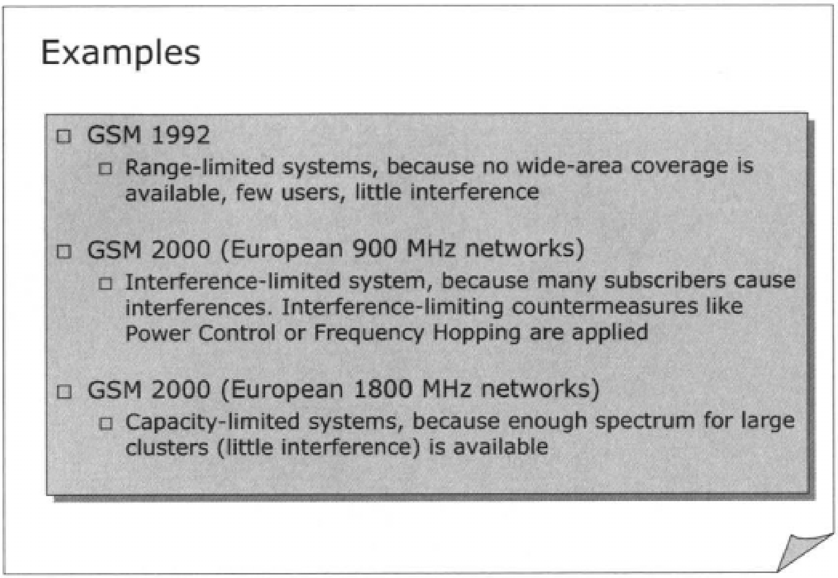

Figure 2.9: Examples of range, interference and capacity-limited systems

not reached until all channels in a cell are occupied, i.e., in this case these networks are capacity-limited.

The gateways between these boundaries are fluent and can also change.

2.7 Handover and location update

The smaller the cells of a cellular mobile radio system, the higher is the probability that a user will change cells during an active call. When the user moves across a cell boundary, the call has to be switched from one cell to the next. This procedure is difficult because the user should be unaware of the changeover. The cell change during an active call is also referred to as a handover or a handoff.

Automatic mobility management was not available in the first networks. A caller wanting to reach a mobile user had to know which region the user was located in and dial the corresponding dialling code. The AMPS network introduced automatic subscriber locationing. The network kept track of where the user was located and could automatically route calls to the right cell.

The mechanism that enables this tracking outside an active call is called location update. A group of cells is combined into a location area (see Chapter 10).

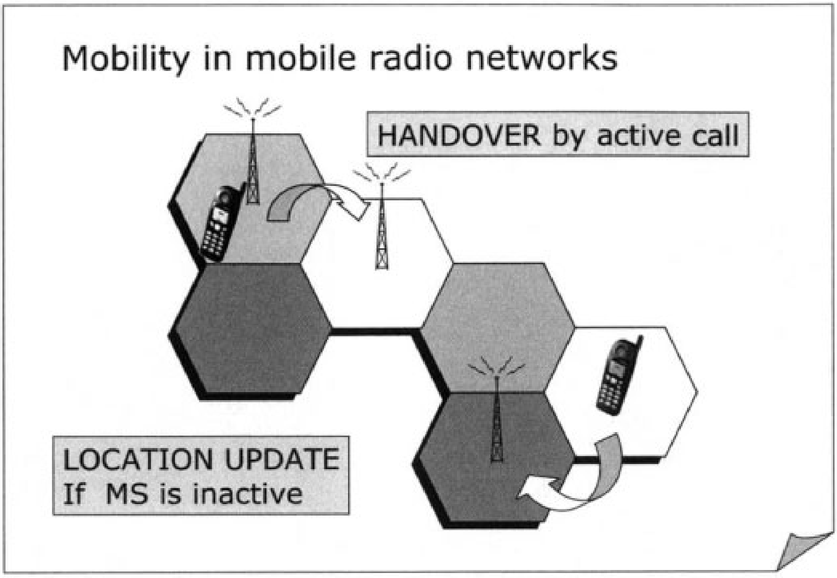

Figure 2.10: Handover and location update

Based on the system information transmitted by each base station, the mobile station detects the *location area* in which it is located. If it moves into a new *location area*, it registers with the network. The network in turn stores the new location of the user in a database so that incoming calls can be routed to the right cell or *location area*. Figure 2.10 shows both mechanisms, handover and location update.

These two mechanisms were essential for making the implementation of second generation small-cell mobile radio networks possible. Cell changeover procedures in UMTS are of an even greater significance since the aim in UMTS networks is to make the average cell sizes even smaller.

3 Standardisation and Spectrum

3.1 From 2G to 3G

The development of public mobile radio systems as we know them today was not a process that advanced in the same way all over the world. Instead the step from the early systems to the second generation was handled differently from region to region. This resulted in a profusion of incompatible systems (see Figure 3.1).

In the past, two main systems established themselves in the United States: *Time Division Multiple Access* (TDMA) systems IS-54 and IS-136 are based on a time slot structure (see Chapter 6) and are in part similar to the European GSM. *TIA Interim Standard 54* (IS-54) is a mixed TDMA/*Frequency Division Multiple Access* (FDMA) system with 30 kHz channel bandwidth. It was introduced by the *Telecommunications Industry Association* (TIA) in 1991 and was backward compatible to the old analog AMPS system, that brought mobile communications to the US in 1983. *TIA Interim Standard 136* (IS-136) evolved from IS-54 and is also called just TDMA or *Advanced Mobile Phone Service* (D-AMPS) on the market. It is a purely digital system, but still uses the channel bandwidth of 30 kHz introduced by AMPS. The main difference between IS-54 and acIS-136 is, that IS-136 uses TDMA also on the control channels. In December 2001, the number of mobile subscribers using IS-136 technology was 94.4 million worldwide according to figures given by [7]. This represents 10% of the worldwide subscriber base. IS-136 networks are mainly operational in North and South America, the Caribbean and in Asia.

The other systems are IS-95 systems that represent the first commercially operated *Code Division Multiple Access* (CDMA) systems. This transmission technology, which originates from military communication technology, also forms the basis for the radio interface in *Universal Mobile Telecommunication System* (UMTS) and will be looked at in detail in Chapter 6. With a channel bandwidth of 1.23 MHz, IS-95 systems are relatively narrowband systems and therefore are also referred to as *narrowband*CDMA (N-CDMA).

In addition to the US, IS-95 was also able to establish itself in South America, Central Africa and Asia. According to statistics published by the CDMA Development Group (www.cdg.org), over 90 million people used IS-95 systems to make calls in March 2001, of which more than 39 million lived in Asia and more than 33 million were in North America.

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff
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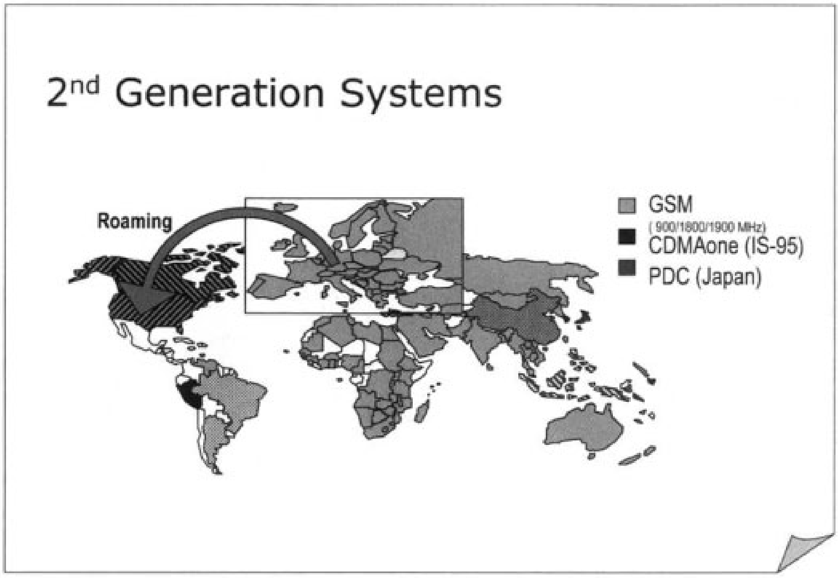

Figure 3.1: Worldwide distribution of 2nd generation mobile radio systems

Japan also developed its own standard: ***Personal Digital Cellular*** (PDC). PDC also uses TDMA technology (3 time slots, 25 kHz channel bandwidth) and operates at 800 MHz and 1500 MHz. The modern mobile telephones are small, sophisticated and offer long operating times. In June 2000 over 50 million Japanese people were using PDC. PDC-P is an enhancement that enables packet-switched data transmission with PDC at a transfer rate of up to 28.8 kbit/s. This technology is the basis for Japan's very successful i-Mode service, which offers access to Internet pages, emails and local information. Compared to other regions, Japan has a smaller distribution of Internet access than Europe or the US. Consequently, many subscribers use the service to call up information found elsewhere on the Internet. However, PDC had no success in expanding beyond the borders of Japan to other countries (see Figure 3.2). The i-Mode service, however, has been introduced in several European countries and is now competing against *WAP Next Generation* (WAP-NG) and MMS-based information services.

Probably the best-known system is one that originates in Europe and the use of which has spread from there to all parts of the world. *Global System for Mobile Communications* (GSM) was designed in the late 1980s by the state-owned national telecommunication companies and harmonised for use throughout Europe. The first systems started operating at 900 MHz (GSM900) in the early 1990s. This was followed by systems operating at 1900 MHz (GSM1900)

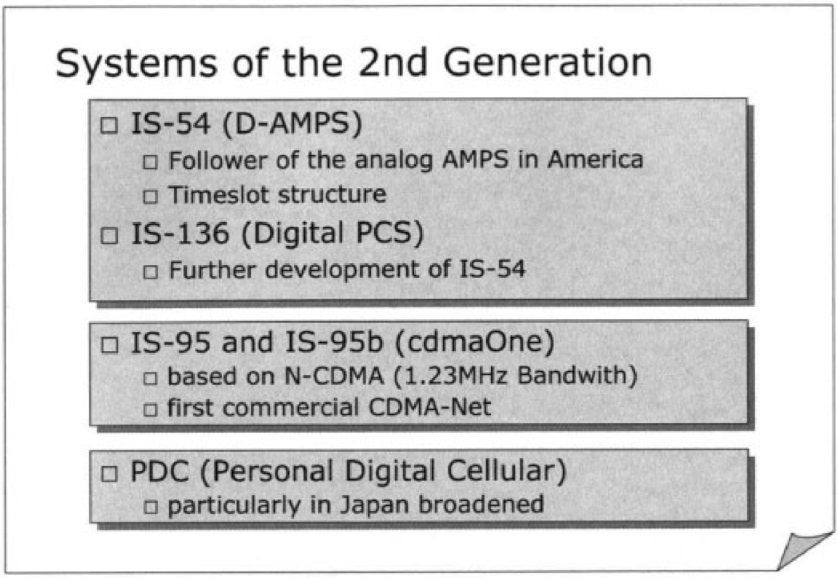

Figure 3.2: 2nd generation mobile radio systems

in America and 1800 MHz (GSM1800) in other counties. GSM also employs TDMA technology and uses 8 time slots on a 200 kHz wide carrier frequency. GSM900 has a total of 124 frequency channels and GSM1800 even has 374. GSM is used by over 400 operators in more than 171 countries in Europe, Asia, Australia, North and South Africa, and America. The projection of the GSM Association is that approximately one billion subscribers will be using this technology by the end of 2003.

These systems are currently competing for the mobile communication market. Each system incorporates its advantages and disadvantages, but one thing is common to all three: the systems were initially designed for narrowband speech telephony with bitrates between 5 and 15 kbit/s [24]. Now the emphasis is being shifted towards data services. Although the user numbers for wireless access to the Internet are still relatively low, this is an area where the next growth spurt is anticipated, especially considering that in some countries, more than 60% of the population are already using mobile telephones [33].

Even though multi-band and multi-mode devices are available, the different 2G systems are not compatible with one another, i.e., it is difficult and complicated to use different 2G systems worldwide.

If one looks at the reasons for the success of GSM, the main one is the open standardisation that was responsible for its initial success. Many of the ideas

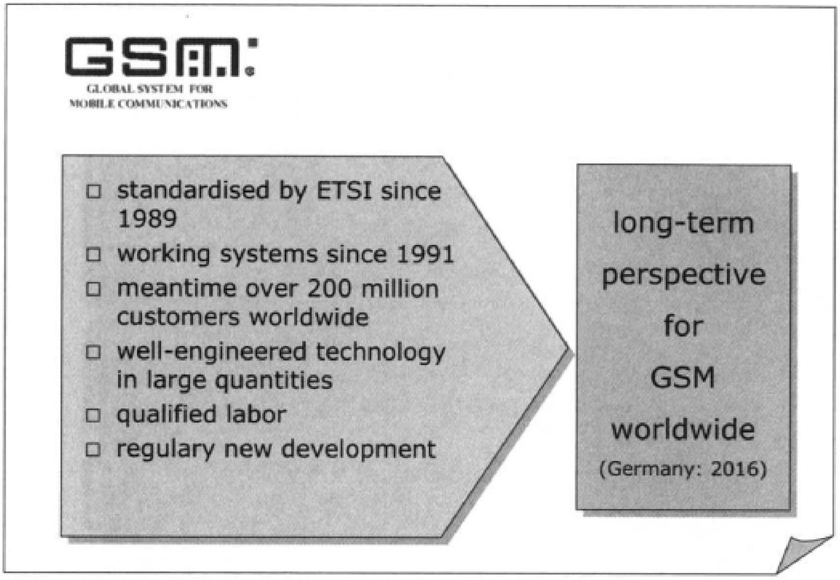

Figure 3.3: A perspective of GSM

it incorporates show an incredible vision that has kept the system open for further enhancement and development. This has enabled GSM to adapt to new developments without becoming incompatible with existing products.

Because of the early entry of GSM to Europe, there was also an early market for infrastructure and terminals. This resulted in cost reductions that in turn contributed towards GSM's rapid growth. Today, the technology is being produced in very high quantities and therefore is extremely cost-effective.

Due to the wide distribution of GSM, the number of qualified personnel with experience in the set-up and operation of GSM networks has grown.

Since its introduction to the market, GSM has continued to develop. The *Half Rate Codec* (HRC) increased capacity and the *Enhanced Full Rate Codec* (EFRC) improved voice quality considerably. Interference reduction methods, such as *frequency hopping* and *power control*, are being employed and with increasing traffic, network operators are introducing new hierarchical cell structures. The next segment attracting development will involve the evolution of data services.

These factors will ensure that existing systems will not be switched off because of the new third generation systems (see Figure 3.3). The opposite is the case: the plan is that GSM, IS-95 and PDC will coexist with their successor systems for a long time. In Germany, all 2G licences have been assigned until 2009 and some even until 2016.

Until 1995 the operation of GSM networks was concentrated on voice telephony. Use of the circuit-switched data service at 9.6 kbit/s and the fax service did not catch on to any noticeable degree until about 1996.

The data service was subsequently enhanced with *High Speed Circuit Switched Data* (HSCSD) that enabled channel coding to be adapted to the quality of the radio channel (9.6 kbit/s/time slot or 14.4 kbit/s/time slot) and permitted the bundling of several time slots. HSCSD is currently enabling data rates of up to 57.6 kbit/s (4 time slots per each 14.4 kbit/s). In Germany the service has been introduced by D2 Vodafone and E-Plus.

The next step will be the introduction of packet switching at the radio interface. The *General Packet Radio Service* (GPRS) protocol dynamically allocates a physical channel (time slot) to various users so that they can alternately transmit data. This process benefits from the typical characteristics of data connection and allows the existence of terminals that are essentially permanently linked to the network. GPRS also continuously uses *coding schemes* to adapt channel coding to the quality of the radio channel (CS1: 9.05 kbit/s, CS2: 13.4 kbit/s, CS3: 15.6 kbit/s, CS4: 21.4 kbit/s) and is able to use several time slots per connection. GPRS will allow a maximum of 171.2 kbit/s (8 time slots with CS4) to be achieved; in practice typical values are currently slightly over 30 kbit/s with 3 time slots per frame and CS2. All German network operators have introduced GPRS and, along with *Wireless Application Protocol* (WAP) over GPRS, are offering mobile Internet access over GPRS. Tariffs are normally based on volume but can also involve some time components.

EDGE is currently being standardised as a development of GPRS. Along with the channel coding, *Enhanced Data Rates for GSM Evolution* (EDGE) can also switch the modulation schemes at the radio interface between *Gaussian Mean Shift Keying* (GMSK)(Standard GSM) and 8PSK. The 8PSK modulation takes 3 bits to form a modulation symbol (see Chapter 1). This capability enables the transmission of up to 59.2 kbit/s with one time slot per time frame. Bearer services with a data rate of 384 kbit/s are being planned for EDGE. This makes EDGE very suitable as a gateway or alternative to UMTS. However, it is still not certain whether vendors will be able to provide EDGE-enabled infrastructure and terminals in sufficient numbers and whether there are network operators that want to introduce this technology. Since EDGE is closely related to GPRS, no problems are anticipated in this technology being mastered.

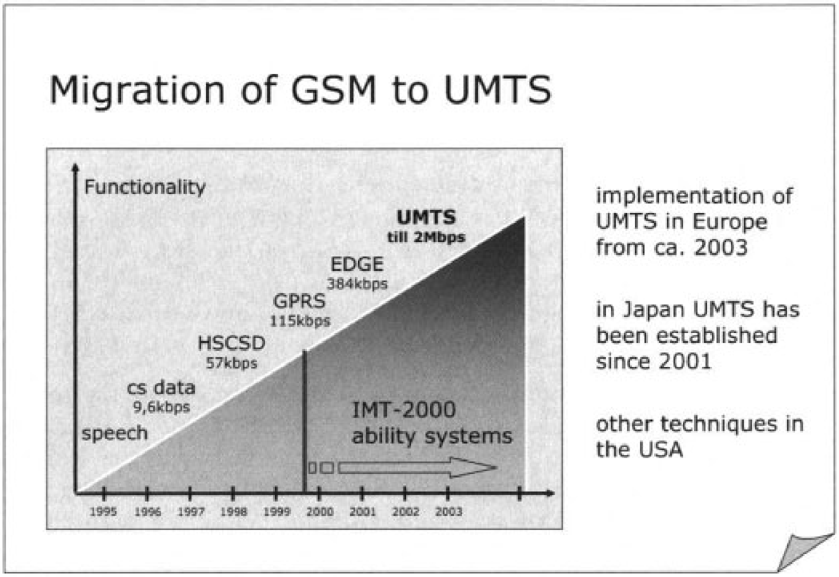

Figure 3.4: The route from GSM to UMTS

3.2 The IMT-2000 family

While coordinating the development of third generation systems, the ***International Telecommunications Union*** (ITU) defined a catalogue of requirements that specified what is expected of third generation mobile radio systems (3G). Figures 3.5 ff. list these requirements [13].

The emphasis is mainly on the requirements necessary for new kinds of data services: high data rates, efficient support of asymmetric traffic, packet-switched transmission at the radio interface and high spectrum efficiency [8].

Voice quality for the voice telephony services already available will be increased to the standard of the fixed network level. Moreover, the considerable existing investment in second-generation systems has to be protected, i.e., migration concepts are needed on how existing systems can be developed for the next generation. The development stages of 2G towards 3G such as GPRS are also called IMT-2000-enabled.

The ITU requested that the existing fragmentation into many different, incompatible systems should be resolved to provide a family of compatible systems. As a result, travellers would be able to have global access to different mobile services with a single device. Plans are underway to ensure that applications and service subscriptions by users in their local networks are available

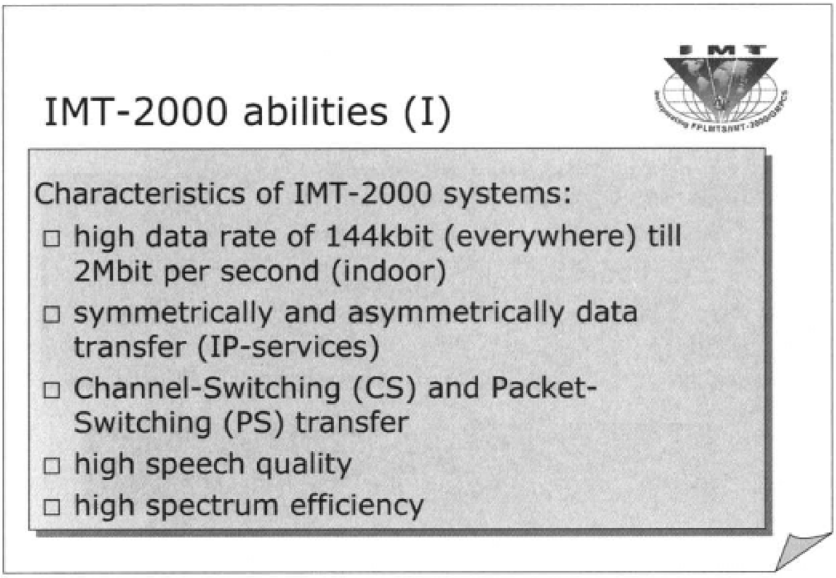

Figure 3.5: Characteristics of IMT-2000 systems

to them irrespective of the network they visit or which terminal they use. These applications will appear in the same way as usual in their home network. This concept of a *Virtual Home Environment* (VHE) is examined in detail in Chapter 10.

After the catalogue of requirements was drawn up (Figure 3.6), there were 15 different proposals worldwide on how a system based on these requirements should look. Ten of them related to the terrestrial segment; the other five were satellite systems. The system proposals were tested and evaluated by the ITU and finally six different systems were incorporated into the *International Mobile Telecommunications at 2000 MHz* (IMT-2000) family.

The six proposals for the terrestrial segment can be divided roughly into four categories (see Figure 3.7):

- W-CDMA systems: These include the *Frequency Division Duplex* (FDD) components of the UMTS standard in Europe and Japan as well as the US cdma2000.
- TD-CDMA: This group contains the *Time Division Duplex* (TDD) components of UMTS as well as the Chinese TD-SCDMA, which has now also been integrated into the UMTS-TDD mode.
- TDMA: As a further development of IS-136 and GSM, the UWC-136 system has been incorporated into the IMT-2000 family.

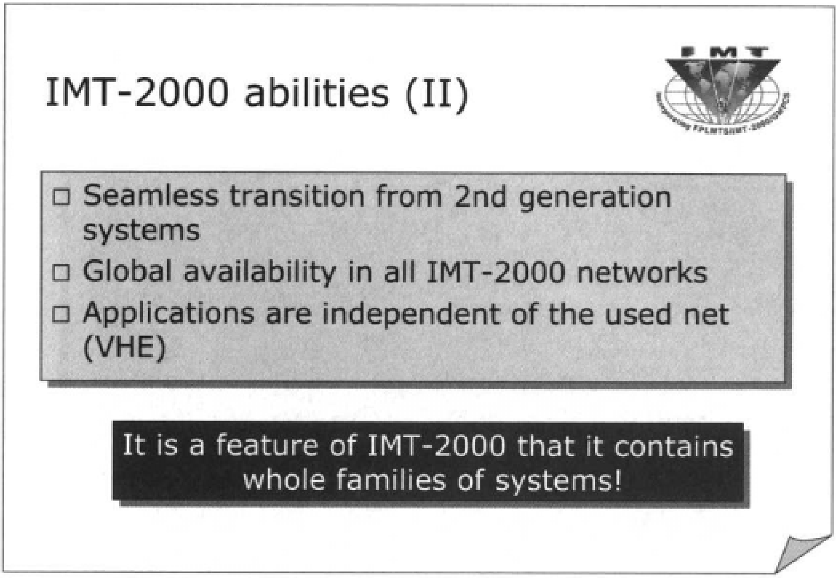

Figure 3.6: Characteristics IMT-2000 systems

- FDTDMA: The further development of the European cordless telephone standard *Digital Enhanced Cordless Telecommunications* (DECT) has also been adopted for applications with low mobility.

Thus IMT-2000 is a whole range of different systems (Figure 3.7) that have been closely coordinated with one another during the course of standardisation for the purpose of facilitating the development of multi-mode terminals. These terminals should function in different IMT-2000 systems and should guarantee global access.

The term UMTS auction has often been wrongly used in connection with the auction of third generation mobile radio licences in Europe. In fact, these are usually licences for the deployment and operation of UMTS/IMT-2000 compatible systems. This means that a network operator is not obliged to set up a UMTS network but certainly can select a different system from a range of IMT-2000 systems. The operator has to comply with some parameters of the regulatory authorities such as population coverage and spectrum issues, but otherwise has freedom of choice. This is especially true for the European Union, where EU law now avoids the narrowing of licences onto a single technology but rather encourages the competition between different standards. In other countries such as Switzerland, the operators are bound to the UMTS-standard for their 3G-licences.

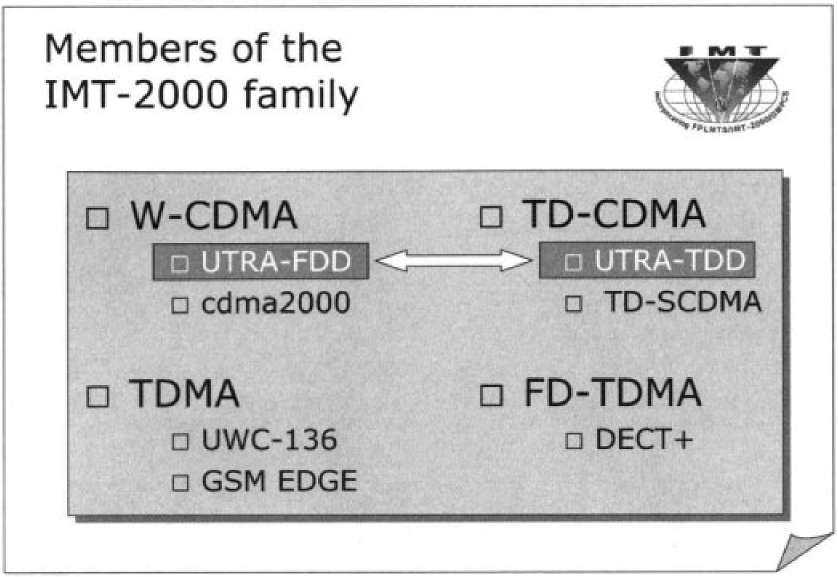

Figure 3.7: Overview of IMT-2000 systems

Since the GSM advancement EDGE is also a member of the IMT-2000 family through the UWC-136 path, German operator could definitely operate an EDGE network instead of UMTS in the new spectrum. The delays with UMTS mentioned below and the potentially smaller number of base station locations required with EDGE are aspects that support a thorough check of the options available.

Various migration scenarios to 3G systems exist based on the 2G systems introduced in the various regions of the world (see Figure 3.8):

1. TDMA (IS-136) has a large number of users in the USA and in other North and South American countries as well as in Asia. These users will mainly be utilising 3G services through UWC-136/EDGE. It is possible that en route packet-switched services through GPRS will also be introduced to offer higher data rates in the short term.

2. Packet-switched services based on GPRS are currently being introduced in a large number of GSM networks. To provide full-value 3G services, these operators can either develop the networks into full EDGE networks that work within the existing GSM spectrum or that use the new 3G spectrum. Still, the most probable path is that 2G-GSM networks will also be using WCDMA technology, thus UMTS, in the long term. The equipment vendors keep most of the migration paths open for the oper-

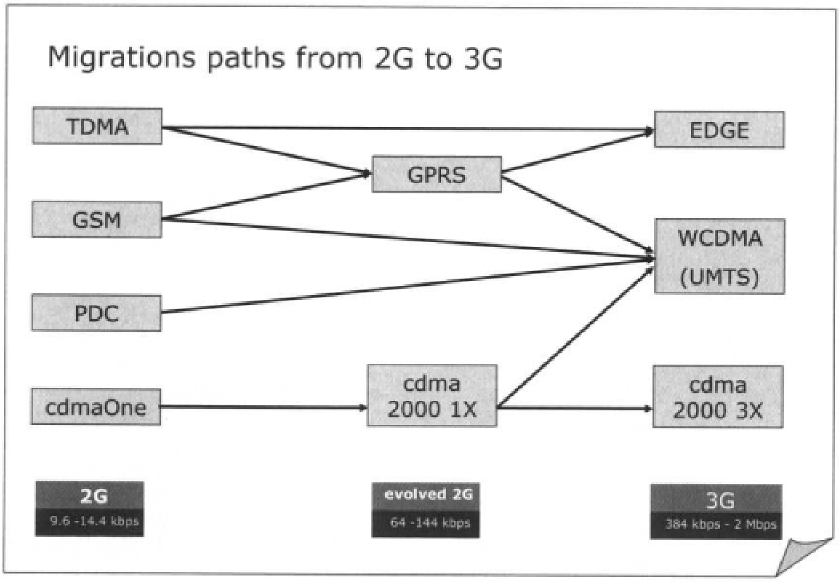

Figure 3.8: Possible migrations from 2G to 3G mobile radio systems

ators, since some modern base stations can support all three standards, GSM, EDGE and also W-CDMA/UMTS.

3. Japan was a trailblazer in the standardisation and introduction of W-CDMA. No 3G system other than UMTS is being planned in Japan. Currently, the number of subscribers in the so called FOMA is behind expectations, but data rates of up to 384kbit/s are available.

4. Networks already using N-CDMA based on IS-95 (cdmaOne) will be able to use higher data rates on the downlink (cdma2000-1X) as a first step. A later step is that these data rates will be increased even more and developed into cdma2000-3X. Although also UMTS is an alternative, the use of cdma2000 is more likely since cdma2000 is backward compatible with IS-95 and current 2G terminals can be re-used.

Because individual network operators have already invested large sums in existing networks, system decisions always depend on given requirements. It is even probable that in some countries all three technologies will be in use at the same time by different operators.

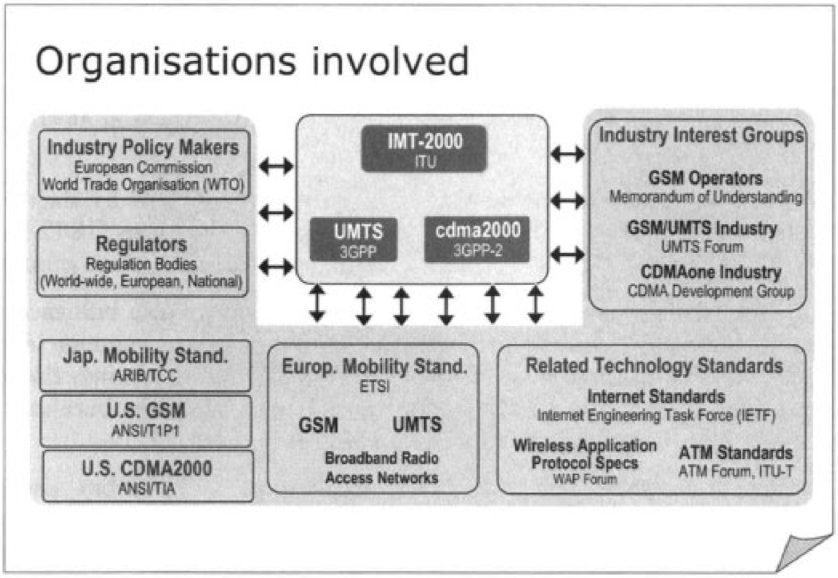

Figure 3.9: Overview of organisations involved in UMTS standardisation

3.3 Standardisation of UMTS

A large number of organisations, each pursuing their own goals and interests, are involved in the ongoing standardisation of 3G (Figure 3.9). Consequently, the standardisation is an extremely complex and sometimes also a political process, in which technical decisions are made in an environment full of different and contradictory interests.

Based on the parameters defined by the ITU, the *Third Generation Partnership Project* (3GPP) is currently standardising the UMTS system. The *Third Generation Partnership Project 2* (3GPP2) has taken over similar tasks for the *Code Division Multiple Access 2000* (cdma2000) evolution. Direct members of the 3GPP include the standardisation bodies of the different regions, thus the *European Telecommunications Standards Institute* (ETSI) (Europe), the *Association of Radio Industries and Businesses* (ARIB) (Japan), the *Chinese Wireless Telecommunications Standards* (CWTS) (China), T1 (USA), the *Telecommunication Technology Association* (TTA) (Korea) and the *Telecommunication Technology Committee* (TTC) (Japan). The large telecommunication vendors, operators and the regulatory bodies participate in the standardisation process via their membership in these regional organisations. The results do not flow back from the 3GPP directly to the ITU but through the regional bodies [22].

The 3GPP interacts with political bodies such as the *World Trade Organisation* (WTO) and the European Commission. At the same time the requirements of the local regulatory authorities, in Germany the *Regulierungsbehörde für Telekommunikation und Post* (RegTP), participate in the ongoing standardisation efforts, i.e. on issues such as spectrum coordination at country borders.

3G systems use many technologies that are already supported by their own standardisation groups. This includes the Internet technologies (IETF) and ATM (ATM Forum). The 3GPP also maintains contact with these groups.

Finally, lobbyist groups from GSM and UMTS operators also influence the standardisation activities. Their interest is motivated by the commercial impact of technical decisions. For example, the *UMTS Forum* pursues the idea of large frequency blocks for the operators of UMTS networks because this could help reduce the investment needed in infrastructure.

In the meantime the 3GPP is not only standardising UMTS but also has taken responsibility for the support of the GSM, GPRS and EDGE standards, which are used globally as well. Figure 3.10 shows the 3GPP structure as of autumn 2002. Within the 3GPP the *Project Coordination Group* (PCG) assigns existing tasks and resources to the five different *Technical Specification Group* (TSG). There the results of the various working groups (*Working Group* (WG)) are processed and adopted as *Technical Specification* (TS).

The *Technical Specification Group* (TSG) *Core Network* (CN) deals with the fixed network infrastructure, i.e. it is focused on the protocols and the distribution of tasks between the different network nodes. The *Technical Specification Group* (TSG) SA designs the system architecture and the mechanisms that can be used to provide services in the network. Also security issues and the management of 3G networks as well as source coding are addressed. The *Technical Specification Group* (TSG) *Radio Access Network* (RAN) is in charge of the radio access network, i.e., it deals with the radio interfaces, the fixed network protocols between the elements of the radio access network and the protocol stack at the radio interface. The *Technical Specification Group* (TSG) T handles the aspects of terminals that require standardisation. For example, this includes the interfaces over which a terminal communicates with a SIM card or other external units. An important aspect of this TSG is the conformance testing, which designs the test suits for harmonised testing of 3G terminals and sets the minimum requirements a terminal must fulfil. The *Technical Specification Group* (TSG) *GSM/EDGE Radio Access Network* (GERAN) formally has taken over responsibility for the standardisation and maintenance of the GSM, GPRS and EDGE standards.

Figure 3.11 shows the principal sequence of standardisation. The standardisation process of UMTS began with basic research [18]. Immediately after GSM was introduced, countries in the European Union researched technologies that seemed important for 3G systems. Within the framework of the *Research*

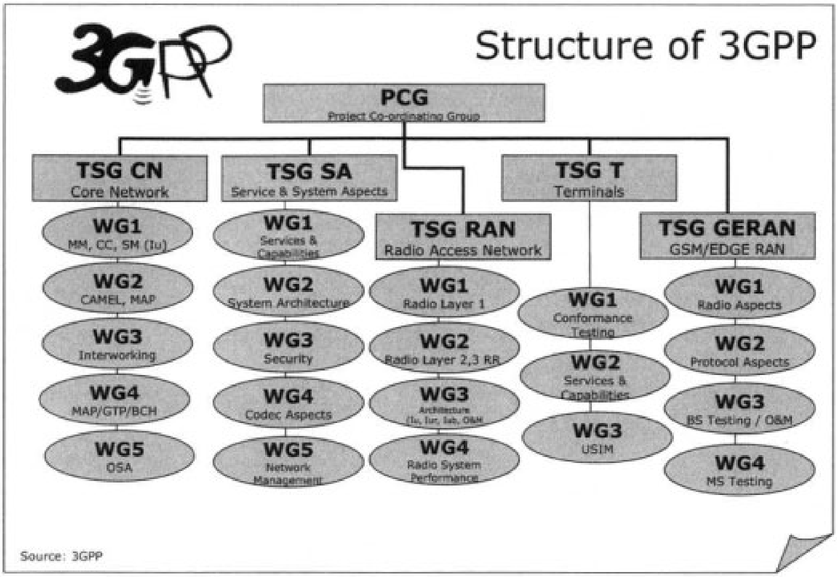

Figure 3.10: Organisational chart of 3GPP

Analysis, Communication, Evaluation (RACE)-1 programme, the *European Union* (EU) promoted basic research on cellular concepts, radio propagation, handover, dynamic resource allocation, modulation, coding, channel management and fixed network architectures for future systems. The programme ran from 1989 to 1992.

Based on the results of RACE-1, the RACE-2 programme followed from 1992-1995 focusing on the development of full-scale system concepts. This included a comparison of the CDMA and TDMA technologies by the *Code Division Testbed* (CODIT) and the *Advanced TDMA* (ATDMA) projects.

Subsequently to RACE-2, the fourth framework programme *Advanced Communications Technologies & Services* (ACTS) designed radio interfaces and evaluated them in terms of their performance. The *FRAMES Multiple Access* (FMA) subprojects of this programme again saw a competition between TDMA (FMA1, broadband TDMA technique without a splitting algorithm) andCDMA (FMA2: broadband CDMA technique).

Five candidates were evaluated at the end of the design phase. At the historic ETSI conference in January 1998 a decision was made on the system design ETSI would be submitting as an IMT-2000 candidate to the ITU. The ETSI members agreed on a compromise: The ETSI design was to have two modes, one based on the W-CDMA proposal and one based on the TD-CDMA design.

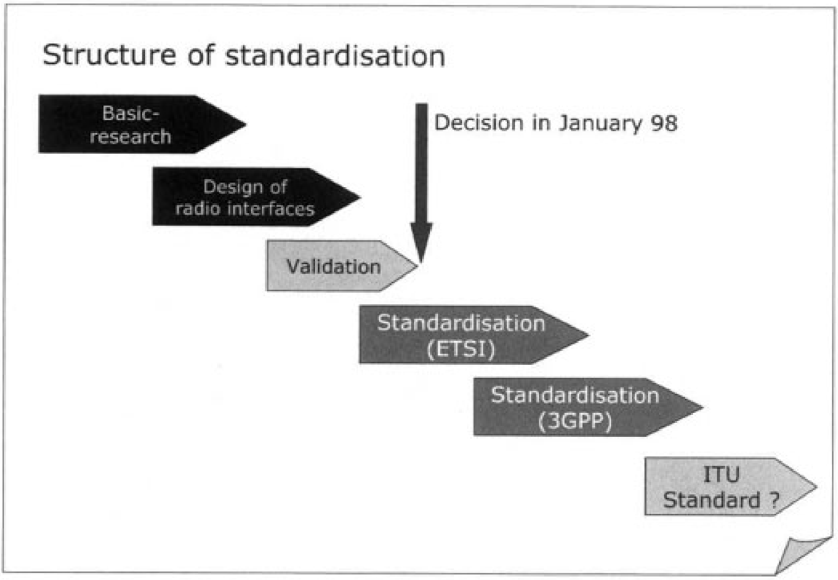

Figure 3.11: The chronological development of UMTS standardisation

These two modes are reflected in the ETSI submission to the ITU and today are the two UMTS modes FDD and TDD (see Section 7.3).

ETSI first standardised UMTS at the European level. The other international standards bodies, above all ARIB in Japan, simultaneously developed comparable standards. In December 1998 the various activities were bundled into the 3GPP to ensure coordination of the standardisation efforts. Modifications of certain parameters were also carried out in cooperation with other organisations to ease harmonisation between different ITU family members. For example, the initial chip rate for UMTS of 4.096 MChip/s was reduced to 3.84 MChip/s to enable simpler and more cost-effective *Application Specific Integrated Circuit* (ASIC) designs usable both for UMTS and for cdma2000.

The ultimate aim of all these standardisation activities is worldwide recognition of a set of standards with a high level of acceptance so that the vision of global mobile access can become a reality and in turn spawns a huge market for 3G services.

The far-sighted decision of researchers and companies in Europe to start research for 3G immediately after the introduction of the first GSM systems is remarkable and allowed Europe to play a leading role in mobile communications for 2G and 3G.

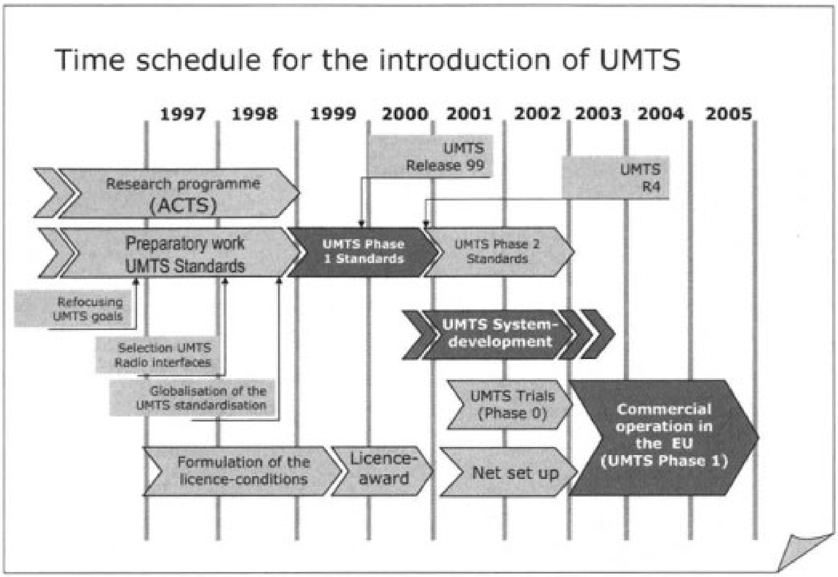

Figure 3.12: Timetable for the introduction of UMTS

3.4 Timetable for the introduction of UMTS

Figure 3.12 shows the current timetable for the introduction of UMTS. The licences have already been assigned in most European countries and most operators are currently involved in building the networks. One of the biggest problem is finding suitable sites for the base stations. Due to the smaller cell radii, the number of sites required is significantly higher, especially in well-populated areas. Still worse, the population is currently showing massive concerns about health implications from radiation that would be emitted as a result of the erection of even more antennas. The battle is sometimes fought on a very emotional level and has, in some places, caused serious problems in the aquisition of new antenna sites. In some regions, this could even delay the planned commercial start of UMTS.

Throughout 2000 and 2001, regulatory bodies in Europe issued licences for the deployment and operatation of 3G networks. In some countries, the licences were issued by means of an auction. The enormous sums paid by the bidders reached US$ 35 billion in the UK and US$ 45.8 in Germany. In other countries like Switzerland, the licenses were given away for some 30 million US$. As a result these operators have a significantly higher chance to see a return of investment in a reasonable amount of time. In consequence, the euphoric perspective for 3G systems changed somewhat and several operators already announced that they could not find the additional money needed to deploy the network infrastructure for 3G.

To ease the burden of the high licence fees, the Germany ***Regulierungsbehörde für Telekommunikation und Post*** (RegTP)[Regulatory Body for Post and Telecommunications] gave the licensees permission to share the use of antenna sites (called ***site sharing***) when building their networks - so long as they retain authority over their networks and services. This could reduce the number of sites needed considerably, at least in the lead-in phase of 3G. There is talk about related cost savings of 20 per cent and more.

Parallel to these activities, the UMTS standard is undergoing further development. Whereas until now activities have focussed on the radio interface and the basic system architecture, efforts are now being concentrated more on the mechanisms required to provide sophisticated services (such as OSA, SIP, see Chapter 10).

Some operators were planning to commence commercial operations in time for CeBIT 2002. However, it proved that neither terminals nor sufficient infrastructure was available by then so a network start of late 2002/early 2003 appears more realistic. The first terminals commercially available in Europe are expected to arrive throughout 2003.

Because of the enormous development and licensing costs, the new technology must be introduced as quickly as possible. Due to the high level of complexity involved, it is not possible to anticipate all the technical problems that can occur, so vendors and network operators are constantly fighting delays in the tightly staggered timetables.

3.5 Release 99, Release 4 and Release 5

Like GSM, UMTS standardisation and deployment take place in several stages. The first stage, which will serve as the foundation for the first systems in Japan and Europe, is also referred to as Release 99, abbreviated R99 (Figure 3.13). This stage does not yet include the availability of all options eventually planned for UMTS.

UMTS R99 contains a fixed network similar to the one in GSM Phase 2+. The new UMTS radio access network represented by high data speeds and mechanisms for quality of service control is attached to this network. The data speeds will initially be comparable to those of ISDN (64-128 kbit/s), whereas new service platforms such as *Open Service Architecture* (OSA) and the *Virtual Home Environment* (VHE) will not be available until later.

The new SIM cards for user identification, called USIM in UMTS, will be available at the time of system start. Operators who are already operating GSM networks and who now are migrating to UMTS have to extend the

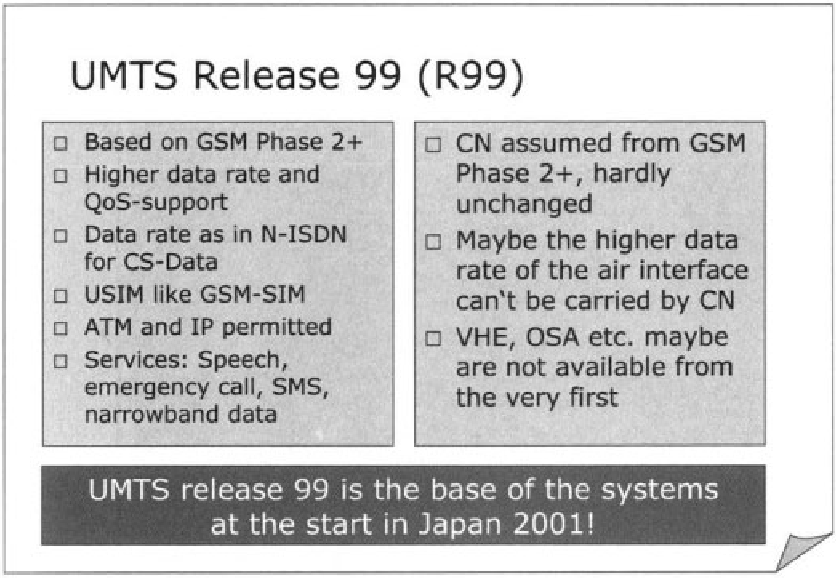

Figure 3.13: UMTS Standard, Release 99

capabilities of their fixed networks at the same time as they are building the new radio networks.

The first terminals available will be based on R99 but they will not yet offer the expected high data speeds of 2 Mbit/s, but will more likely offer 128 kbit/s. Moreover, these data speeds are only achievable in special cases (see Sections 8.2.2 and 11.6). Announcements from terminal vendors indicate a talk time of about 2 hours and an increased size when compared with small 2G terminals. The reason for this is the larger display, but also the larger space needed for the battery due to the higher power consumption.

Figure 3.14 presents the structure of the UMTS standard R99. The first column shows the categories into which the standard is divided. The next column has the familiar numbering of the GSM standard prior to Release 2000. The integration of the GSM standard into the domain of the 3GPP has also caused a change in the numbering: for Phase 2+, release 4, of the GSM standard, the old GSM numbers are increased by 40 before the comma, there are now three digits after the comma. Thus, the standard GSM 01.0x now becomes GSM 41.00x.

The UMTS standards correspond to those of the GSM standard but in each case are 20 numbers lower in the document numbering. The places after the commas are also three digits. So as of now there is only one series of

Structure of the standard R99

	GSM before R4	GSM after R4	UMTS R99+
Requirements	01.xx	41.xxx	21.xxx
Service aspects	02.xx	42.xxx	22.xxx
Tech. Realization	03.xx	43.xxx	23.xxx
Signalling (UE-NW)	04.xx	44.xxx	24.xxx
(U)TRA aspects	05.xx	45.xxx	25.xxx
CODECs	06.xx	46.xxx	26.xxx
Data	07.xx	47.xxx	27.xxx
Signalling (RSS-CN)	08.xx	48.xxx	28.xxx
Signalling (intra-FN)	09.xx	49.xxx	29.xxx
Management	10.xx	50.xxx	30.xxx
UIM/SIM	11.xx	51.xxx	31.xxx
O&M	12.xx	52.xxx	32.xxx
Security Aspects	13.xx	53.xxx	33.xxx
Test Specs			34.xxx
Security Algorithm			35.xxx

Figure 3.14: Overview of the structure of UMTS Standard, Release 99

documents. The series 21-35 deal with UMTS and the series 41-53 encompass the newer GSM standards.

The version numbers 3.x.y of the document series describe R99. As it became evident that the successor version R2000 would not be completed on time in 2000, a decision was made in September of that year to change the numbering scheme and split R2000 into two parts: Release 4 (R4) and Release 5 (R5). Consideration is currently being given to renaming R99 as R3. R4 comprises the version numbers 4. x. y, and version numbers 5. x. y are being provided for R5. For June 2003, Release 6 of the 3G standards is planned.

The further development of the standard is implemented in Releases 4, 5 and 6 (Figure 3.15). The following summary will give an impression of the architectural changes of the different releases. A more detailed and up-to-date summary is available on the website of 3GPP.

In R4 the ATM connections in the fixed network are to support quality of service control, i.e., *Quality of Service* (QoS) control will not only be supported at the radio interface but also in the fixed network. In addition, execution environments such as *SIM Application Toolkit* (SAT) and *Mobile Execution Environment* (MExE) as well as the service architecture OSA will be developed to enable them to provide new services. Further support for Location Services is integrated and to improve speech quality, certain calls can be connected through network without speech transcoding.

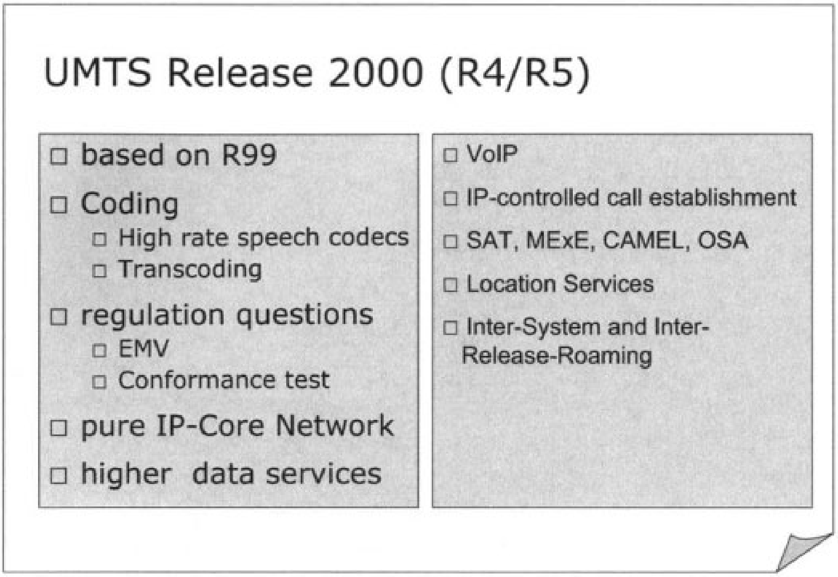

Figure 3.15: Further development of UMTS Standard

The Chinese proposal for IMT-2000, *Time Division - Synchronised Code Division Multiple Access* (TD-SCDMA), is integrated in R4 as a *Low Chip rate Option* (1.28 Mchips/s) in TDD mode.

A totally new fixed network concept is being planned with R5: the previous architecture, which is similar to the GSM fixed network, could be replaced by a completely *Internet Protocol* (IP)-based fixed network. This architecture, discussed in Chapter 4, requires a full set of new protocols in the fixed network. Additionally, also IP-transport within the RAN is considered. Other evolutionary steps in all other segments of the standard are planned.

The features of Release 6 are not fixed yet. Support of *Multiple Input Multiple Output* (MIMO) antenna systems can increase the radio channel capacity in rich scattering environments, i.e. in indoor environments. The algorithms for radio resource management across the areas controlled by a single radio network controller and between GSM and UMTS will be improved. The interworking between wireless LANs and UMTS as well as Hiperlan/2 and UMTS is addressed. On the service side, *Multimedia Broadcast/Multicast Service* (MBMS) and *Digital Rights Management* (DRM) are added.

It should be noted, however, that it takes some time after the specifications of a given release are frozen, until network elements and terminals supporting the new features become available. Some features of Release 6 are still far ahead.

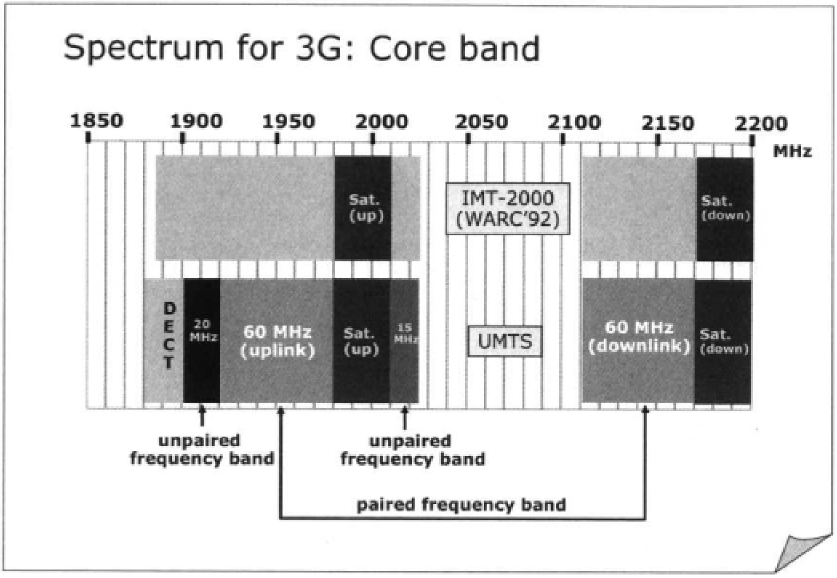

Figure 3.16: Frequency spectrum for 3G systems

Because R4 and R5 will entail changes to systems architecture, in practice UMTS networks based on R99 (or R3) will coexist with systems based on R4 and R5. In addition to Inter-Network-Roaming (in which a user changes over to another UMTS network of the same type), it is therefore also necessary to offer Inter-System-Roaming (user changes to GSM [2G] or to another IMT-2000 system [3G]) and Inter-Release-Roaming (user changes to another UMTS network with a different version release).

3.6 Frequency spectrum for UMTS

At the *World Radio Conference* (WRC) 1992 the frequency range shown in Figure 3.16 was reserved for third-generation systems. A total of 175 MHz for terrestrial systems is available from 1880 MHz to 1980 MHz, from 2010 MHz to 2025 MHz and from 2110 MHz to 2170 MHz. Additionally, two segments with 30 MHz each are available for satellite-based systems [9]. For comparison, in Europe in total 220 MHz have been reserved for cellular 2G systems. These consist of 2 x 10 MHz from 880-890/925-935 MHz (UL/DL) for *Enhanced GSM* (E-GSM), 2x 25 MHz from 890-915/935-969 MHz for GSM900 (P-GSM) and 2x 75 MHz from 1710-1785/1805-1880 MHz for GSM1800.

In Europe, parts of the spectrum are already occupied by existing 2G systems. For example, 1880 MHz to 1900 MHz is the frequency band used for digital cordless telephones operating in accordance with the DECT standard. Directly below DECT operates the GSM1800 system.

UMTS systems use a channel bandwidth of 5 MHz. A total of seven *unpaired* channels are available within the bands 1900 MHz to 1920 MHz and 2010 MHz to 2025 MHz. This means that one 5 MHz channel has to implement the transmission direction mobile station-base station (uplink) as well as the opposite direction base station-mobile station (downlink) (see Section 6.4).

In the range 1920 MHz to 1980 MHz 12 channels of paired spectrum are available, i.e., for each 5 MHz channel in this band, another channel between 2110 MHz and 2170 MHz exists. These bands are called *paired* bands (see Section 6.3). The paired satellite bands have not yet been allocated to a system or even operator.

In the licence allocation, up to six licences were sold for systems operating in the paired bands. This means that each of the six licensees may use 10 MHz for the uplink between 1920 MHz and 1980 MHz as well as the 10 MHz in the corresponding downlink range between 2110 MHz and 2170 MHz.

In addition to these frequencies, licences were issued for one of the seven unpaired 5 MHz blocks. Asymmetric Internet services are to be offered eventually in this additional spectrum. In certain areas, some of the unpaired blocks are also foreseen for unlicensed use with applications similar to cordless telephony.

In Japan the 1900 MHz to 1920 MHz band is already occupied by *Personal Handyphone Service* (PHS), which means this frequency range is not available for 3G. In the US the spectrum is already being used by a variety of 2G systems, including GSM1900. Consequently, it will not be easy for the European or Japanese 3G systems to be operated in the US. No definitive allocation has yet been made although the government has instructed the regulatory authority to deal with this problem as quickly as possible.

The 3G spectrum should be allocated by mid-2002, but is still delayed. Three bands are currently being discussed: 698 MHz to 960 MHz, 1710 MHz to 1885 MHz and 2500 MHz to 2690 MHz. As a result of the events of 11th September 2001, relocation of the military and governmental systems using the spectrum between 1710-1770 MHz and 2110-2170 MHz could be delayed resulting in a further delay of the introduction of 3G into the US [1].

Figure 3.17 shows the existing allocation of spectrum for 2G systems as well as the allocation of terrestrial 3G systems in Europe. The plan is to use additional extended bands as of 2005. In this connection the UMTS Forum is discussing the advantages and disadvantages of existing candidates. The band that looks most probable at this point is the one from 2520 MHz to 2670 MHz. Although other systems are operating in this spectrum, it would be relativly easy to transfer them to other bands. This block of another 150 MHz would

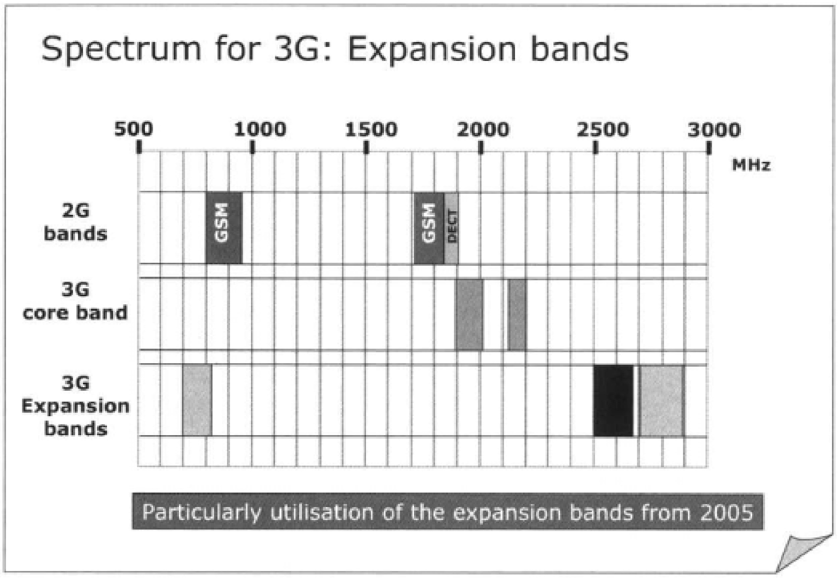

Figure 3.17: Extended spectrum for 3G systems

enable the implementation of most of the planned applications and is located close to a second candidate band (2700 MHz to 2900 MHz) that could also be used for 3G during a third stage (also see Chapter 11). Still, relocation of existing systems into another band is a difficult, costly and enduring task and might well take much longer to complete than until 2005.

The various bodies at the same time are already contemplating making use of existing 2G spectrum for 3G systems. Thus 3GPP is already standardising UMTS for the operation in the bands previously reserved for GSM. Thought is also being given to the operation in television bands below 800 MHz.

Studies undertaken by the chair for communication networks at Aachen University of Technology indicate that it may even be possible under certain conditions to operate narrowband 2G systems such as GSM together with broadband 3G systems such as UMTS in the same band. If this were the case, network operators could either also operate UMTS in their existing GSM band or, instead of UMTS, use GSM/EDGE technology in the new 3G bands. This could possibly result in a more cost-effective way to build and develop the networks.

3.7 Questions

3.1 Which second-generation systems are being used worldwide?

3.2 Why is the i-Mode service in Japan more successful than WAP in Europe?

3.3 List three reasons for the great success of GSM.

3.4 Outline the development stages from GSM to UMTS.

3.5 Comment on the following statement: "GSM will become obsolete as a result of UMTS."

3.6 How long will the GSM licences run in Germany?

3.7 List three requirements of IMT-2000 systems.

3.8 List examples of IMT-2000 systems other than UMTS.

3.9 Why will cdma2000 play a particularly important role in the USA?

3.10 The different versions of the standard are summarised in so-called releases. Which release will form the basis for the first commercial systems? What happens to the installed infrastructure when a change is made to a new release?

3.11 When is a commercial start for UMTS anticipated in Europe?

3.12 What are the risks you foresee that could affect the timetable for introducing UMTS? Name at least two.

3.13 Which frequency bands are being provided for UMTS in Europe?

3.14 For which extension bands does the UMTS-Forum have a preference?

4 UMTS System Architecture

4.1 Basic system architecture

Figure 4.1 presents the basic architecture of a UMTS network. A distinction is made between four logical blocks, each with totally different responsibilities.

The *UMTS Subscriber Identity Module* (USIM) is shown on the left. As in GSM, this chip card contains user-specific information and the authentication key that authenticates a user's access to a network. Since the USIM belongs to the network operator, a contractual relationship usually exists between the user of the card and the network operator. There is a growing trend to use the SIM card not only for the storage of individual information but also as an execution environment for programmes. In this case, the card has to be equipped with a microprocessor (see Chapter 10).

In UMTS the terminal is called *Mobile Equipment* (ME). Incorporated in the ME are the protocol stack of the radio interface as well as the operating elements for the user interface. Along with the keyboard and display, in the future these will also include a camera and a video codec. The USIM is inserted in a slot in the ME. Since it does not incorporate its own operating elements, it uses those of the ME.

The fixed network infrastructure that contains the facilities for transmitting over radio is called *Radio Access Network* (RAN). The components of the RAN are the base stations, which are called *Node B* in UMTS, and control nodes (*Radio Network Controller* (RNC)), which connect the RAN to the *Core Network* (CN).

The RAN encapsulates all tasks connected with the transmission of information over radio. Thus it should be possible to support other radio interfaces through the exchange of the RAN. In reality, the plan is eventually to be able to link HiperLAN/2 systems to a CN. A RAN is also often referred to as *Radio Network Subsystem* (RNS).

The *Core Network* (CN) is the long-range network that transports a user's data to its respective destination. Consequently, the CN contains a multitude of switching systems as well as gateways to other networks, such as the *Integrated Services Digital Network* (ISDN) or the Internet. It also includes databases that are used for mobility management, user management and billing. Moreover, facilities for network management (*Operations and Maintenance Centre* (OMC)), which manage the acCN as well as the RAN, are embedded in the CN [14].

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff
 ISBN: 0-470-84557-0

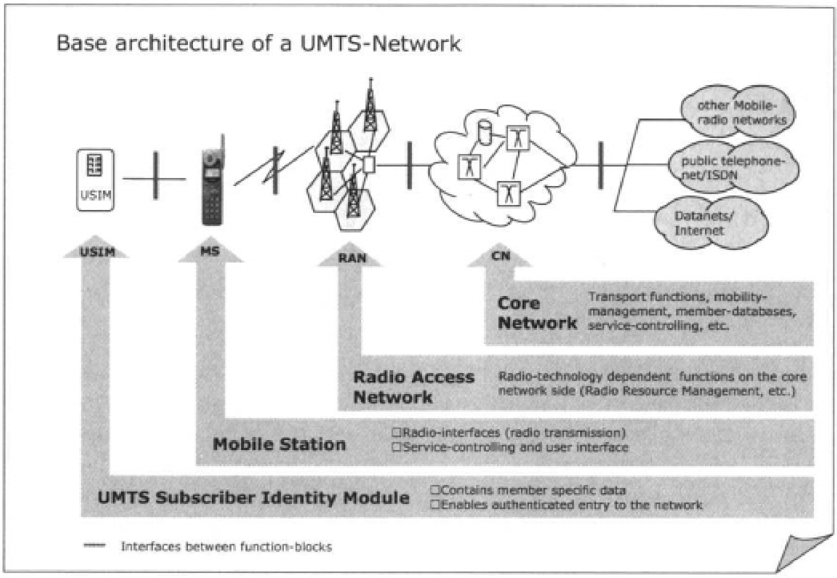

Figure 4.1: UMTS basic architecture

4.2 Functional units in UMTS

In the UMTS standard the units presented are combined into function blocks called domains (Figure 4.2). Thus the SIM card is part of the *USIM* domain. The functions of the terminal belong to the *Mobile Equipment* domain. Both these domains together form the *User Equipment Domain*. When talking about a UMTS mobile telephone, one often just simply says *User Equipment* (UE). The Mobile Equipment domain comprises all functions a user requires for access to the UMTS network.

All nodes and functions of the *Radio Access Network* (RAN) are contained in the *Access Network* domain. The CN is divided into three domains that can be identical in special cases:

- The *Serving Network* domain contains the functions of the specific CN that a user uses at a specific point in time for access to UMTS services.
- Access to certain services requires the implementation of database queries in the home network of the user. In the event that the *serving network* is not linked directly to the *home network*, the data passes through so-called transmit networks. The functions of these transit networks are contained in the *Transit Network* domain.

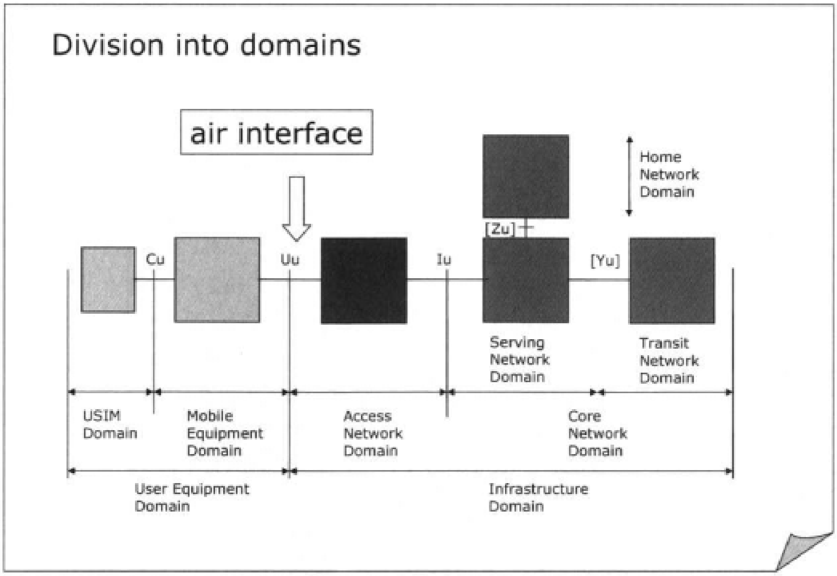

Figure 4.2: Division into domains

- All functions implemented in the home network of the user fall under the *Home Network* domain.

These three domains are combined into the *Core Network* domain. Together with the *Access Network* domain they form the *Infrastructure* domain.

Interfaces are defined between the domains: The Cu interface occurs between the USIM domain and the mobile equipment domain. The definition of this interface comprises the electrical and physical specifications as well as the protocol stack between the USIM card and the terminal. This guarantees that the USIM cards of different network operators can work together with all terminals.

Between the terminal and the RAN is the Uu interface that is described at length in Chapters 6–8. The Iu interface, described below, is located at the gateway from the access network domain to the serving network domain. The interfaces to the home network domain and the transit network domain are logical interfaces that will not be dealt with here.

If one considers the communication processes between the individual network elements, it is possible to combine this communication and the participating partners into something called a *stratum*.

The access stratum encompasses the terminal and the radio access network (RAN). The protocol stack at the Uu interface, which is described in Chap-

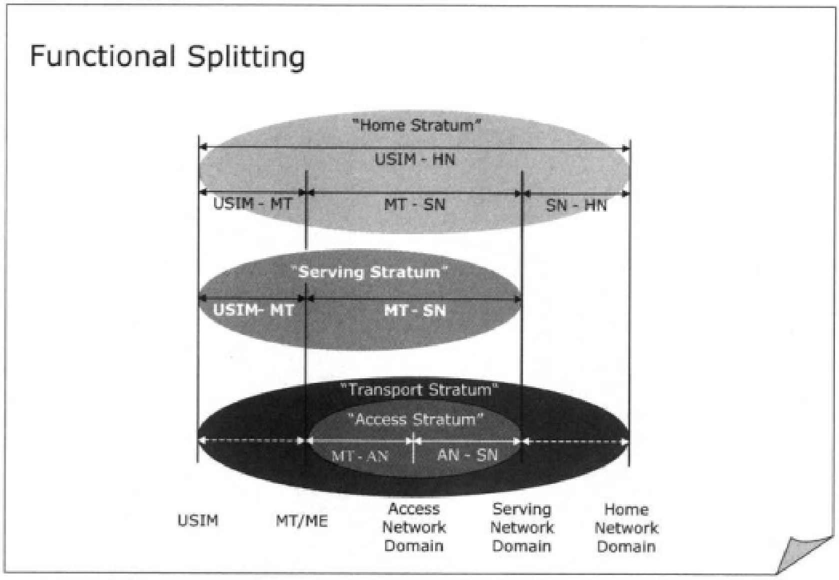

Figure 4.3: Functional division in UMTS

ter 5, works within the access stratum. Above these protocols is the signalling between the USIM or the UE and the CN. This communication is merged together in the Non-Access-Stratum (NAS). There are other ways of combining p rocesses (see Figure 4.3). For example, the exchange of keys for user authentication takes place within the home stratum.

Unlike domains that are divided up, non-functional units are separated and bundled as communication partners together with the appropriate protocols.

4.3 Types of switching

There are mainly two possibilities for exchanging information between two communicating partners. With the first method, a channel is set up at the beginning of the communication. This channel remains operational for the entire duration of the transmission. After all information has been transmitted, the channel is released again. If both partners are connected with one another not directly but over a connection network, hops are set up and occupied for the entire duration of the communication.

Since the communication path is constantly available once set-up has taken place, the individual data blocks that flow over a channel do not have to carry

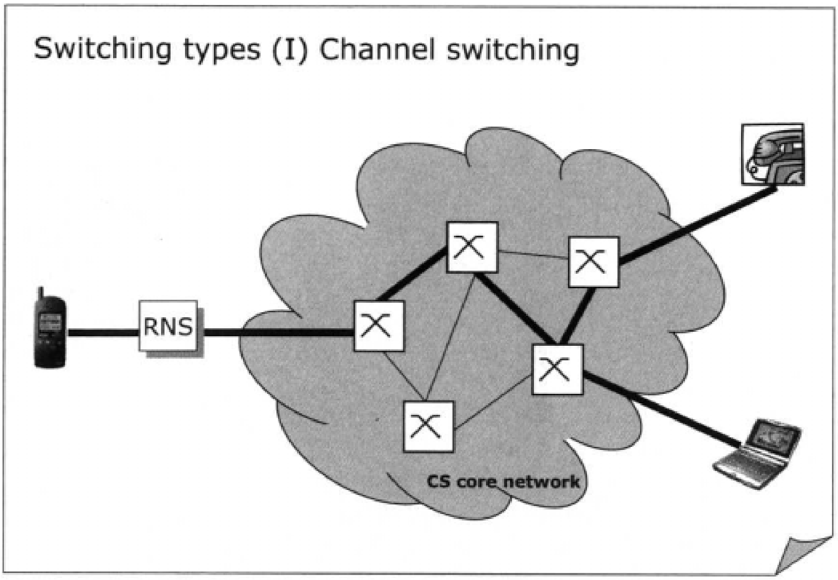

Figure 4.4: Circuit switching

any addressing information. The selection of channels is enough to provide unique identification of the receiver.

This type of communication is familiar from voice telephony. User access to the Internet is also normally implemented through a modem connection over which the access node is dialled. This way of switching data between users is called *circuit switching*, or CS for short.

Figure 4.4 shows circuit-switched transmission over a UMTS network. The UE communicates over a fixed line either with another voice user (top right) or is connected with another data terminal (bottom right).

The initial telephone exchanges were exclusively dominated by circuit switching. It is easy to form a mental picture of how the operators at the exchange had to plug in a cable to set up a connection for transmission before a call could be made.

Circuit switching is sufficient for most applications but has one disadvantage: if circuits are not used in the intervals, they remain blocked anyway for the entire duration of the connection. This reduces the capacity available to other users who could be transmitting data during the pauses.

This disadvantage can be avoided through packet switching (or PS for short). A data stream is subdivided in the transmitter into small data packets. Each

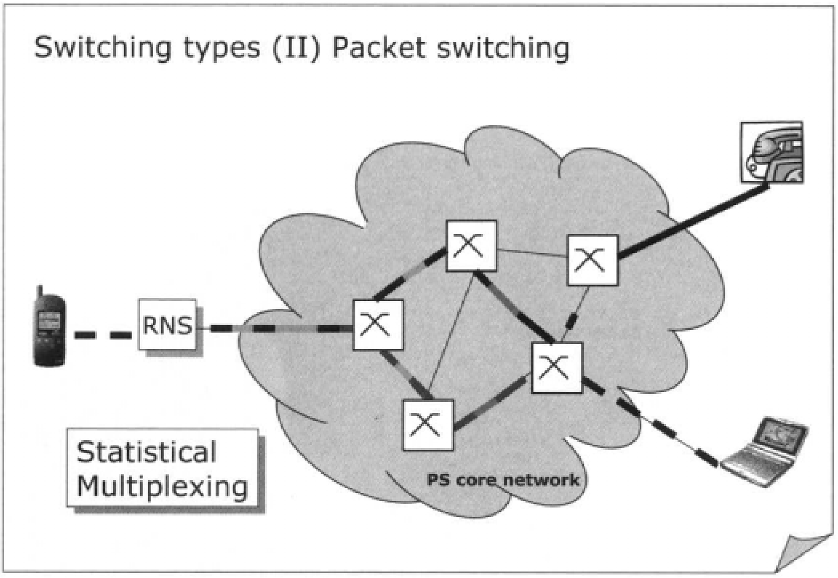

Figure 4.5: Packet switching

individual packet is provided with a destination address so that it can be transmitted individually through the intermediate stations. The major advantages of this switching technology are its robustness in the event of a failure of the individual network nodes and its favourable utilisation of existing transmission paths.

When a node breaks down in a packet-switched network, the other network nodes, depending on the availability of appropriate protocols, can identify a different route for incoming packets and still maintain end-to-end transmission.

Since individual users are no longer exclusively using the circuits, this is referred to as a *statistical multiplexing* of several connections. This type of transmission mainly functions with data transmission. The traffic volumes are typically bursty in character rather than steady. Phases in which there is an intensive use of a connection are offset by long pauses in which other users can use the gradually available resources for their own transmission needs.

The best-known example of a packet-switched network is the Internet. The Internet was developed to remain largely intact even in the event of a massive failure of many network nodes. Transmission is over IP packets that carry both the transmitter and the destination addresses.

Figure 4.5 shows transmission over a packet-switched UMTS network. The mobile station is transmitting individual data packets that are sent along with other packets over the circuits in the fixed network. Ultimately they either reach a computer that is also transmitting on a packet-switched basis (bottom right) or the data is converted into a circuit-switched connection in an intermediate node (top right). A typical application would be the transmission of voice over an IP network (*Voice over IP* (VoIP)). Since the user being called possibly only has a normal telephone, the incoming voice packets have to be converted into the *Pulse Code Modulation* (PCM) representation normally used in telephone networks.

One problem with packet switching is that jams can easily occur on the path from sender to receiver if too many data streams are being routed over a section at the same time. This effect is familiar from the Internet: the average transmission speed falls during the day when large numbers of users are surfing on the WWW. This is especially critical for applications that have a sensitive reaction to variable delays, e.g., audio and video transmission. The current Version 6 of the IP protocol (*Internet Protocol Version 6* (IPv6)) introduces quality of service control that users can utilise to request a certain quality of service. This requires priority control on the network nodes.

The data transmission techniques familiar from GSM, *High Speed Circuit Switched Data* (HSCSD) and *General Packet Radio Service* (GPRS) (see Chapter 3), are effective examples of the two types of switching. With HSCSD a channel or several channels are reserved for the entire duration of a connection, irrespective of whether data is being transmitted. In GPRS, on the other hand, the physical channel is already divided up between the various users at the radio interface. This means that a user can retain a constant connection with the network without incurring connection costs.

4.4 Architecture of the access plane

UMTS networks support both types of switching, in each case with special nodes in the CN. The nodes necessary for circuit-switched transmission are shown in the top-right corner of Figure 4.6. This part of the network is heavily based on the existing GSM networks. The nodes used for packet-switched transmission appear underneath. These nodes have already been introduced with GPRS into the GSM architecture, even if some of the protocols are different.

Both parts of the CN use the same radio access network. The interface between CN and RAN (Iu-interface) is divided into the interface for the circuit-switched part of the CN (Iu_{CS}-interface) and the interface for the packet-switched part (Iu_{PS}-interface).

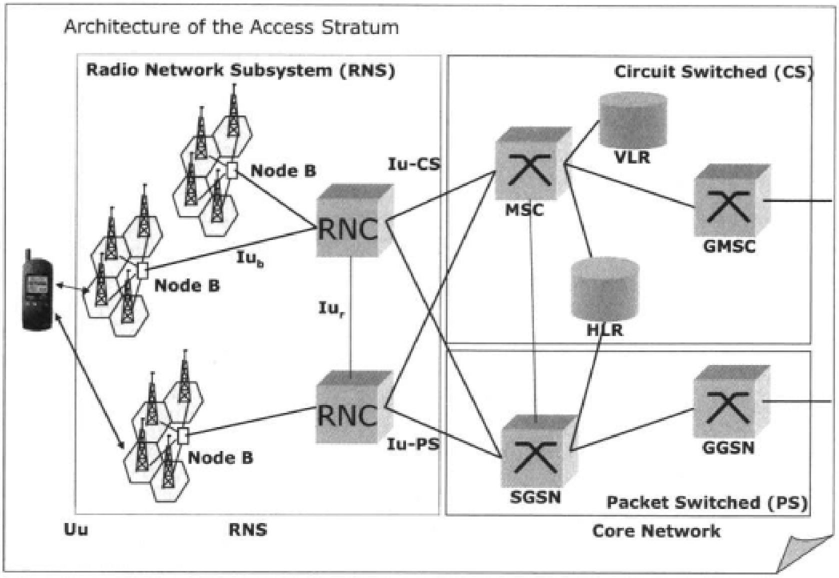

Figure 4.6: Architecture of the access plane

The RAN contains two types of nodes: the *Radio Network Controller* (RNC), which controls resource management in one or more base stations (Node B). Node B in turn supplies one or more radio cells. The interface between RNC and Node B is called the Iu_b-interface. A new feature, and not available in this form with GSM, is the direct connection of RNCs over the Iu_r-interface.

The UE is connected with Node B over the Uu-interface.

The individual elements of a UMTS network have totally different tasks, which will be described below (Figure 4.7).

4.4.1 Mobile Services Switching Centre (MSC)

The *Mobile Services Switching Centre* (MSC) is a switching node that supports circuit-switched connections. In addition to its switching tasks, an MSC must also support user mobility. If a user moves area while maintaining a connection, the MSC forwards the connection over the appropriate RNCs and Node Bs to the location area of the user (Handover). In addition, the MSC stores (in attached databases - see below) the current location area of the user so that a connection can be set up in the right cell in the event of an incoming call (location management). The MSC also participates in the mechanisms for user authentication as well as in the encryption of user data.

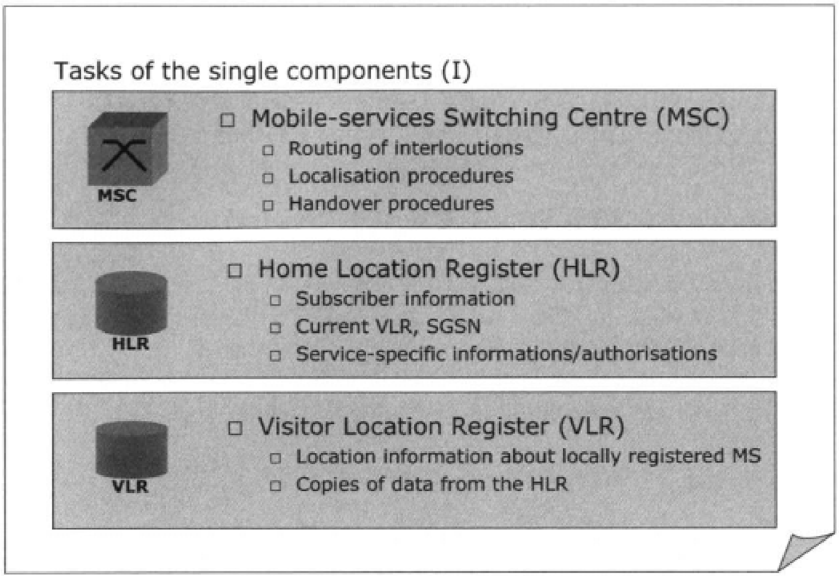

Figure 4.7: Tasks of MSC, HLR and VLR

The *Gateway Mobile Services Switching Centre* (GMSC), which also offers interfaces to various external networks, e.g., the *Integrated Services Digital Network* (ISDN), is a special variant of an MSC. The MSC is the central element of the circuit-switched part of the CN.

4.4.2 Home Location Register (HLR)

When a contractual relationship is established, the user data and associated authorisations and keys are stored in a database called the *Home Location Register* (HLR). A reference is stored in the HLR indicating in which part of the mobile radio network (see VLR) a user is currently operating so that an incoming call can be forwarded accordingly.

4.4.3 Visitor Location Register (VLR)

The *Visitor Location Register* (VLR) is a database similar to the HLR and stores a local copy of the data from the HLR. However, the data in a VLR is dynamic. As soon as a user changes *location area*, the information in the VLR is updated. The advantage of this multi-level concept is that not all information has to be interrogated in the central database, and this prevents considerable overloading of the latter.

The interaction of HLR and VLR and the way data is stored are explained in detail in Section 4.7. Figure 4.8 presents the elements of the CN that are used for packet switching. These elements are described below.

4.4.4 Serving GPRS Support Node (SGSN)

The *Serving GPRS Support Node* (SGSN) carries out tasks for packet-switched transmission similar to those of the MSC and VLR nodes in the circuit-switched part. The current position of a user is stored in the SGSN so that an incoming data packet can be routed to the user. In addition to routing functions, the SGSN also handles authentication and stores a local copy of the user information.

4.4.5 Gateway GPRS Support Node (GGSN)

The gateways to other packet data networks, such as the Internet, are connected to the *Gateway GPRS Support Node* (GGSN). Consequently, the GGSN usually incorporates a firewall. Incoming data packets are packed in a special container by the GGSN and forwarded over the *GRPS Tunnel Protocol* (GTP) protocol to the SGSN.

4.4.6 GPRS Register (GR)

The information required for the operation of a packet-switched transmission is stored in the GR, a database that is part of the HLR. It includes, for example, a user's authorisations for access to the Internet. The interaction of SGSN and GGSN is described in detail in Section 4.7.

4.4.7 Radio Network Controller (RNC)

The *Radio Network Controller* (RNC) is the central node in a radio access network (Figure 4.9). It takes the place of the *Base Station Controller* (BSC) familiar from GSM and assumes the management of the resources in all attached cells (channel allocation, handover, power control). A large number of the protocols between UE and RAN are implemented in the RNC (see Chapter 5). The RNC concurrently communicates over the Iu-interface with a maximum of one fixed network node MSC and SGSN at any given time. Thus each RNC is allocated to an MSC and an SGSN. It also has the option of using the Iu_r-interface to communicate over the CN with neighbouring RNCs.

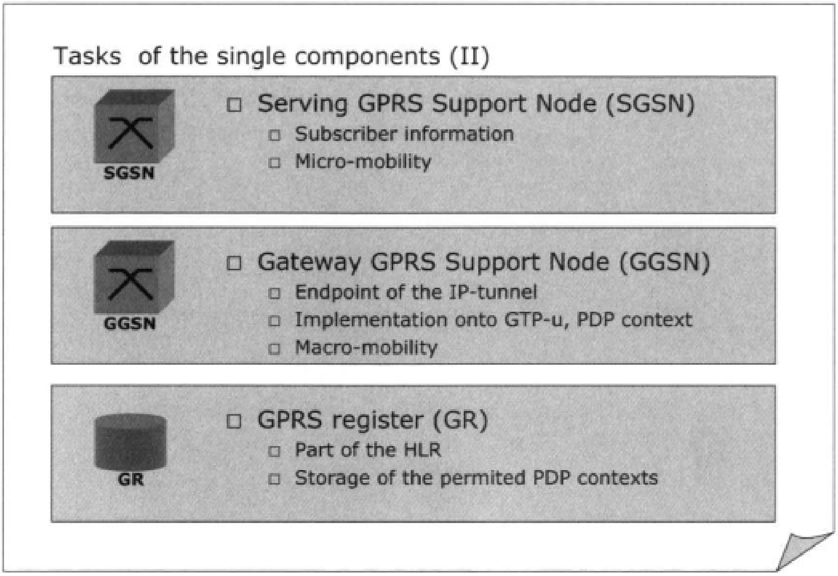

Figure 4.8: Tasks of SGSN, GGSN and GR

Figure 4.10 lists the tasks of an RNC. An RNC autonomously performs all tasks related to data transmission over the radio interface. These tasks are combined in the concept of ***Radio Resource Management*** (RRM). The RNC essentially is responsible for the following:

1. Call admission control: Unlike the situation in GSM, the transmission technology CDMA provides a large number of possible channels at the radio interface, although not all of them can be used at the same time. The reason is the problem of interference that increases as more channels are used (see Chapter 9). Consequently, the RNC must calculate the current traffic load for each individual cell. On the basis of this information, ***Call Admission Control*** (CAC) then decides whether the interference level after the channel requested is occupied is acceptable and, if necessary, rejects the call.

2. Radio resource management: The RNC manages the radio resources in all attached cells. In addition to planning channel use, this includes calculating interference and utilisation levels and priority control.

3. Radio bearer set-up and release: In UMTS the user data channel within the access stratum above the ***Radio Link Control*** (RLC) sublayer (see Chapter 5) is called the ***radio bearer***. The RNC is responsible for setting up, maintaining and ultimately releasing radio bearers as required.

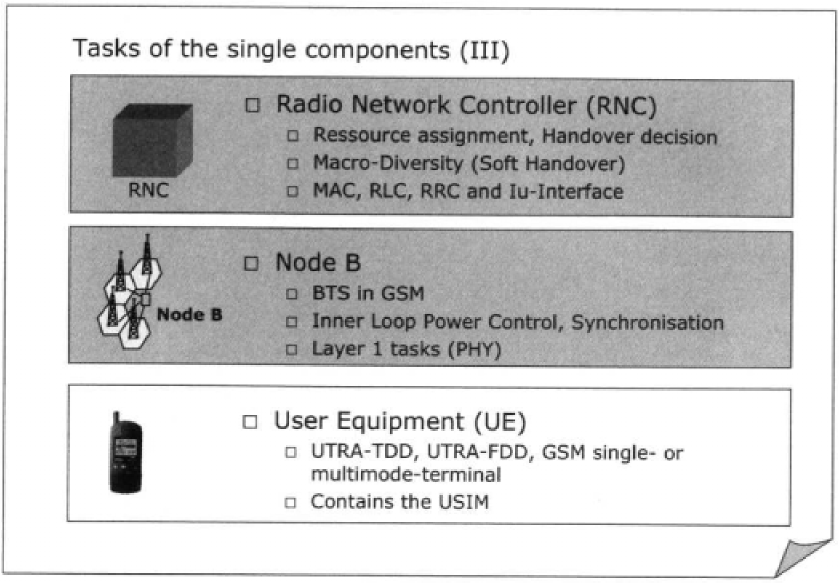

Figure 4.9: Tasks of RNC, Node B and UE

The set-up of a radio bearer is comparable to the establishment of a logical data connection and does not indicate whether packet-switched or circuit-switched data is being transmitted over the radio bearer.

4. Code allocation: CDMA codes in UMTS are managed in what is called a code tree (see Chapter 6). The RNC allocates part of this code tree to each mobile station and can also change the allocation during the course of a connection.

5. Power control: It is important for the efficient operation of a CDMA network that the transmitter power of all users is controlled. The actual fast control process takes place in Node B but the target control values are established in the RNC (see Section 7.4). Along with the measured interference values, information from other cells and in some cases even beyond RNC boundaries are included in the control.

6. Packet scheduling: With packet-switched data transmission several mobile stations share the same resources at the radio interface. The RNC has the task of cyclically allocating transmission capacity to the individual stations, at the same time taking into account the negotiated quality of service.

7. Handover: Based on the measurement values supplied by Node B and UE, the RNC detects whether a different cell is better suited for a current

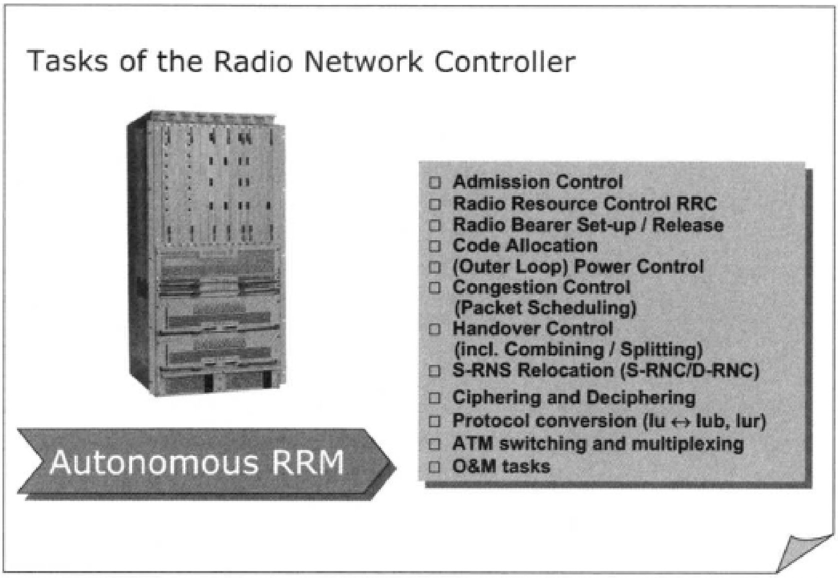

Figure 4.10: Tasks of RNC

connection. If the RNC decides on a handover, it takes responsibility for the signalling with the new cell and informs the mobile station about the new channel. The different types of handover and the concepts of combining/splitting will be examined below.

8. SRNS relocation: It is possible that a mobile station will move out of the area managed by the RNC. In this case, another RNC has to assume control for the connection (see Section 4.5.1).

9. Encryption: Data arriving from the fixed network for transmission over the radio interface is encrypted in the RNC.

10. Protocol conversion: The RNC must handle the communication between CN, neighbouring RNCs and connected Node Bs.

11. ATM switching: The communication paths between Node Bs and RNC, between RNCs and between the RNC and the CN are normally based on ATM routes. The RNC must be able to switch and connect ATM connections to enable communication between the various nodes.

12. O&M: This abbreviation encompasses the administrative functions involved in network management. Available data must be transmitted over defined interfaces to an *Operations and Maintenance Centre* (OMC).

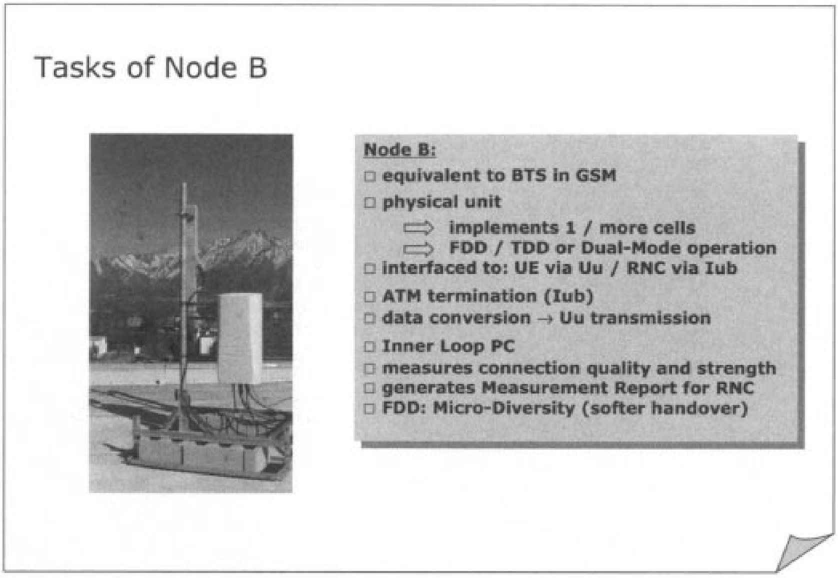

Figure 4.11: Tasks of Node B

4.4.8 Node B

The name *Node B* is an unfortunate choice: During standardisation this name was planned as a temporary solution until the introduction of a more appropriate term. However, the name stuck nevertheless during the course of the standardisation activities and therefore the base station in UMTS is called Node B. This node corresponds to the ***Base Transceiver Station*** (BTS) familiar from GSM (see Figure 4.11). The tasks directly connected to the radio interface are handled in the BTS. The inputs comes from the RNC. A Node B can manage one or several cells and is connected with the RNC over the $\mathrm{Iu_b}$-interface.

Node B is the counterpart of BTS in GSM. It supplies one or several cells. Along with the antenna system, Node B includes a CDMA receiver that converts the signals of the radio interface into a data stream and then forwards it to the RNC over the $\mathrm{Iu_b}$-interface. In the opposite direction the CDMA transmitter prepares incoming data for transport over the radio interface and routes it to the power amplifier.

There are three types of Node B corresponding to the two UTRA modes: UTRA-FDD Node B, UTRA-TDD Node B and Dual-mode Node B, which can use both UTRA modes simultaneously.

Currently, the Node B is linked over an ATM link to the RNC. Due to the possible large distance between Node B and RNC and the length of the processing times, certain particularly time-critical tasks cannot be stored in the RNC: this includes *Inner Loop Power Control* that in a CDMA network ensures that all users receive at the same signal strength.

The RNC has to have as exact a picture as possible of the current situation in a cell so that it can make sensible decisions on handover, power control and call admission control. Consequently, mobile stations and Node B periodically carry out measurements of the connection quality and interference levels and transmit the results to the RNC.

In the special case of softer handover, the splitting and combining of data streams of the various sectors are also already handled in Node B.

It is estimated that approximately 100,000 antenna sites are needed for a national UMTS network in Germany. This number will be reduced if antenna systems are shared by different operators. The project stages extend from macro-Node B, which supplies large cells through the use of high antenna masts, to a smaller micro-Node B, which broadcasts individual traffic movements below roof edges, and a pico-Node B, which is responsible for the interior supply to buildings. The smallest pico-Node Bs should achieve a volume of below two litres. Figure 4.11 shows a Node B that is experimentally installed on a roof.

4.4.9 User Equipment (UE)

The last important network node is the user terminal (UE). This equipment can support one or more radio standards and contains the USIM. It is simultaneously the counterpart to Node B, RNC and the CN (Figure 4.12).

Like Node B, the UE is responsible for processing the radio signal. This compute-intensive task comprises error correction, spreading and signal modulation as well as radio processing up to the power amplifier. On command of the RAN the mobile station must adapt to the transmitter power (power control, see Section 7.4).

As a counterpart to the RNC, the mobile station participates in the signalling for connection set-up and release as well as in the execution of handovers. For this purpose it measures the received field strength of neighbouring cells and transmits the measured values to the RNC. The encryption and decryption of communication also take place with the RNC in the UE.

The UE delivers information about the current location area to the CN (mobility management). Negotiation over the quality service required for a particular service and reciprocal authentication also takes place between UE and CN.

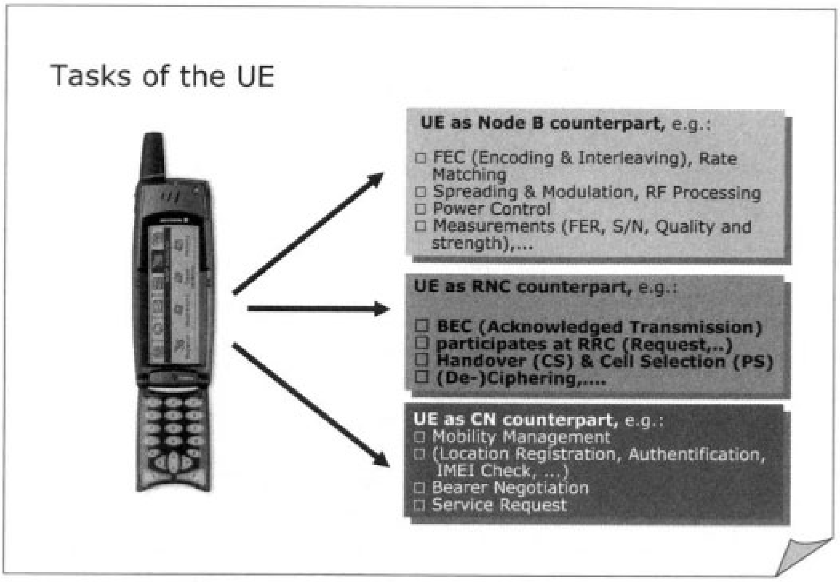

Figure 4.12: Tasks of UE

This long list of tasks handled in a UE is accompanied by the users' request for larger displays that also support the decoding of video data. It is also planned that a camera with an accompanying MPEG codec will be integrated into the equipment. Since a terminal should also enable the playing of games, an efficient processor with substantial memory is to be used. The most important thing, however, is that the equipment should remain small and easy to handle.

What is obvious is that these requirements are in partial conflict with one another. Above all, the power requirements of the efficient components necessary could result in short operating times of the equipment. This is something customers who are used to the operating times of GSM that last several days will only accept to a point. Consequently, the initial UMTS terminals are expected, for example, to be integrated into a notebook computer because, first, a larger battery will be available and, second, customers are used to and accept lower operating times of several hours when working with computers.

The complexity of UE is very high and it will take a masterstroke on the part of the developers involved to accommodate all the requirements and still develop attractive terminals.

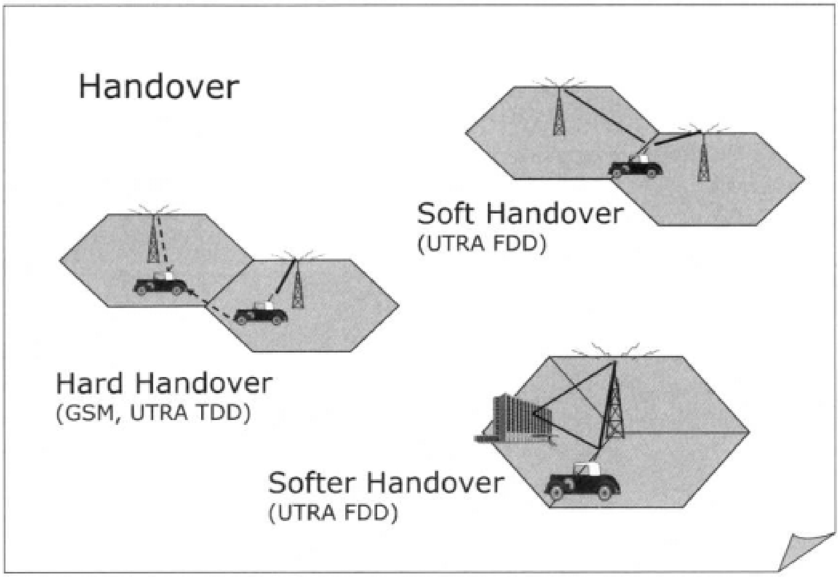

Figure 4.13: Different types of handover

4.5 Handover in UMTS

It was already explained in Chapter 2 that the two methods *handover* and *location management* have to be taken into account for user mobility in mobile radio networks. The term handover always relates to services that are operated on a circuit-switched basis. Packet-switched services use a different technique, which will be explained later.

In UMTS there are mainly three different types of handover. These are shown in Figure 4.13 and will be briefly explained here:

1. With *hard handover* already familiar from GSM, a connection is switched hard at a particular time. This method is also used in UTRA-TDD-mode, because sufficient time is available between the individual transmitter and receive phases in a mobile station to switch to a new cell. The changeover to the new cell thus occurs from one frame to the next one.

2. A *soft handover* is when a mobile station communicates simultaneously with up to three sectors from different Node-Bs. The data is split up in the RNC (splitting), broadcast over the Node Bs and combined again in the mobile station. Data from all participating Node Bs is received on the uplink and forwarded to the RNC. The RNC combines the two

data streams again and transfers the data to the CN. The soft handover gets its name from the fact that there is no fixed switchover point and instead a *soft* connection is transferred from one base station to the next one. The new base station initially only contributes very little to the transmission; however, the further the UE moves into the new cell, the more responsibility the new base station assumes. Finally, the connection to the old station is terminated and the mobile station leaves the soft handover state.

This technique is also called macro-diversity and offers several advantages:

a) The connection becomes more resistant to shadowing due to the reduced probability that, considering all supplying base stations, the mobile station will end up in shadowing. If an interference object cuts off a connection to a base station in a soft handover, there is the possibility the connection will function over the second station and the communication will not be cut off.

b) When the minimum received power is calculated, a small reserve against fast fading through multipath propagation can be incorporated. Since the drop in received power through multipath propagation is almost static in the case of static transmitters and can amount to up to 30 dB, it is possible that a static mobile station will not be supplied adequately. Soft handover offers the option of transmitting data over the second Node B and thus maintaining the communication.

c) Furthermore, a soft handover offers the possibility of reducing the near-far effect. This effect is described in Chapter 9.

3. *Softer handover* is a special version of soft handover in that transmission can also run in parallel over different sectors of the same Node B. The advantages mentioned in conjunction with soft handover also apply to softer handover, although the Node B can already be entrusted with the task of combining the two data streams and only transferring one data stream to the RNC.

The two methods of soft handover and softer handover along with their advantages for the operation of cellular CDMA networks are described in detail again in Chapter 9.

4.5.1 The role of RNC in a handover

A soft handover is a relatively simple and easy to understand matter if the participating Node Bs belong to the same RNC. However, there is a problem if the Node Bs are controlled by different RNCs: The CN is not allowed to

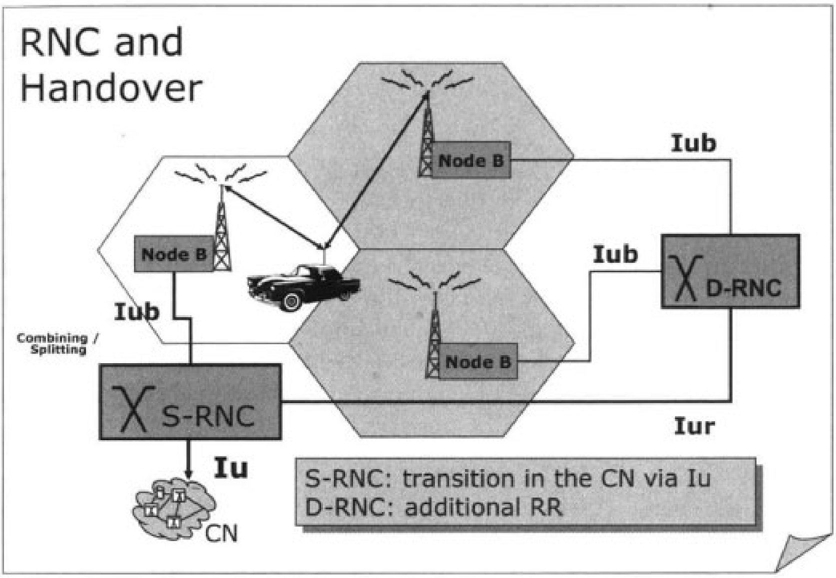

Figure 4.14: The role of RNC in a handover

be aware of problems that occur in the RAN or take over any tasks that are directly connected to the radio interface. However, this would be necessary if the two RNCs were not able to communicate directly with one another over the Iu_r-interface.

Let us play through the process using an example: The mobile station shown in Figure 4.14 is supplied by the left cell 1. The RNC shown on the left controls the connection and maintains it with the fixed network over the Iu-interface. Consequently, this RNC is also called a *Controlling RNC* (CRNC).

If the mobile station moves towards the right to the edge of cell 1, a soft handover occurs. The mobile station is supplied by two Node Bs (1 and 2). In this case, Node B 2 is controlled by a different RNC in which the CRNC is reserving radio resources (*Radio Resource* (RR)) for the mobile station. However, control over the connection remains with the RNC on the left. It remains the *Serving RNC* (SRNC), whereas the RNC on the right is controlled remotely over the Iu_r-interface with regard to this connection. It becomes the *Drift RNC* (DRNC). It is the task of the SRNC to handle the *combining* of the data that is transmitted from the mobile station on the uplink. The DRNC forwards the data unprocessed to the SRNC. On the downlink the SRNC sends a copy of the data arriving from the CN to the DRNC (splitting), which then forwards it over the attached Node Bs to the mobile station.

The role of Node B 1 gradually decreases as the mobile station moves more deeply into the supply area of Node B 2. Ultimately the connection over the old Node B can be terminated. Resources are now occupied in two RNCs although the connection could exclusively be handled by the RNC on the right.

There is a technique in this case that is called *SRNS relocation* in the standard. The Iu-reference point where data is transferred from RAN to CN is shifted from the left RNC to the right RNC. This is the only type of handover in UMTS that incorporates the CN into the handover process. The DRNC thus becomes the CRNC and now manages the connection on its own. (There is another scenario: if the Iu_r-interface is not implemented in a UMTS network, connections have to be hard-switched from one RNC to the other. The CN is also involved in this case. The connection to the network is briefly interrupted and then immediately set up with the new RAN.)

The terms SRNC and DRNC relate to an individual connection. Thus, in regard to another connection, the right RNC can assume the task of the SRNC while the left RNC functions as DRNC.

4.5.2 Handover types in UMTS

A distinction is made between various handover types depending on which network elements participate in the handover (see Figure 4.15):

1. Within a Node B a connection can be switched hard or soft from one sector to the next (Intra-Node B, Intra-RNC Handover).
2. Between different Node Bs of the same RNC a connection can be hard-switched (Inter-Node B, Intra-RNC-Handover).
3. A connection can be soft switched between the different Node Bs of the same RNC (Inter-Node B, Intra-RNC-Soft-Handover). In this case, the RNC is responsible for the combining/splitting.
4. A connection can also be forwarded between RNCs within the RAN over the Iu_r-interface. This is the case described above in which SRNS and DRNS are involved (internal Inter-RNC handover). The term *internal* handover relates to the fact that the handover is carried out totally within the RAN.
5. Switching with relocation of the Iu-reference point is called *SRNS relocation*. This is an external Inter-RNC-Handover. All the handover types mentioned so far occur within the domains managed by switching nodes (MSC)(see Figure 4.16).

Handovers that exit the area of a 3G-MSC are also conceivable:

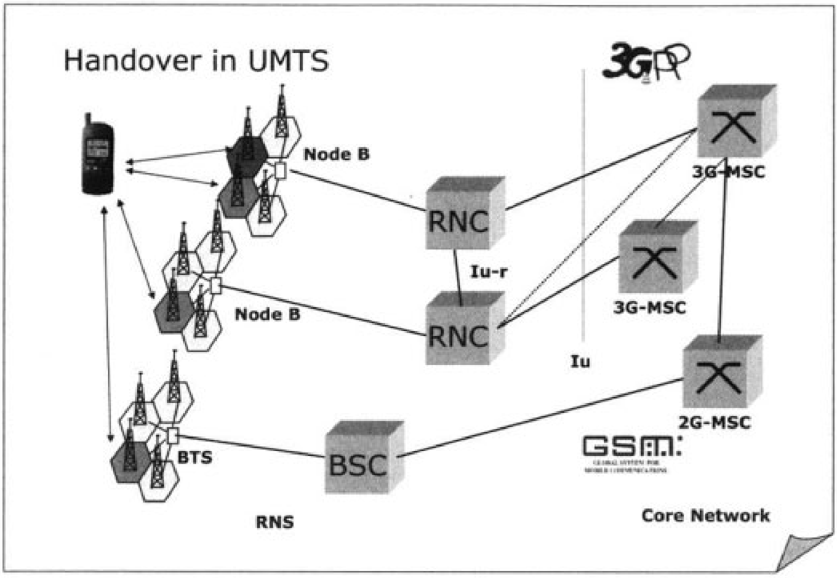

Figure 4.15: Network elements involved in handover

1. The new RAN can occur in the area of a new MSC. In this case, the connection is hard-switched over the CN to the new *Serving Radio Network Subsystem* (SRNS). The CN does not support a soft handover in this case.

2. When a connection leaves a region supplied by UMTS, it is important that it can be operated over GSM networks if they exist. This capability is mandatory for achieving satisfactory customer acceptance, because especially during the development phase UMTS will mainly only be operated in small, regional islands. The reverse route of a handover from GSM to UMTS is less critical since wide-area GSM coverage already exists (Inter-System-Handover).

3. It is conceivable that different systems of the IMT-2000 family could border each other geographically. Consequently, handover between different 3G systems will be defined later so users can change seamlessly between networks (Inter-System-Handover).

4. Figure 3.16 in Chapter 3 shows that 30 MHz is reserved twice for satellite-supported components of UMTS. Although it is still not clear which technique will be used in these bands, there are plans for handover from terrestrial UMTS to satellite-supported UMTS(S-UMTS) (Inter-Segment-Handover).

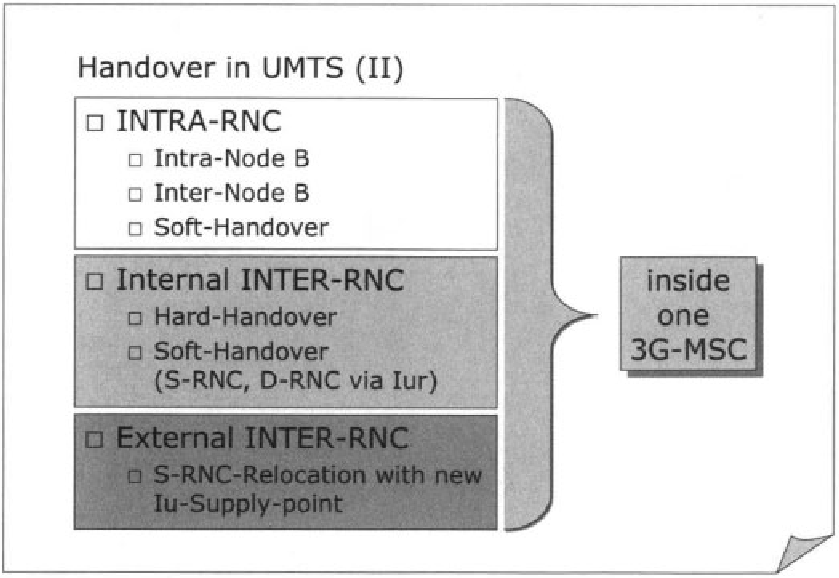

Figure 4.16: Handover types in UMTS

Thus there are various handover types with totally different requirements for signalling at the radio interface in the RAN and the CNs (see Figures 4.16 and 4.17).

4.6 Location management

To transfer incoming calls and data packets to a user, the network must know the exact location area of the user (see Figure 4.18). The location area is stored in various network nodes and updated as needed by the mobile station. Also see Chapter 2.

Parts of the network are combined into ***location areas*** so mobile stations are made aware of the fact that updates are necessary. In these sub-areas all base stations beam a specific number, called *Location Area Index* (LAI). Whenever this parameter changes, the UE is aware that an update of the database is necessary.

The network can also instruct the UE to report its current location area periodically (called *Periodic Location Update*).

Whenever the network wants to set up a connection to a mobile station, it pages the mobile station in all cells belonging to the stored location area. As

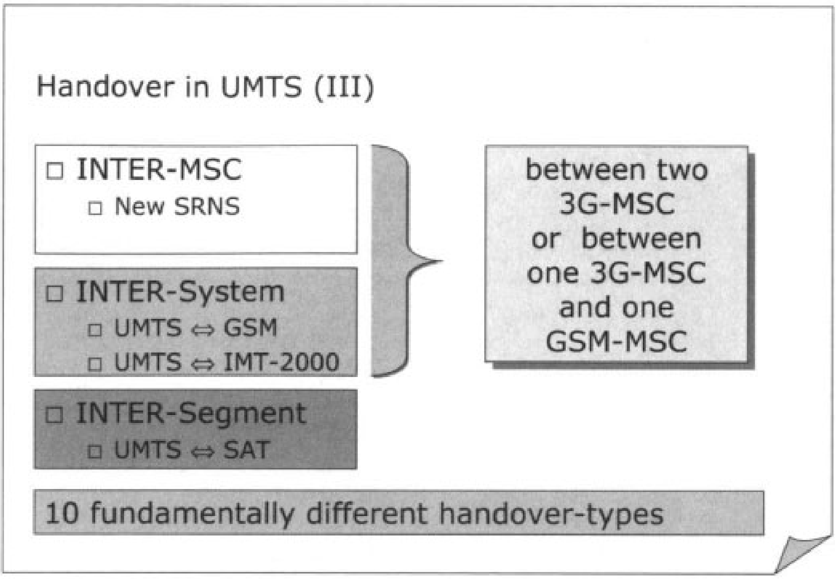

Figure 4.17: Handover types in UMTS

soon as the mobile station makes contact in a cell, the communication only still needs to be carried out over this Node B. This shows that consideration has to be given to what is the optimal size of a location area:

- A very large location area ensures that users only seldom move beyond the boundary of the location area. This reduces the signalling needed for location updates.
- On the other hand, the number of cells in which users are paged during connection set-up is very high. Thus radio resources are needed in many cells although they are not used productively.

These factors resulted in the need to define separate areas for circuit-switched and packet-switched services in location areas (Figure 4.18).

The more often a new connection is established with a mobile station, the more difficult it is to balance the factor that the UE is unnecessarily being paged in some cells. In contrast to circuit-switched services, with packet-switched services users frequently receive short data packets. Consequently, an effort is needed to define small area sizes for packet-switched services, whereas large areas are sufficient for the comparatively seldom activated circuit-switched services (e.g., voice) (Figure 4.19).

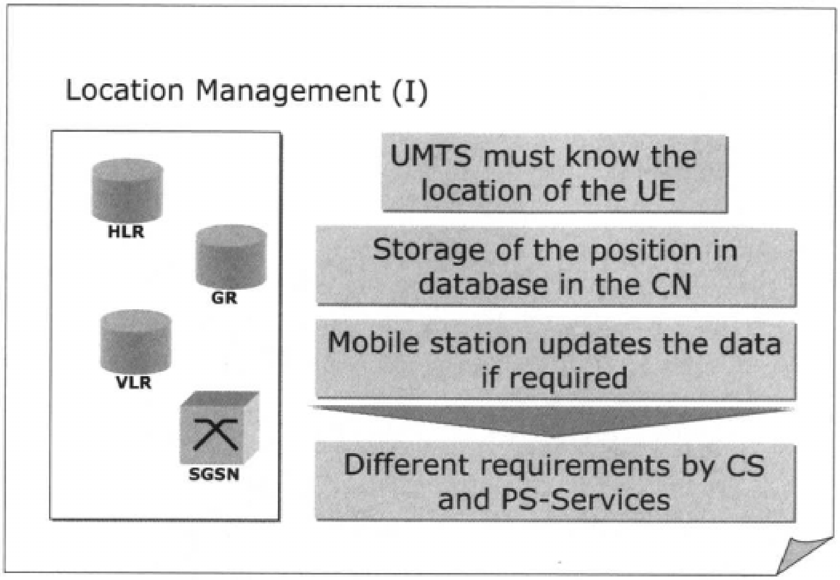

Figure 4.18: Location management

The *Routing Area* (RA) concept has been introduced in connection with packet-switched data transmission in GPRS. The principle is the same as with location areas, the only difference being that routing areas are used for PS services. As soon as a mobile station leaves the area of a Routing Area, it executes a *Routing Area Update* with the SGSN. Thus the SGSN is able to locate a mobile station more accurately in the event of incoming packets. During an active data transmission the SGSN knows the position of the mobile station down to the level of the individual cell. After a certain period of inactivity a timer runs out, and the SGSN again has to page the mobile station in the entire routing area and commence with new data transmission.

Figure 4.20 clarifies the division of the network into routing areas and location areas. The smallest unit considered here is the *UTRAN Registration Area* (URA). A URA consists of one or more UMTS cells that are combined in order to avoid a surplus of cell-changing procedures on the edge of cells.

The routing area and location area boundaries do not overlap but a routing area is always completely surrounded by a location area. A routing area consists of one or more *UTRAN Registration Area* (URA) that in turn consists of one or more cells.

As a result of this concept, there is a guarantee of optimal accuracy in the localisation in the network for each type of service. Location area and routing

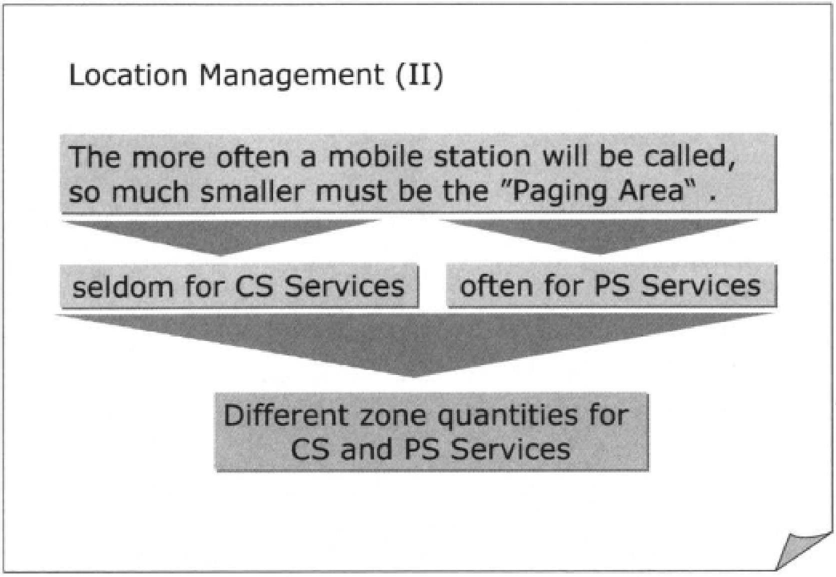

Figure 4.19: Location management

area are partly used congruently in existing GPRS networks, i.e., the areas are identical. This solely depends on the configuration by the network operator.

4.7 Circuit-switched and packet-switched connections

A call set-up example is used to illustrate the interaction between the various network elements. The procedure for call set-up is closely based on GSM.

A telephone user in the fixed network picks up the receiver and dials a UMTS telephone number. For example, the following dialling codes are provided for UMTS in Germany:

- +49 151 DeTeMobil Deutsche Telekom MobilNet GmbH
- +49 152 Vodafone Germany
- +49 155 Auditorium Investments Germany (E-Plus)
- +49 159 O_2 Germany

The call is routed through the public telephone network to a switching node of the UMTS network, the GMSC. Based on the telephone number, the switch-

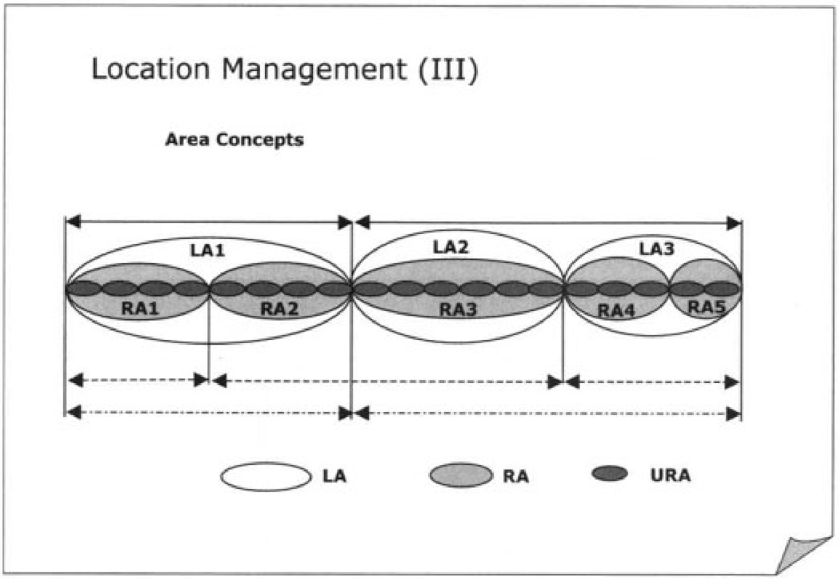

Figure 4.20: Localisation areas

ing node determines the database (HLR) in which the user data is stored. The HLR knows which area of the network the customer is located in and sends a query for a *roaming number* to the VLR responsible for this area. The VLR responds with the appropriate number and the HLR supplies this number, indicating the destination switching node within the UMTS network responsible, to the GMSC.

From there the call is routed to the destination switching node (called MSC in Figure 4.21). Through the attached VLR the MSC knows the RNC responsible for the current location area and requests that this RNC sets up a channel to the mobile station. The RNC pages the mobile station in its last known location area and sets up a connection to the mobile station over the Node B used by the mobile station when it responded to the page.

As soon as the transmission link is established at the Uu-interface, end-to-end signalling takes place and the telephone begins to ring. The connection is switched through as soon as the user picks up the phone.

The use of packet data services is a more complicated process. Figure 4.22 shows the section of the UMTS network that is responsible for packet-switched data transmission. The procedure is somewhat different:

The user of the mobile station on the left-hand side of the illustration wants to exchange data with a computer on the Internet (far right). Before the mobile

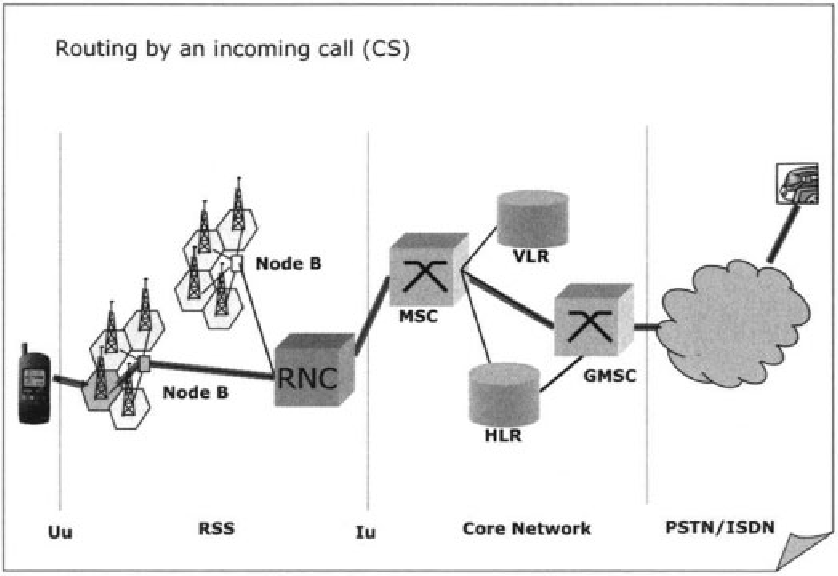

Figure 4.21: Routing an incoming call

station can access the Internet, it must activate what is called a *Packet Data Protocol* (PDP) context in the GGSN. A PDP context is a range of settings that defines which packet data networks a user may use for exchanging data. The list of permitted PDP contexts is stored in the HLR.

For example, a possible context could be access to the Internet. Another context could be access to an in-house Intranet that is not open to all customers but only to a small group of associates. Each user can use several contexts and activate them simultaneously.

When it wants to activate a PDP context, a mobile station establishes a connection over the RNC to the SGSN and sends a message that the user would like access to the Internet. The SGSN forwards the query to the responsible GGSN. A query to the HLR checks whether the user is authorised for access to the external data network. If the reply is positive, the GGSN activates the context and informs the mobile station accordingly. Through this process the mobile station is allocated a temporary IP address that allows it to be reached from outside the UMTS network. The activation of the context creates an IP tunnel. Incoming data packets from the Internet are sent to the tunnel by the GGSN and over the SGSN to the RNC. The RNC unpacks the packets and forwards them over a second tunnel to the mobile station (see Figure 4.29). This procedure separates the traffic within the UMTS network from the user traffic.

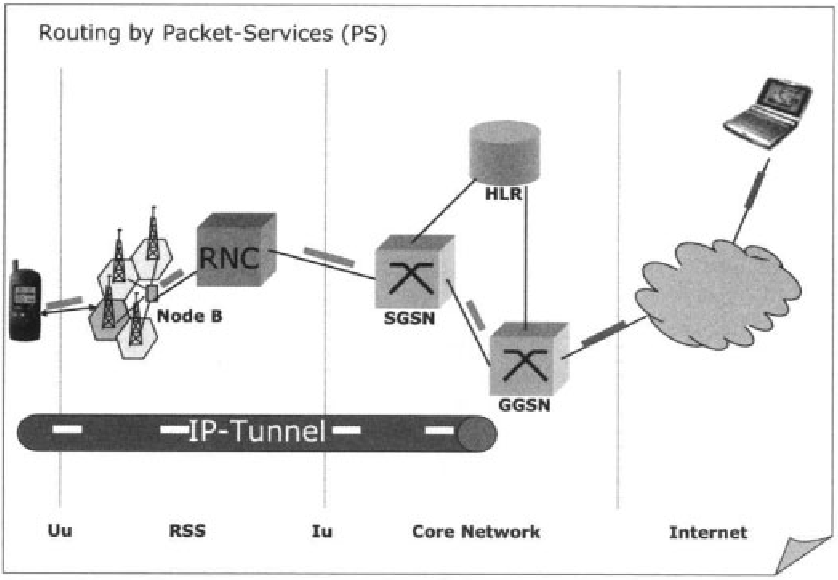

Figure 4.22: Routing a packet-switched connection

The tunnel remains active until the mobile station deactivates the context. During this time the SGSN is constantly informed about the current location area of the mobile station (micro-mobility management). If the user changes the location area for which the original SGSN was responsible, the route in the GGSN is adapted to the new SGSN. In the case of a longer period of inactivity, a timer runs out in the mobile station, the mobile station no longer makes contact with each cell change and the local information in the SGSN with regard to the location area is only at the routing area level. However, the logical connection between GGSN and mobile station continues to be maintained. The tunnel is not released until the context is deactivated or the mobile station disconnects from the SGSN. Afterwards the IP address can be used for another connection.

The transmitted data is calculated and recorded in the CN for later billing. The RN informs the CN if data packets could not be transmitted as far as the mobile station due to problems in the RAN. The quality of service required by the application from the network is negotiated in the request for the PDP context. In addition to the priority, this includes the tolerated delay, the maximum throughput required and the configuration of the transmission security within the CN and at the radio interface. Thus users who need access to their corporate networks can receive a guaranteed minimum quality of service for little additional cost. On the other hand, users who only want to surf the Internet have to tolerate a lower quality of service at a lower cost.

4.8 Protocols in the fixed network

Signalling system No. 7, called SS7 for short, has established itself as the standard signalling system between the normal switching nodes in the telephone network. Its main distinction is its separation between signalling network and user network. A similar situation is found with the subscriber connection signalling DSSI in the ISDN. In that case there are usually two B-channels at 64 kbit/s (user channels) and an additional separate D-channel at 16 kbit/s reserved for signalling.

SS7 incorporates a signalling plane with nodes called *Signalling Point* (SP). Signalling pints communicate directly with one another or over a so-called *Signalling Transfer Point* (STP) that handles the relay function (Figure 4.23). The signalling point negotiate connection set-up and release among themselves and, if signalling is successful in the user network, implement the switching in the coupling networks so that end-to-end communication is possible.

Its strict separation of the two network levels and automatic rerouting of signalling messages in the event of the failure of individual nodes make SS7 a robust and reliable switching system.

The signalling plane of SS7 for its part also provides interfaces on which other applications, such as mobility management and database queries, can be based.

4.9 Protocols at the Iu-interface

At the Iu-interface between RAN and CN there is a separation between the transport network layer and radio network layer (Figure 4.24). The transport network is a normal ATM network and is configured and controlled by the SS7.

A protocol called *Radio Access Network Application Part* (RANAP) for UMTS-specific signalling is used as the application above SS7. It contains all the functions necessary for the operation of a UMTS network at the Iu-interface. The signalling messages from RANAP are exchanged over the signalling channels of SS7. The RANAP has to be implemented in RNC, MSC and SGSN.

The user data is transported as a separate data stream directly over ATM. The ATM adaptation layer AAL5 is used for the exchange of signalling data, whereas AAL2 is used for the user data. AAL2 is mainly suitable for the

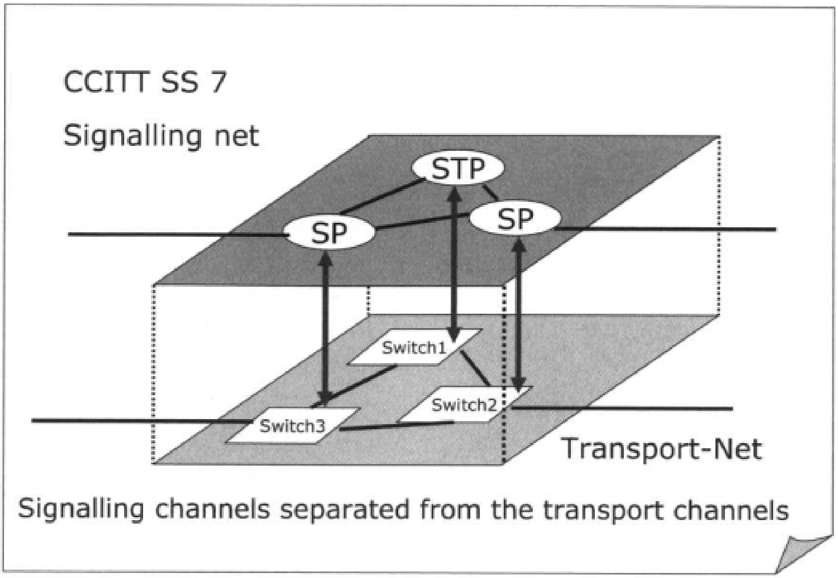

Figure 4.23: SS7 Signalling network

transmission of data that reacts very sensitively to changing delay, such as voice and video data.

4.9.1 Radio Access Network Application Part (RANAP)

The functions of RANAP are listed in Figure 4.25. This is an application used by the SS7 network to transport its messages. Approaches now also exist involving the use of IP-based protocols as the basis for the transport of RANAP messages. The protocol establishes a separate logical connection between RAN and CN for each individual mobile station being controlled.

RANAP supports the functions necessary for connecting the RAN to the CN:

1. RANAP provides for the relocation of *Serving RNC* (SRNC) due to mobile station mobility.
2. RANAP sets up and releases connections to mobile stations. This also includes releasing the corresponding RANAP connection.
3. The RAN can use the RANAP to inform the CN of data packets that were not transmitted so that they either will not be billed or will be retransmitted over a different route to the mobile station.

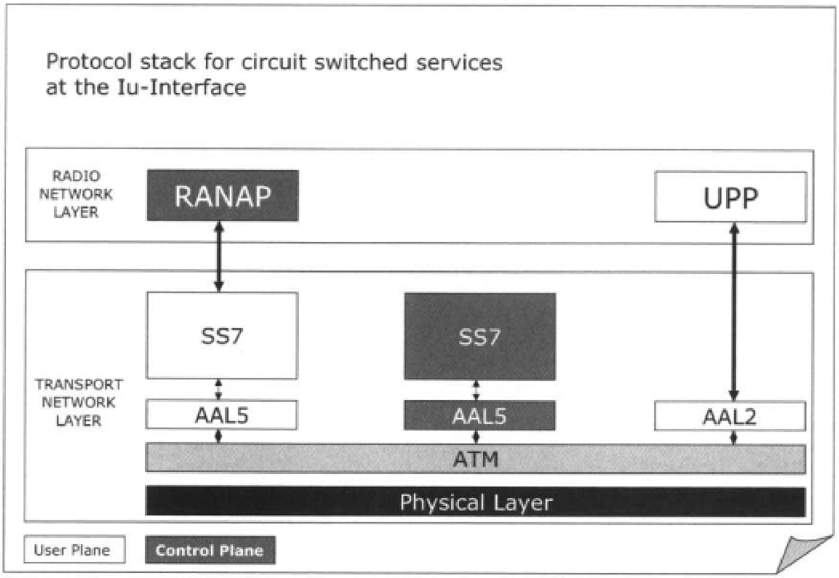

Figure 4.24: Protocol stack for circuit-switched services at the Iu-interface

4. RANAP offers functions for paging mobile stations in a location area in order to instigate a connection set-up.
5. RANAP also offers functions for direct signalling between UE and CN, for example, for mobility management (location update, routing area update, etc.).
6. Ciphering on the route to mobile stations takes place in the RNC. MSC and SGSN can use RANAP functions to control encryption.
7. The protocol also allows management functions occurring in conjunction with the operation of RANAP to be handled over the RANAP. This includes overload protection and defined restart of the protocol in the case of error. However, all connections affected by restart are cut off. RANAP also supports the activation of traces in which all activities of a particular mobile station within the RAN are recorded for diagnostic purposes.

4.9.2 Radio Network Subsystem Application Part (RNSAP)

The protocol stack used at the Iu_r-interface between various RNCs has a similar structure, except that the *Radio Network Subsystem Application Part* (RNSAP) application is used instead of a RANAP.

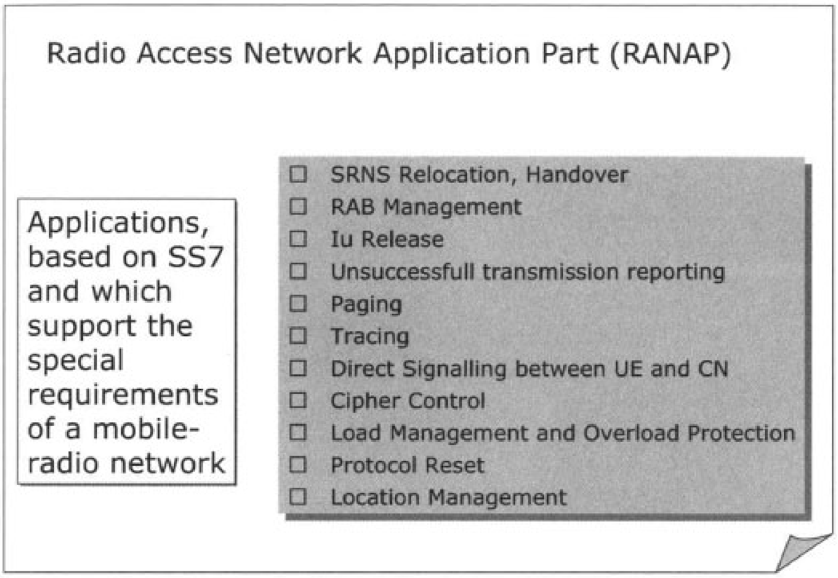

Figure 4.25: Radio Access Network Application Part

The range of RNSAP functions are divided into four groups, only parts of which can be implemented by vendors (Figure 4.26):

1. Basic functions: The first group of functions provides the basic functions absolutely necessary for operation of the Iu_r-interface. The functions for *SRNC relocation* are based on these functions but due to the complexity of their tasks they are difficult to implement. The basic functions are only used for signalling. No user data flows over the Iu_r-interface.

2. Support for dedicated channels: A network operator can also use the functions of the second group to transmit circuit-switched data for soft handover over the Iu_r-interface. This is essential for soft handover between two RNCs. Dynamic AAL2 connections can be established over the Iu_r-interface for this purpose.

3. Support for shared-use channels: The third group of functions also enables support for packet-switched services over the Iu_r-interface. This prevents the need for a change of the Iu-reference point during a current packet-switched data transmission. The mobile station moves into the area of a new RNC and continues to be supplied with data over the Iu_r-interface. Since the MAC layer in the protocol stack (see Chapter 5) has to be split between SRNC and DRNC for the support of this function, this function was not uncontroversial in the standardisation. It

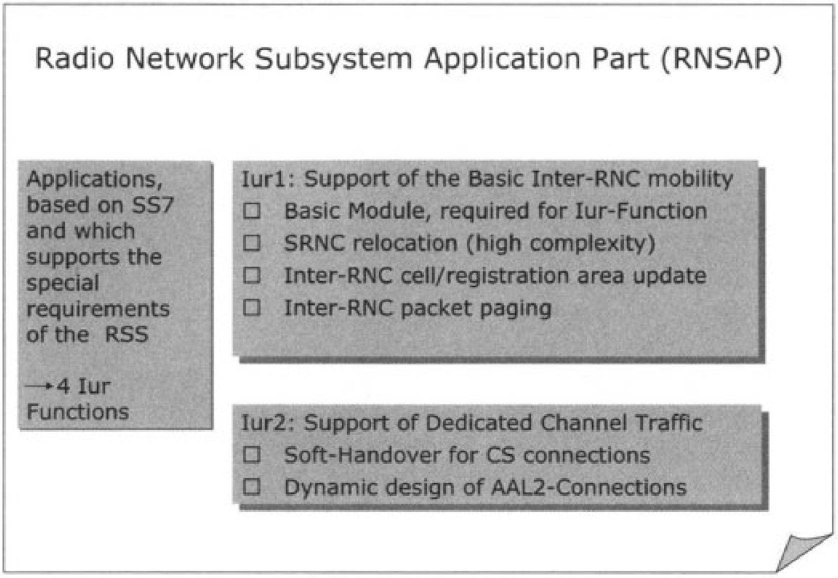

Figure 4.26: Radio Network Subsystem Application Part

is contained in the current version of the standard but is not critically necessary for the support of packet-switched services, since *SRNC relocation* can be implemented in the pauses between the individual data packets without a noticeable interruption to the user.

4. Global resource management: For the efficient planning of radio resource allocation it can be helpful if information about the current situation in neighbouring RNCs is available. Consequently, RNCs can exchange measurement values, for example on the interference levels measured in certain cells, over the Iu_r-interface. Information can also be exchanged with regard to the timings of Node Bs for the implementation of smooth handover.

   Figure 4.27 shows function groups 3 and 4 and the Iur-interface.

4.9.3 Protocol stack for circuit-switched services

User data is transmitted over the ATM ASL2 connection simultaneously with the transmission of signalling data over the RANAP. Figure 4.28 shows what happens to the data in the *UMTS Terrestrial Radio Access Network* (UTRAN): The data arriving over the ATM AAL2 connection in the RNC is packed out there and prepared for transmission over the radio interface.

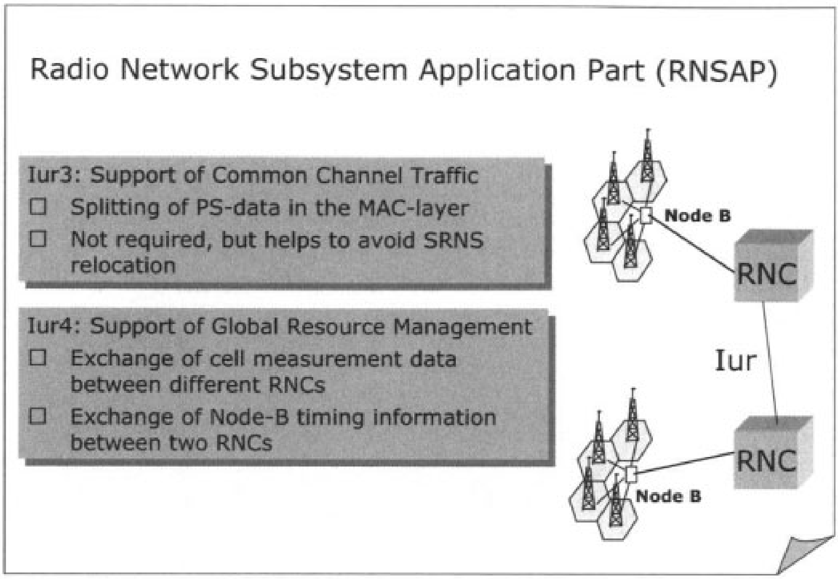

Figure 4.27: Radio Network Subsystem Application Part

The protocol stack at the radio interface is partially implemented in the RNC (MAC sublayer and RLC sublayer) and partially in Node-B (PHY layer). Time-critical tasks associated with CDMA transmission technology can be directly implemented in Node B PHY layer.

The sublayer AAL2SAR splits up data packets arriving from the RLC layer into small blocks for transmission in ATM cells and reassembles the ATM cells arriving from the opposite direction into the original data stream for transmission over the RLC layer. The tasks of the layers in the protocol stack at the radio interface are explained in Chapter 5. The recoding in the RNC is transparent for the application, for example, a circuit-switched data transmission or a voice telephone call.

4.9.4 Protocol stack for packet-switched services

The protocol stack is more complicated for the transmission of user data over packet services (Figure 4.29).

The GGSN is shown on the far right: the IP tunnel mentioned in Section 4.7 is established through the activation of the PDP context. IP packets destined for the mobile station (top protocol stack in GGSN) are packed by the GGSN into what is called a *GRPS Tunnel Protocol User Part* (GTP-u) protocol that

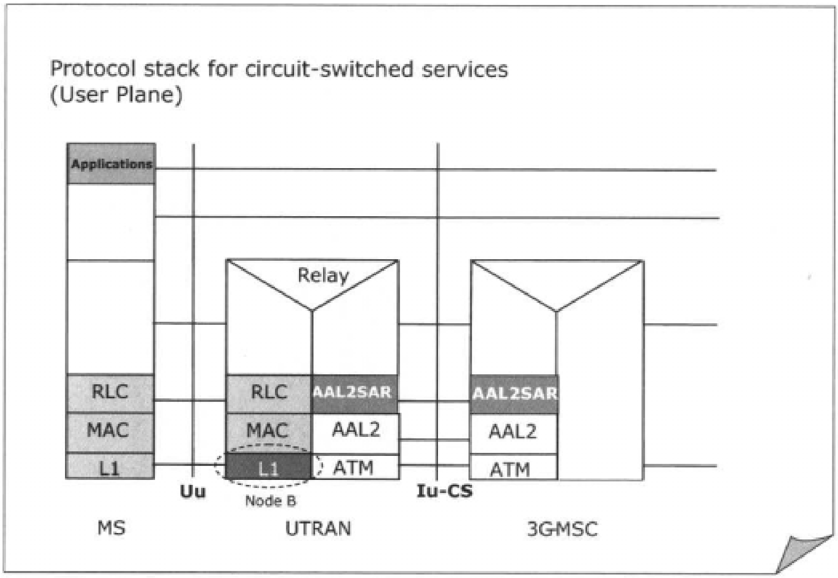

Figure 4.28: Protocol stack for circuit-switched services

transports the data through the UMTS network to the RNC. In addition to IPv4, GTP can also transport other packet data protocols, such as IPv6 and OSP.

For transport through the UMTS network GTP uses an internal IP network that is logically totally independent of the external IP network. The transport protocol is the *User Datagram Protocol* (UDP). There are plans also to use the *Transport Control Protocol* (TCP) for transmission although it reacts very sensitively to variable packet delays. Since TCP is already usually employed for reliable end-to-end connection at the application level, the *User Datagram Protocol* (UDP) is used in normal circumstances.

In the RNC data is transferred to the *Packet Data Convergence Protocol* (PDCP) sublayer that compresses the IP header data and then transmits the data packets over the RLC, MAC and PHY layers in the UTRAN protocol stack over packet data channels to the mobile station.

The IP tunnel shown in Figure 4.29 is actually two tunnel connections that are directly coupled together in the RNC: The PDCP layer provides a transmission medium for IP packets over the radio interface up to the RNC. The RNC packs the data directly into the GTP protocol for transmission to the GGSN.

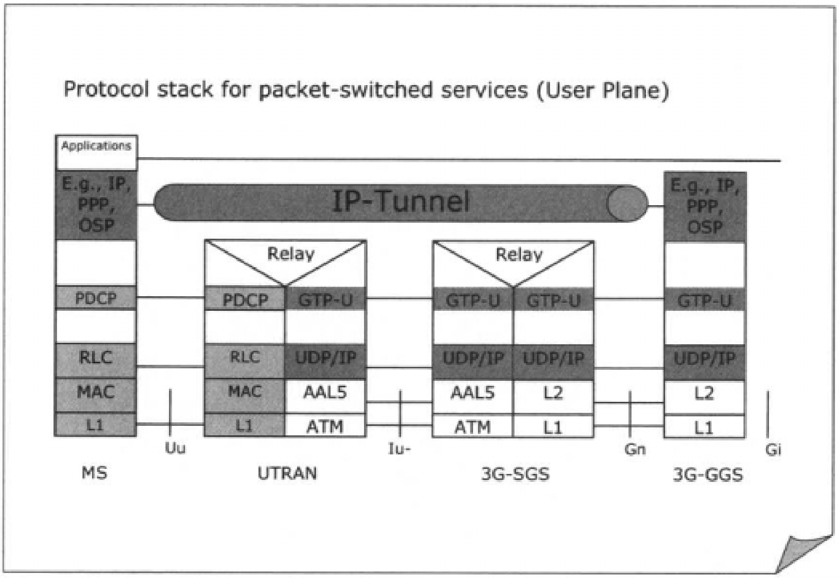

Figure 4.29: Protocol stack for packet-switched services

This protocol architecture is similar in structure to the one in GPRS, although the GTP protocol in UMTS is used up to the point of communication with the RNC. In GPRS, the SGSN is responsible for packing the data out of the GTP protocol into the *Sub-Network Dependent Convergence Protocol* (SNDCP), and the BSC is not required to convert any IP protocols.

4.10 Pure IP core network architecture

A new CN architecture in which parts of an existing node are replaced by a pure IP solution is being planned for Release 5 of UMTS [27]. Figure 4.30 illustrates what such a network could look like:

The nodes of the PS domain replace the tasks of the nodes of the CS domain. The various database functions are combined into a new database called the *Home Subscriber Server* (HSS). All services, circuit-switched as well as packet-switched, are supplied on the basis of the Internet protocols. Consequently, the Iu-CS interface to the otherwise unchanged RAN is no longer necessary.

The transmission of signalling and user data over the internal IP network takes place within the CN. As in R99/R3, IP tunnels are established over the SGSN and the GGSN. Once these tunnels have been established, data can be

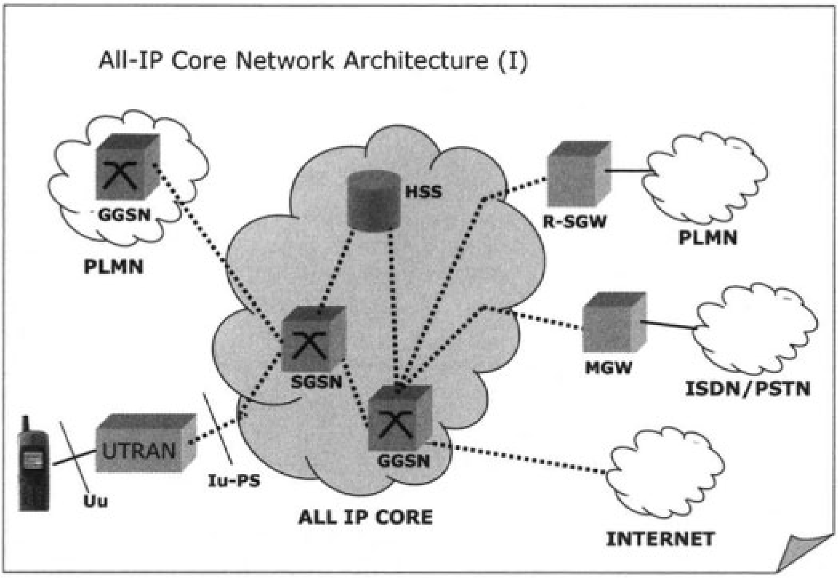

Figure 4.30: Pure IP core network architecture

forwarded to the Internet. Voice data is converted *Voice over IP* (VoIP) voice data and forwarded from the GGSN to a *Media Gateway* (MGW). There the voice packets are converted into conventional voice data that can be transmitted in the ISDN/PSTN networks.

For the transition to mobile radio networks without IP signalling, the protocols for call set-up, mobility management, and so forth also have to be converted into internal IP signalling. This task is handled by the *Roaming Signalling Gateway* (R-SGW). Mobile radio networks also based on this architecture can be attached without additional gateways.

The pure IP-CN is shown in more detail in Figure 4.31. The solid lines represent the channels over which signalling and user data can be transmitted, whereas the dotted lines are only used for signalling. The new central signalling node is what is called a *Call State Control Function* (CSCF). This node communicates with other CSCF nodes and the mobile station over what is the currently standardised *Session Initiation Protocol* (SIP) protocol for the switching of services over IP signalling. Yet the CSCF accesses the databases of the home subscriber server in which all user-specific and service-specific data is filed.

The switching functions for circuit-switched services - for example, the telephone network - are provided over the MGW node. The signalling is trans-

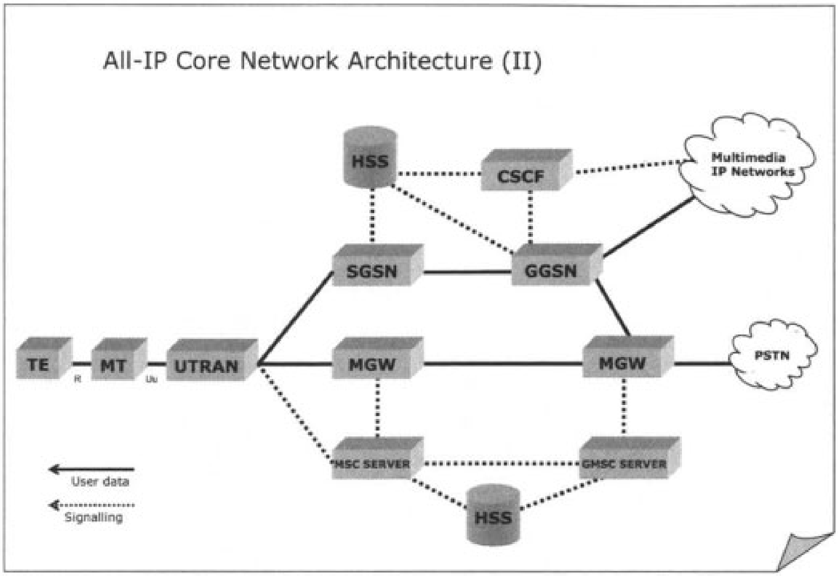

Figure 4.31: Pure IP core network architecture

ferred to the *Mobile Services Switching Centre* (MSC) servers that also communicate with the HSS.

The SGSN and the GGSN nodes are already familiar to us from R99. They are necessary for setting up IP connections over the RAN to the UE. In the process, the SGSN handles the communication with the RNC in the RAN and ensures that the mobility of the mobile stations is dealt with. The GGSN routes the traffic to the regional SGSN responsible, checks authorisations and routes the traffic between UE and the Internet.

4.10.1 Session Initiation Protocol (SIP)

An example is used to illustrate the signalling over the *Session Initiation Protocol* (SIP) [20]. Figure 4.32 shows four mobile radio networks based on R5: User A (bottom right) is currently a visitor in a mobile radio network (grey cloud on right) and would like to set up a call to user B. The following steps are necessary:

1. First, the user registers in the visited network and activates a PDP context. This allocates an IP address to the user who can then transmit IP packets. A search function allows the user to make contact with the local *Call State Control Function* (CSCF) and convey its request. Since

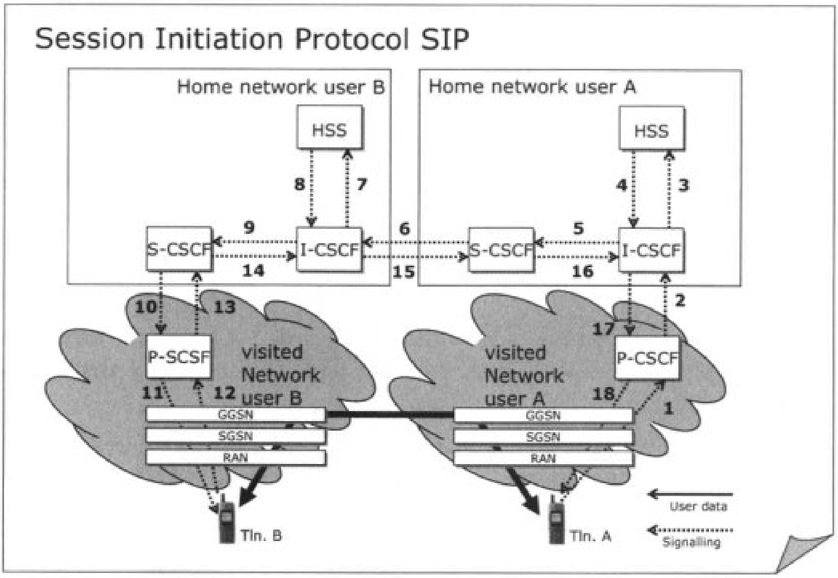

Figure 4.32: Session Initiation Protocol

this CSCF forwards the data to the user's home network, it is also called a *Proxy Call State Control Function* (P-CSCF).

2. The P-CSCF forwards the request to the *Interrogating Call State Control Function* (I-CSCF) in the home network of user A. This node represents a gateway to neighbouring networks over which such requests have to be filtered and processed.

3. The I-CSCF interrogates the central database HSS to find out how to continue with the connection set-up request. It is here that the authorisations of user A are checked. If the requested telephone number of user B is a special number (abbreviated dialling, emergency call, and so forth), the HSS can modify the destination information that is being supplied to the I-CSCF. This would, for example, allow the implementation of an 0800 number.

4. The response of the HSS is transmitted back to the I-CSCF. The I-CSCF can now forward the call to an available node that then continues with the call set-up procedure.

5. This available node is called a *Serving Call State Control Function* (S-CSCF). This function in the home network becomes immediately responsible for the further connection sequence. Using the destination telephone number sent by the I-CSCF, the S-CSCF identifies the home

network of the user being called and establishes a connection with the relevant gateway, the I-CSCF.

6. The I-CSCF receives the call and also has to send a database query to the local HSS with regard to the current location area of the user being called.
7. This query from the I-CSCF reaches the HSS. The database checks the authorisations of user B.
8. If the results of the check are positive, the HSS transmits the current location area of the called user back to the I-CSCF.
9. For its part the I-CSCF can now identify an S-CSCF and forward the call to it for further processing along with the results of the database interrogation.
10. The transmitted data enables the S-CSCF to detect the current location area of user B and it then sets up a connection to the local signalling node in the network being visited.
11. The P-CSCF in the network being visited by user B receives the query. Since user B is already registered in the network, a logical connection exists to this user. The user then receives the connection request over this logical connection.
12. The response (acceptance/rejection of connection request) together with the IP address of the user called is transmitted back over the same route (12, 13, 14, 15, 16, 17, 18) to user A.
13. Based on the identified address of user B, user A can establish a direct IP connection to this user and exchange data.

Thus the CSCF can assume three roles in the signalling: P-CSCF, I-CSCF and S-CSCF. However, physically the same node is involved. The signalling between UE and P-CSCF as well as between the various CSCF is handled over the SIP.

4.10.2 IP core network - pros and cons

There is some controversy among the standardisation bodies about the new CN architecture. The industry, however, is split into two minds: one view is that the upgrade is neither fast enough nor does it go far enough. The other view is suspicious of IP and the fear is that the high expectations cannot be met. There are solid arguments to support both views (Figure 4.33). The pros of the new architecture:

- Many networks are currently being converted to an IP basis. A large number of experts consider this architecture to be modern and forward-looking.

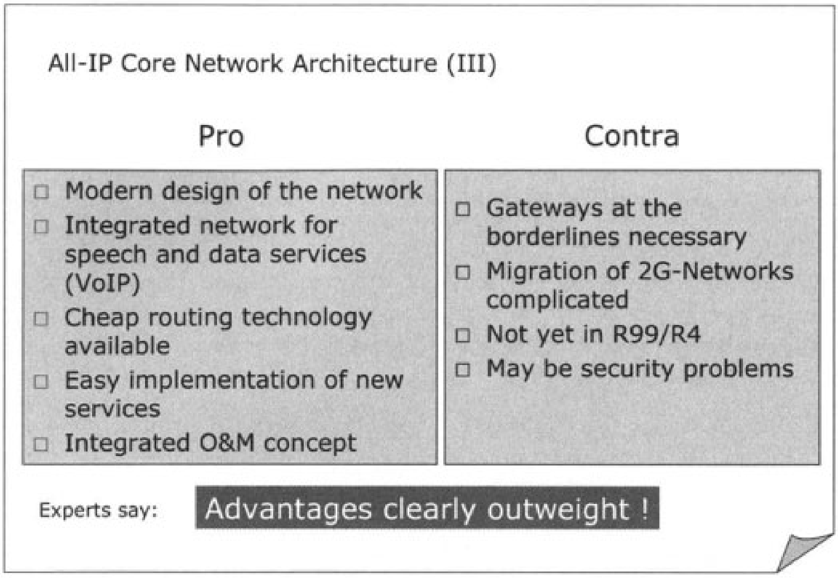

Figure 4.33: Pros and cons of a pure IP core network architecture

- An integrated network for voice and data services is what is needed to enable the offering of services. For example, one could imagine augmenting an instant-messenger service (e.g., ICQ) with voice and video telephony that could be linked up as necessary. But it would also be easy to see *Computer Telephony Integration* (CTI) with such a concept. Many company telephone systems are currently going down a similar path. Telephones are directly linked to a LAN and can communicate with the computers.
- A reduction in the cost of switching technology is anticipated due to the large number of routing technologies for IP that exist. However, many products still do not support all IPv6 possibilities. The price levels could be brought in line with the introduction of these performance features.
- New services are easy and convenient to introduce due to the flexible layout of the signalling technology and the easy scalability of the bandwidth of the direct connection between terminals.
- A multitude of network monitoring options already exist for the current Internet technology. These can easily be integrated into a new architecture.

On the other hand, the following arguments stack up against a new CN architecture:

- The new architecture requires new nodes on the outer boundaries for the implementation of gateways to existing networks.
- The existing second generation networks were not designed for this architecture and consequently cannot easily be integrated into it. Furthermore, the initial UMTS networks were based on GSM so that operators will be required to make a higher investment for CN since large parts of the network will no longer be usable in the future.
- The issue of security arises as a result of the direct coupling of networks with the Internet. In comparison with classic signalling systems, many more people in the world will try to gain access to IP networks. Consequently, the security systems will have to be established and developed with precise specifications.
- Many consider the transmission of voice over VoIP technology as being unsophisticated and of poor quality. The traffic-related changes possible to data transmission speed (jitter) are already creating problems in existing networks. The upgrade to IPv6 could be helpful.

Overall, experts say that most of the new possibilities offered by the enhanced CN more than make up for the associated risks and problems. Since this area of the standard changes particularly quickly, the most current version of the standard should be used for any further questions.

4.11 Questions

4.1 What is the purpose of the separation between the permanently installed infrastructures in RAN and CN?

4.2 Give two advantages for dividing a mobile unit into UE and USIM.

4.3 What are the advantages of packet-switched transmission? Which effect is made use of? Which problem arises with packet-switched transmission when services require a certain quality of service?

4.4 Which elements make up a RAN? What are the functions of the different nodes?

4.5 Explain what the tasks are of the different databases in a CN.

4.6 Explain the difference between an MSC and an SGSN.

4.7 List the three basic handover types in UMTS. What are the advantages of using the new handover types?

4.8 Explain the roles of an SRNC and a DRNC during a handover.

4.9 Why is there a difference in the accuracy of how the position of a mobile station is stored in a network depending on whether the services are packet-switched (PS) or circuit-switched?

4.10 When does a *location area update* or a *routing area update* occur?

4.11 Between which network nodes is the *GPRS Tunnelling Protocol* (GTP-U) used? What is its function?

4.12 Comment on the following statement: Using a completely IP-based fixed network infrastructure makes it easier for a network operator to introduce new services.

4.13 List three problems that can occur when an IP-based fixed network is introduced.

4.14 What is the *Session Initiation Protocol* (SIP) used for? How does user data flow in the case of an SIP-switched connection?

5 The Protocol Stack at the Radio Interface

5.1 The ISO/OSI reference model

The ISO (*International Organization for Standardization*) divided the functions of protocols into categories, called layers, and designed the *Open System Interconnection* (OSI) layer model in order to facilitate the understanding of complex digital protocols. With this model each layer takes on special, largely defined tasks in order to provide a service to the layer directly above it. To perform these tasks, it uses the layer directly below it [23].

This recursive approach is illustrated in Figure 5.1. The *protocol stack* is divided into seven layers. It exists with the transmitter as well as with the receiver of messages. Each layer in the transmitter has a corresponding layer in the receiver with a logical connection existing between the two.

A well-known pictorial explanation of the ISO/OSI protocol stack relates to the story of two philosophers: one, Philosopher A who lives in India, and another one,Philosopher B, who lives in Brazil. A gets the idea of notifying B of a new idea. However, A cannot read or write and so he dictates the formulation of the idea to his secretary in Hindi. The secretary writes down the idea and after correcting it passes it on to a translator who translates it into English. The translator then sends the document back to the secretary's office. There a covering letter is prepared and together with the English document inserted into an envelope. This envelope is taken by the secretary's office to the mailroom, from where it is finally taken to the Indian post office. In conjunction with other agencies it interfaces with, the post office arranges for the letter to be transported all the way to the office building of the Brazilian philosopher B. A week later the letter arrives in the mailroom there and passes through the same chain of events in reverse: mailroom, secretary's office, translation from English to Portuguese. B's personal secretary finally has the letter in her hands. Since B can neither read nor write either, the secretary reads the ideas of A to B. B then reflects for a while and then dictates a response to his secretary.

Thus a logical connection is set up between the two philosophers that otherwise would have no direct possibility of communication, allowing them to exchange ideas. A logical connection also exists between the two secretaries. Each unit only communicates with the unit either directly above it or below

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff
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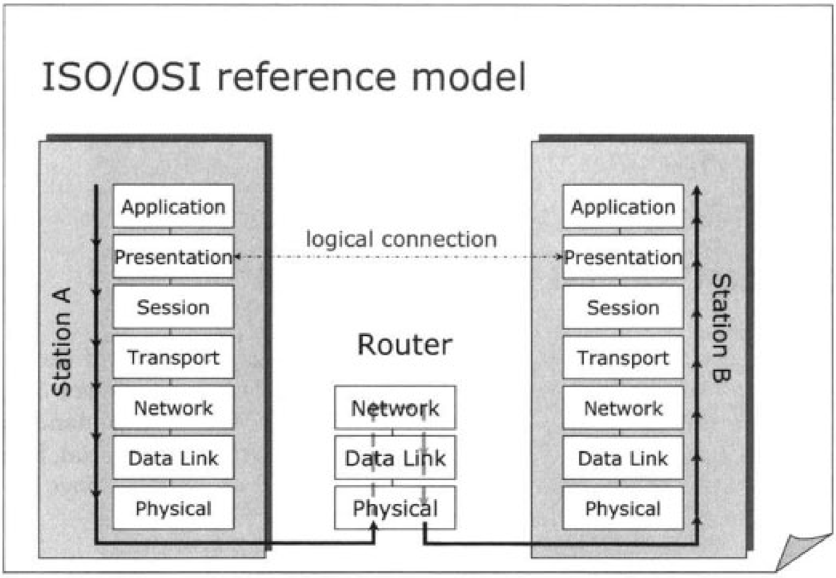


Figure 5.1: The ISO/OSI reference model

it. Only the Indian post office, which is responsible for the physical transport of the letter, has direct contact with other postal agencies.

Although the OSI protocol stack has not generally established itself *vis-à-vis* the TCP/IP architecture, it is an excellent reference model that facilitates the understanding, representation and classification of protocols.

The seven layers have different tasks that are briefly explained below:

1. The physical layer is responsible for direct communication with a partner over a common physical medium. In the example of the two philosophers, the letter corresponds to the physical layer.

2. The data link layer above it is often divided into two sublayers: the *Medium Access Control* (MAC) sublayer is responsible for coordinating access to a common medium, whereas the *Logical Link Control* (LLC) layer above it handles error correction.

3. Above the data link layer is the network layer, which is responsible for transmitting messages over several intermediate stations.

4. Above these three lower layers are the layers where logical connections already run from one end to another. The transport layer guarantees a secure end-to-end connection.

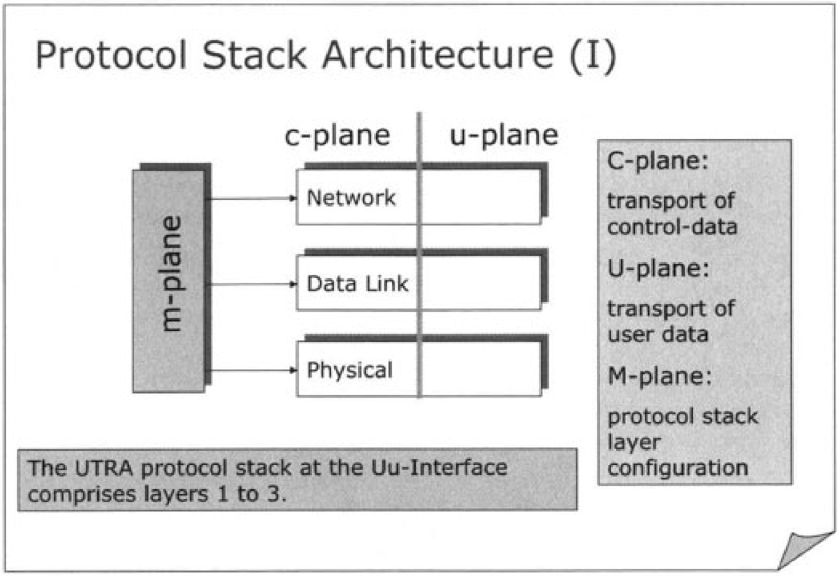


Figure 5.2: Separation of protocol stack into c-plane and u-plane

5. Above it is the session layer that is responsible for opening and terminating communication.
6. The presentation layer converts the information being transmitted so that the remote station can understand the presentation. The translator in our example works within the presentation layer.
7. At the top of the protocol stack is the application layer that represents the interface to the user. The personal secretary of the philosopher is a component of the application layer, whereas the philosopher himself works as a user outside the protocol stack.

The protocol stack at the Uu-interface between the UE and the UTRAN comprises only layers 1-3. All higher functions are negotiated directly between UE and CN based on the communication paths provided by these layers. Three different types of tasks can be distinguished that are handled by the protocol stack (see Figure 5.2):

- Exchange of control information: Before direct communication can take place between participating partners, the partners must agree on common parameters. In our example the two translators both talk English so this language can be used in the presentation layer. If a new philosopher should start to participate in the discussion in Paris, the translators will again have to agree on a common language. This communication is totally invisible to the philosophers and takes place in the control (control plane).

- Exchange of user information: The actual user data is transmitted in this plane. The exchange of information from our example was totally restricted to the user (user plane).
- Configuration of individual layers: Whereas in the other two planes both communication partners communicate with one another, the management plan includes the functions necessary for the internal configuration of the protocol stack. Back to our example: the personnel department that hires the translators belongs to the *management plane.*

The individual layers of the protocol stack demarcate different task areas, whereas the planes categorise the information flow.

5.2 The UTRA protocol stack

Figure 5.3 shows the protocol stack at the Uu-Interface. The physical layer, which provides *transport channels* at the interface to the MAC layer above it, operates down at the bottom. The MAC layer converts the data transferred over *logical channels* from above so that it can be transmitted over the transport channels. Above the MAC layer is the *Radio Link Control* (RLC) sublayer, which is mainly responsible for error protection and error-free data transmission.

So-called *radio bearers* are normally directly available at the upper edge of the RLC sublayer. These are the channels through which data is transported through the access stratum. In the user plane the two sublayers *Packet Data Convergence Protocol* (PDCP) and *Broadcast and Multicast Control* (BMC) can stack on the RLC sublayer. Adaptations to packet data protocols such as TCP/IP above can be made in the PDCP sublayer, whereas the BMC sublayer is responsible for point-to-multipoint connections.

The sublayers mentioned above the physical layer all belong to layer 2. Control and user data are jointly transported as far as this layer. In layer 3 the *Radio Resource Control* (RRC) sublayer, which is responsible for the configuration of all other sublayers within the UTRA protocol stack, operates in the control plane.

The so-called *duplication avoidance* layer that is responsible for ensuring that data is correctly transmitted in the case of an *SRNS relocation* operates way at the top. All other functions, such as call control for the set-up of end-to-end connections beyond the RAN, are located outside the UTRA protocol stack. Details about the functions of the individual sublayers are provided below.

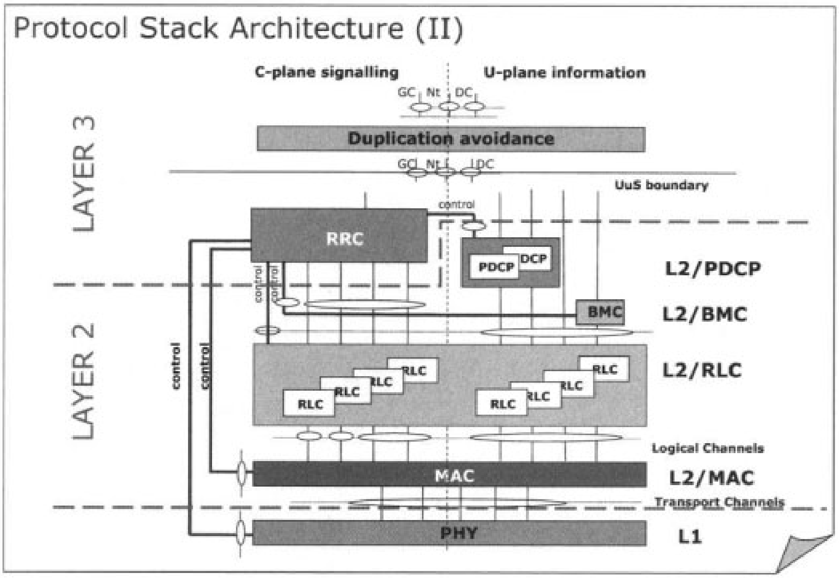


Figure 5.3: The UTRA protocol stack at the Uu-Interface

5.3 The physical layer

The physical layer is implemented in Node B within the RAN. This layer handles all tasks that are directly related to the transmission of data over radio. The CDMA technique is explained in Chapter 6.

At the upper edge the physical layer provides transport channels that are first mapped to the internal *Coded Composite Transport Channel* (CCTrCH) and then to the ***physical channels***. Error protection of the data is carried out by *Forward Error Correction* (FEC) and an *interleaving* of the bits. The level of error protection is configured by the RRC sublayer.

The physical layer works directly at the actual radio unit. Consequently, tasks such as synchronisation and fast power control have to be carried out here. Functions necessary for soft handovers and softer handovers are also implemented. This is sometimes also referred to as macro diversity.

Since the parametrisation of the physical layer is undertaken by the RRC sublayer, the physical layer periodically and upon request must report the current situation to the RRC sublayer. This includes reporting the calculated *Frame Error Rate* (FER), the *Signal to Interference Ratio* (SIR) and the measured interference power. On the basis of these measured values decisions can then be made in the RRC sublayer about the further sequence of a connection.

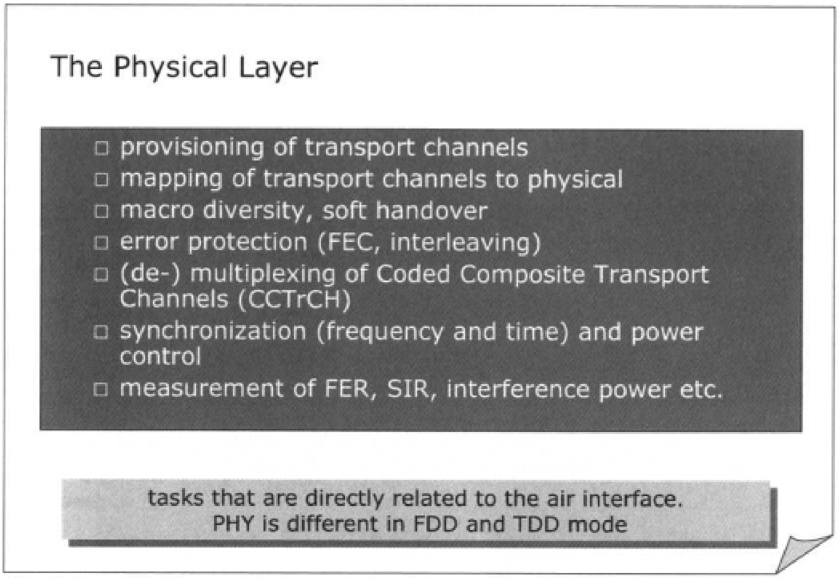


Figure 5.4: Tasks of the physical layer

In summary, it should be noted that all tasks that are directly related to the radio interface are located in the *Physical Layer* (PHY) layer(see Figure 5.4). Consequently, the physical layer is structured differently in UTRA-TDD than it is in UTRA-FDD.

5.4 The MAC layer

The MAC layer coordinates access to the physical medium over which data is transmitted. This means that the MAC layer contains queues in which the different data streams being transmitted are placed. The decision is made in the MAC layer as to which queue the next data packet for transmission to the physical layer should be placed. Along with the filler state of the individual queues, a sophisticated priority control that is parametrised by the quality of service negotiated at connection set-up is also an important aspect of this process. The MAC layer also supplies the current state to the RRC sublayer and receives configuration instructions from it. Under certain conditions (transparent RLC mode) the MAC sublayer is also responsible for the encryption of the data. However, the data packets delivered by the RLC layer above the MAC layer already have to be the right size, because the MAC layer does not carry out any further segmentation.

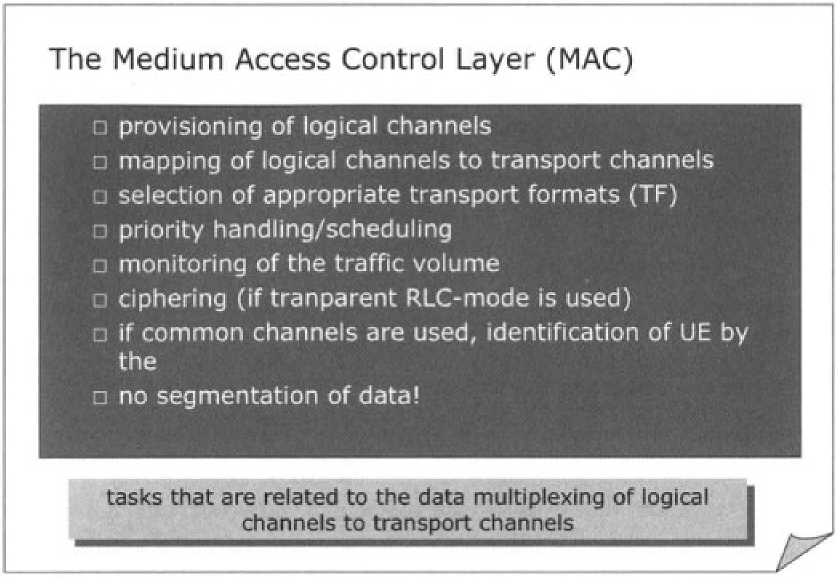


Figure 5.5: Tasks of the MAC layer

With *common channels*, in other words, channels the use of which is shared by several stations, addressing has to be built into the data stream. The addressing enables receiving stations to identify whether a transmitted data packet is destined for the respective station. This address, called *Radio Network Temporary Identifier* (RNTI), is used by the MAC layer to make this selection.

The RLC layer delivers the data to the MAC layer over *logical channels*. A logical channel describes which type of data should be transmitted. The MAC layer maps these logical channels to the *transport channels* that represent the interface to the physical layer. The transport channels are configured over so-called *transport formats* that are explained in Section 5.10. The MAC layer is thus responsible for the multiplexing of several parallel data streams from the logical channels to the transport channels (see Figure 5.5).

5.5 The RLC layer

The second sublayer of the data link layer is the *Radio Link Control* (RLC) sublayer. The task of this layer is to protect the different data streams from errors based on the respective requirements. For this purpose the RLC layer can work in three modes (see Figure 5.6):

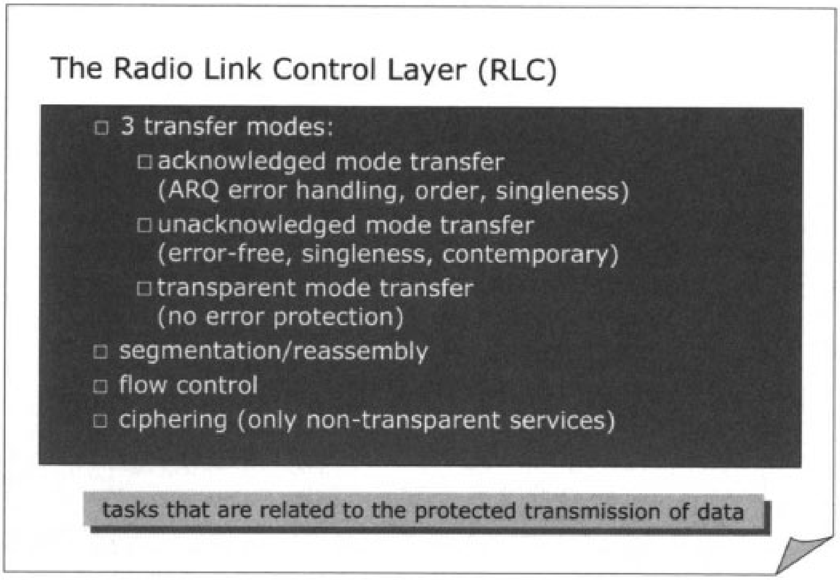


Figure 5.6: Tasks of the RLC layer

- In *acknowledged mode* the RLC layer sends a new request for data blocks detected to have errors. Since this delays the delivery of data already received until missing data packets are redelivered, this procedure is not suitable for error-tolerant video or audio data, but is suitable for data transmission. Through error correction the RLC layer can guarantee that data is delivered error-free only once and in the right sequence.
- In contrast, the RLC layer provides no error correction in *unacknowledged mode.* Instead it discards data packets that are detected as having errors or selects them as being defective before they are delivered. Data is always provided with a sequence number. Consequently, the RLC layer can ensure the uniqueness of the transmitted data in unacknowledged mode.
- In *transparent* mode the RLC layer does not add a separate *header* to the data. Instead it simply forwards the data to the MAC layer. This mode is especially suitable for the transmission of stream data such as video and audio data.

In all cases the RLC layer can segment data packets delivered by higher layers so that the MAC sublayer receives data packets of the right size over the logical channels. This simplifies the queue management in the MAC layer.

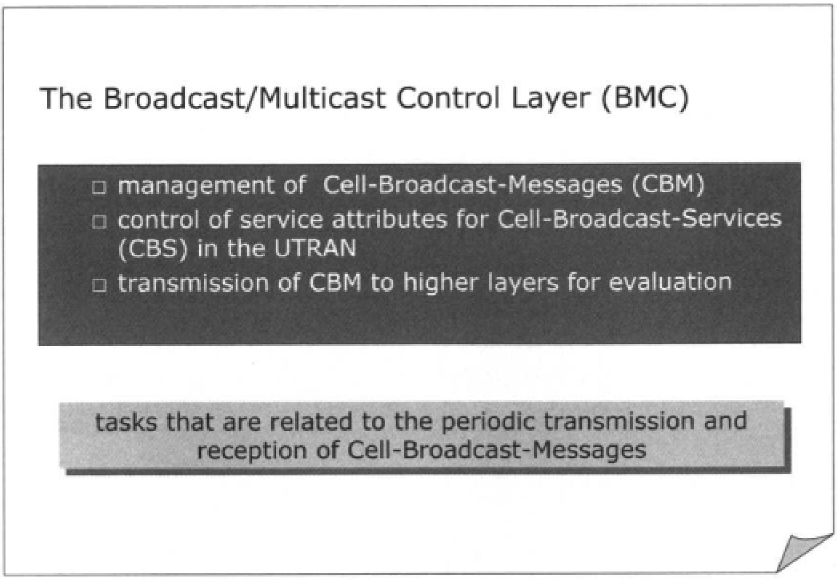


Figure 5.7: Tasks of the BMC layer

The RLC layer is also responsible for the encryption of data. This task is only taken over by the MAC layer in transparent mode. It carries out flow control for the application working above it so that no overflow occurs within the UTRA protocol stack.

The RLC layer is thus responsible for transmitting data from the higher layers with the necessary security. The RRC layer already selects the appropriate mode for a connection at the time of connection set-up and configures the necessary parameters in the RLC layer. The service provided by the RLC layer to an application working above it is also called a *radio bearer*. However, this does not apply if the data is still being routed through the PDCP or BMC sublayers.

5.6 The BMC layer

Functions such as cell broadcast that are necessary for point-to-multipoint applications were transferred to the *Broadcast and Multicast Control* (BMC) sublayer (see Figure 5.7). These are functions with which certain data in a cell can be broadcast. This service already exists in GSM but is hardly used commercially. The BMC sublayer in the RNC manages the information that is to be broadcast in the individual cells.

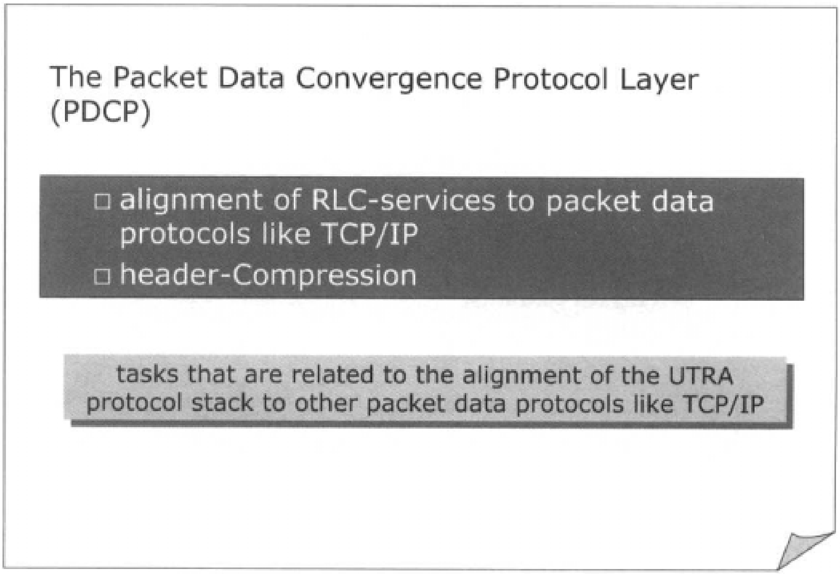


Figure 5.8: Tasks of PDCP layer

5.7 The PDCP layer

The *Packet Data Convergence Protocol* (PDCP) layer adapts the packet data protocols working on the UTRA protocol stack to this layer (see Figure 5.8). Its most important application is the transmission of IP data. For this protocol the PDCP layer carries out a compression of the header information (header compression), which is particularly important for small data packets that occur with a VoIP connection. An IPv4 header has a length of 20 bytes, whereas a typical VoIP packet only contains 20 bytes of user data. The compression reduces the length of the header to a small number of bytes, thereby contributing towards increased efficiency and a higher data rate.

5.8 The RRC layer

The *Radio Resource Control* (RRC) sublayer is in the control plane in layer 3 and contains all algorithms necessary for the configuration and operation of the UTRA protocol stack (see Figure 5.9). On the RAN side this includes the configuration of the ***broadcast channels*** that are broadcast in each cell. On request the RRC sets up the ***radio bearers*** over which applications can exchange data between UE and RAN. At the same time it configures the

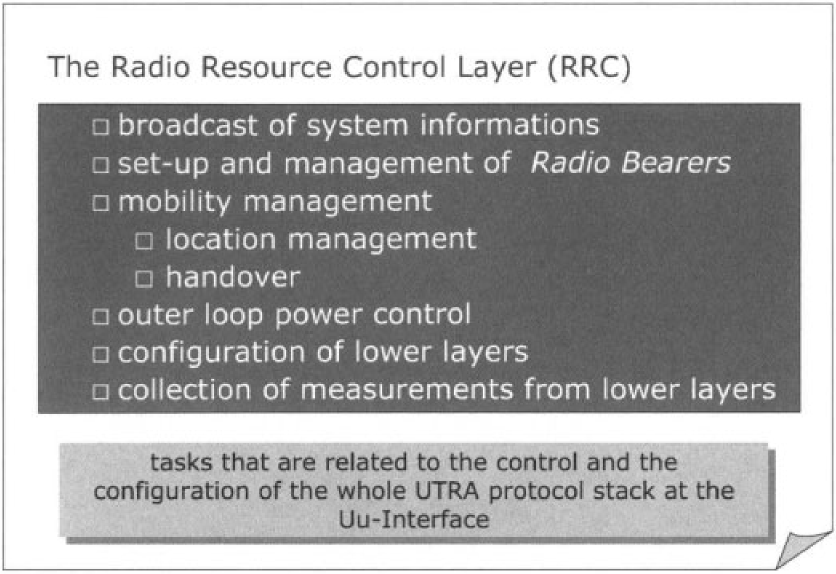


Figure 5.9: Tasks of the RRC layer

RLC, MAC and the physical layers in accordance with the required quality of service parameters.

The RRC layer is also entrusted with the mobility management for active connections. Consequently, it carries out the necessary updates of the data structures in the RAN (location update), roaming and handover. Since the information from all connections in all attached cells merges together in the RRC layer in the RAN, the RRC layer can allow complex algorithms to be executed for an optimal distribution of radio resources. This results in a target value for the transmitter power (*outer loop power control*) of each connection. These values are transmitted to the respective Node Bs and mobile stations. There an attempt is made to achieve and maintain this value (*inner loop power control*).

To carry out its tasks, the RRC layer collects measured values from all other layers in order to generate configuration instructions for other layers using suitable algorithms. Due to the large number of functions, the documentation for the protocol between RRC layers is very extensive and almost 600 pages long.

Since the algorithms within the RRC layer have a direct effect on the performance of a network, different vendors can be differentiated by the performance of their algorithms and consequently the networks based on them.

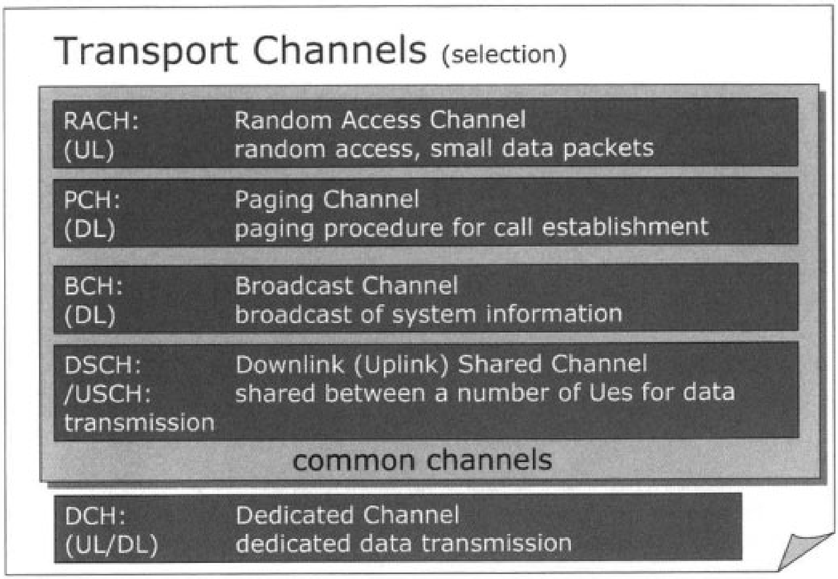


Figure 5.10: Transport channels

5.9 Transport channels

Data is transmitted over ***transport channels*** at the interface between MAC layer and physical layer. Figure 5.10 lists examples of some transport channels.

Random access takes place over the *Random Access Channel* (RACH). Small amounts of user data can be transmitted over this channel simultaneously. The exact random access sequence at the physical plane is described in Section 8.2.4. The RACH only exists on the uplink.

The *Paging Channel* (PCH) is used to reach a mobile station that is not currently maintaining an RRC connection with the RAN. The address of the mobile station is paged once or several times. If the mobile station receives a request to set up a connection, it carries out random access over the RACH and sets up a connection.

The *Broadcast Channel* (BCH) is used to transmit system information. This information is stipulated in the RRC layer and transferred to the RLC layer for transmission. It subsequently is forwarded on the downlink over the logical channel *Broadcast Control Channel* (BCCH) to the MAC layer and from there over the BCH to the physical layer. Here the data is broadcast over the corresponding physical channels. The procedure in the mobile station is precisely the reverse: The data is transferred from the physical layer over the MAC and RLC layers to the RRC layer.

As the name implies, ***shared channels*** are used jointly by several stations, including mobile stations. The *Radio Network Temporary Identifier* (RNTI) mentioned earlier is always included in the transmission to clarify who the recipient is of the traffic being transmitted over these channels

The normal transmission of user and signalling data takes place over the *Dedicated Channel* (DCH). This channel can be set up on the uplink as well as on the downlink. In this case the sender and receiver are fixed, so no further addressing is required within the connection.

5.10 Transport formats

As a rule, not only one channel but instead several are set up between the MAC and PHY layers. The radio interface from UMTS offers a high degree of flexibility. It is the task of the UTRA protocol stack to enable an application to exploit this flexibility.

Figure 5.11 shows three DCH connections that have been set up between the MAC and the physical layer. The channel DCH1 transmits voice data, whereas DCH2 transmits the related video data. DCH3 is used for the transmission of signalling data. These different applications expect a different quality of service from the physical layer. The transport channels provide a sophisticated mechanism for parametrisation to ensure maximum flexibility.

Data packets that are exchanged over transport channels are called ***transport blocks***. A transport block comprises a certain quantity of data. Several transport blocks of the same size can be transferred simultaneously per transport channel. This group of transport blocks is called a *Transport Block Set* (TBS). The time by which a TBS must be transmitted is called a *Transmission Time Interval* (TTI). This interval can vary in length for the different transport channels but always has to be a multiple of the frame length of 10 ms.

A TBS is described by a *Transport Format* (TF). The TF specifies the size of the individual blocks of a set and how many blocks are contained in a set. This information is filed in the dynamic part of the TF. This is joined by a semi-static part that specifies the type of error correction to be used in the physical layer, which coding rate applies, and so forth. The transport format thus precisely specifies what should happen with the TBS in the physical layer. However, the transport channel is not allocated just any transport format: for each TBS the MAC layer can select a transport format from a group that is allocated to the transport channel. This group is called a *Transport Format Set* (TFS) (see Figure 5.12). Since a TFS specifies in which domains the physical parameters of a connection are permitted to change,

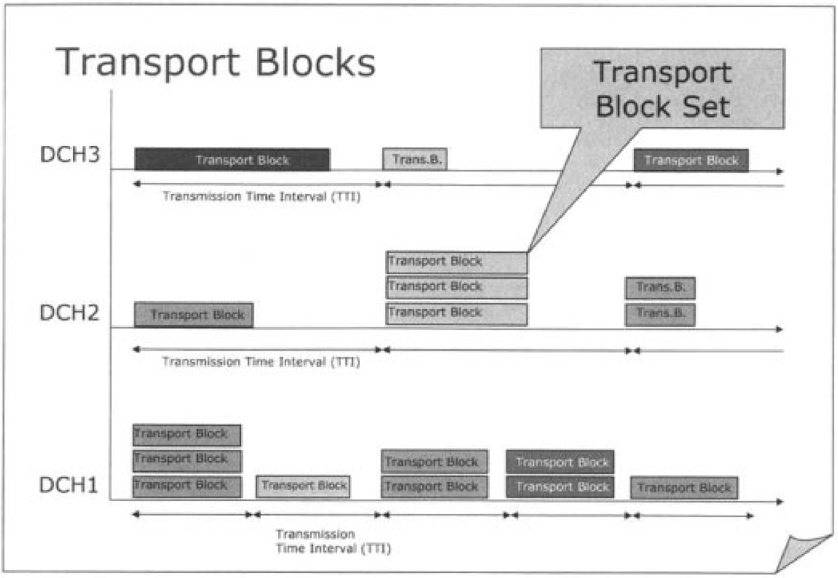


Figure 5.11: Transport blocks and transport block sets

the TFS is dependent on the requested application. A service that requires a constant data rate can only contain a single transport format in the transport format set. A service with a variable data rate possibly has a choice of a large number of transport formats, thereby always enabling it to adapt the parameters of the physical layer quickly to the current data rate.

To review, a data block that is transmitted over a transport channel is called a transport block. Transport block sets are formed since several of these blocks can be transmitted simultaneously at one time. This set is described by a transport format that is selected from a group of permissible transport formats. This group is called a transport format set.

Transport formats cannot, however, be combined in any combination. Take the following example: an application that transmits data over a transport channel is guaranteed a high maximum data rate through the acceptance of a particular transport format into the associated transport format set. The same guarantee cannot then be given for this same point in time to a second application that is using another transport channel. The different transport formats that are used by the different transport channels at a particular point in time are not compatible (see Figure 5.13), because the available resources are limited. As a result, *Transport Format Combinations* (TFCs) that specify which transport formats are compatible with one another were introduced.

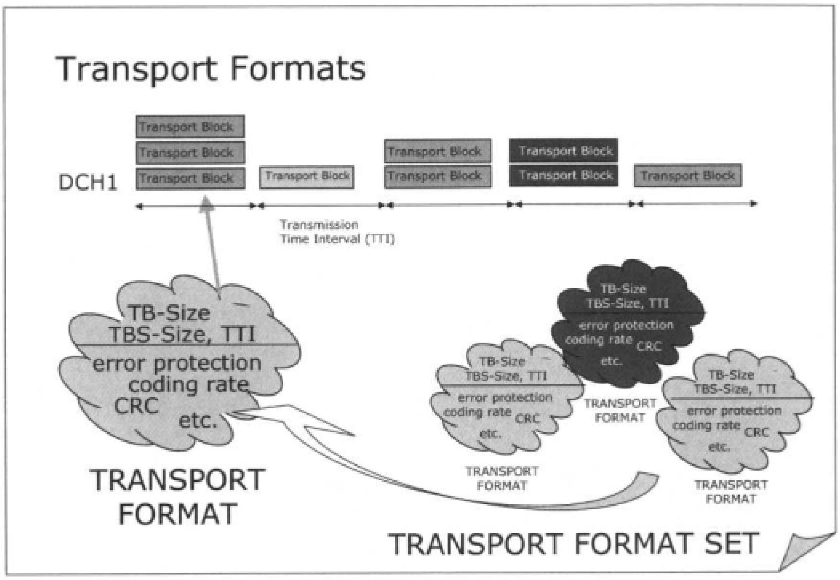


Figure 5.12: Transport formats and transport format sets

A transport format combination specifies, for example, that if transport channel 1 requires 80% of the resources at a particular time, the other two transport channels 2 and 3 can jointly use a maximum of the remaining 20%.

The *Transport Format Combination Set* (TFCS) groups several compatible transport format combinations together. These groups describe the possible transport format combinations from which the MAC layer can make a selection of one at a particular point in time (see Figure 5.14).

This complicated interface offers a big advantage: when a connection is set up, the RRC layer specifies which transport format combinations can be used for the connection. Depending on the data volume in the individual channels, in active operation the MAC layer can dynamically distribute the capacity easily and quickly between the different transport channels. It notifies the physical layer of this distribution over the *Transport Format Combination Identifier* (TFCI). This field is transported over the physical layer to the receiver. There the arriving data stream can be distributed among several transport channels again with the help of the TFCI field. Since the receiver also knows the list of possible transport format combinations, the index TFCI suffices for executing the demultiplexing.

To explain the tasks of the MAC layer in detail: the MAC layer receives data blocks over the logical channels from the RLC layer that divides them into transport blocks and transport block sets on the downlink and transmits them

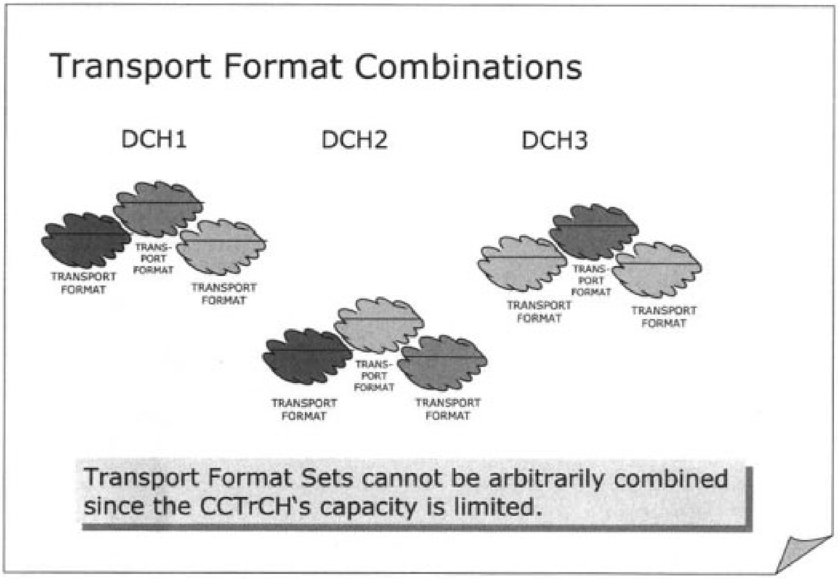


Figure 5.13: Transport format combinations

over the transport channels. On the uplink the received transport block sets are distributed over the different logical channels. The TFCI field from which the transport format to be used is produced for each transport channel is always the one transferred to the physical layer.

It is important that the RRC layer is able to detect the threat of any overflows within the UTRA protocol stack early enough. Consequently, the MAC layer supplies the following as measured values to the RRC layer: the achieved throughput, the TFCS used and the filler state of the individual queues.

5.11 Logical channels

Whereas transport channels are configured on the basis of *how* data is to be transmitted, with the logical channels it is the content that is important. User data is transmitted over the *Dedicated Traffic Channel* (DTCH), whereas control data is transmitted over the signalling channel *Dedicated Control Channel* (DCCH). A selection of other logical channels available is listed in Figure 5.15.

Figure 5.16 shows the possible mapping of logical channels to transport channels from the view of a RAN: The logical broadcast control channel BCCH is

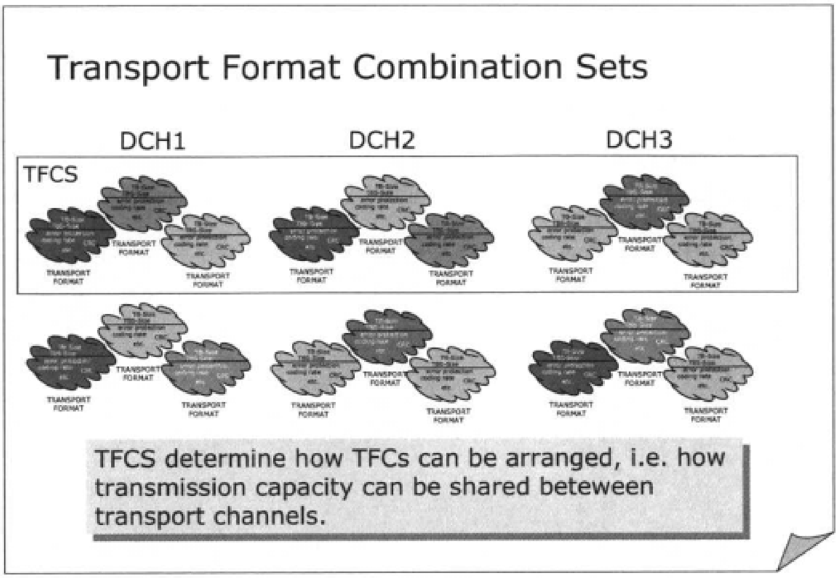


Figure 5.14: Transport format combination sets

mapped to the transport channel BCH. Since this channel only exists on the downlink, the illustration only shows an arrow pointing downward. The same applies to the paging channel *Paging Control Channel* (PCCH).

There are several transport channels in UMTS over which the user data channel DTCH and the signalling channel *Dedicated Control Channel* (DCCH) can be transmitted:

- over a dedicated transport channel *Dedicated Channel* (DCH)
- small data packets can be transmitted efficiently and without complicated signalling over the *Random Access Channel* (RACH) on the uplink and the *Forward Access Channel* (FACH) on the downlink
- over jointly used channels (*Downlink Shared Channel* (DSCH) on the downlink and *Uplink Shared Channel* (USCH) or *Common Packet Channel* (CPCH) on the uplink)

The *Common Control Channel* (CCCH) is used for signalling outside an existing connection and is transmitted over the transport channels RACH and FACH.

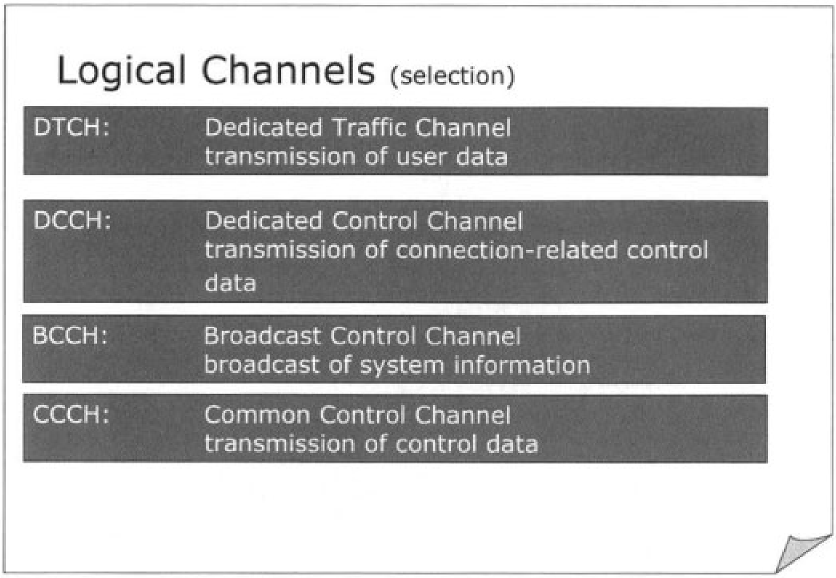


Figure 5.15: Logical channels

5.12 Questions

5.1 Explain the concept of the ISO/OSI layer model. What are the advantages to dividing a protocol stack into layers?

5.2 Why does a logical connection exist between the philosophers in Section 5.1?

5.3 Protocol stacks are also divided into *planes*. What is the purpose of the division *C/U/M planes*?

5.4 Which layers are comprised in the protocol stack at the Uu-Interface?

5.5 Describe the functions of the physical layer in an UTRAN.

5.6 Which types of channels are available to the MAC layer at the upper edge of the physical layer?

5.7 Name three tasks handled by the MAC layer.

5.8 Which functions are embedded in the RLC layer?

5.9 Explain the differences between the three transmission modes in the RLC layer.

5.10 Name three mechanisms that can basically be used for dealing with transmission errors.

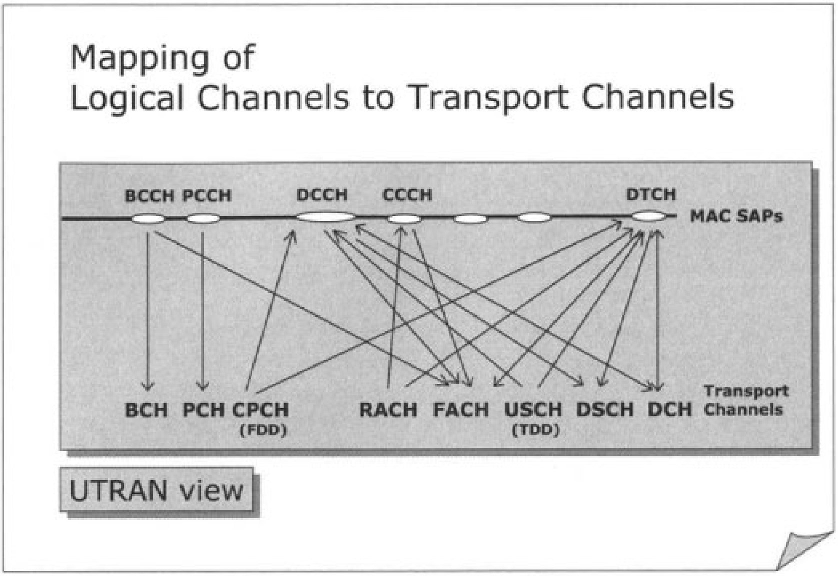


Figure 5.16: Mapping logical channels to transport channels

5.11 In which layer does data encryption take place?

5.12 Describe the functions of the PDCP sublayer. With which services are these functions used?

5.13 Describe the functions of the RRC layer.

5.14 Why is there a difference in the performance of a UTRAN depending on the vendor?

5.15 Allocate the different layers (PHY, MAC, RLC, RRC) to the network elements of a RAN.

5.16 Why were transport channels developed with the complicated mechanism of transport formats, etc.?

5.17 What distinguishes logical channels from transport channels?

5.18 Over which transport channels can user data (logical channel DTCH) be transported in UMTS?

5.19 Estimate how much higher the factor of complexity is of a UMTS terminal *vis-à-vis* a GSM terminal.

6 Data Transmission at the UMTS Radio Interface

This course unit deals with the fundamentals of wireless data transmission in UMTS. It begins with a general introduction to the two duplex techniques used in UMTS. This is followed by an overview of the multiple access techniques commonly used in mobile radio communication. Most of the course unit is devoted to the *Code Division Multiple Access* (CDMA) multiple access technique used in UMTS.

6.1 The UTRA radio interface

The radio interface of a mobile radio system is generally interpreted as the interface between the mobile station and the base station where the communication of which is wireless, thus over radio. The *UMTS Terrestrial Radio Access* (UTRA) radio interface is located between the radio access network, *UMTS Terrestrial Radio Access Network* (UTRAN), and the *Mobile Station* (MS) (see Figure 6.1). All techniques, functions and procedures described in this course unit relate to this radio interface. In the UMTS Standard it is also referred to as the *Uu-interface.*

6.2 Duplex procedures

A duplex procedure generally separates the transmit and receive signals of a station that is able to transmit as well as receive. The separation of transmit and receive signals prevents a situation in which a station receives its own transmitted signal and in some circumstances is not able to separate it from the required receive signal. This would make communication impossible (see Figure 6.2). Due to the long distances between transmitter and receiver typical of mobile radio and the resulting signal attenuation, the transmit signal is much stronger than the receive signal. This reinforces the need for a separation of uplink and downlink.

The duplex procedure thus separates the radio resources of an operator between the two transmission directions of uplink and downlink.

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff

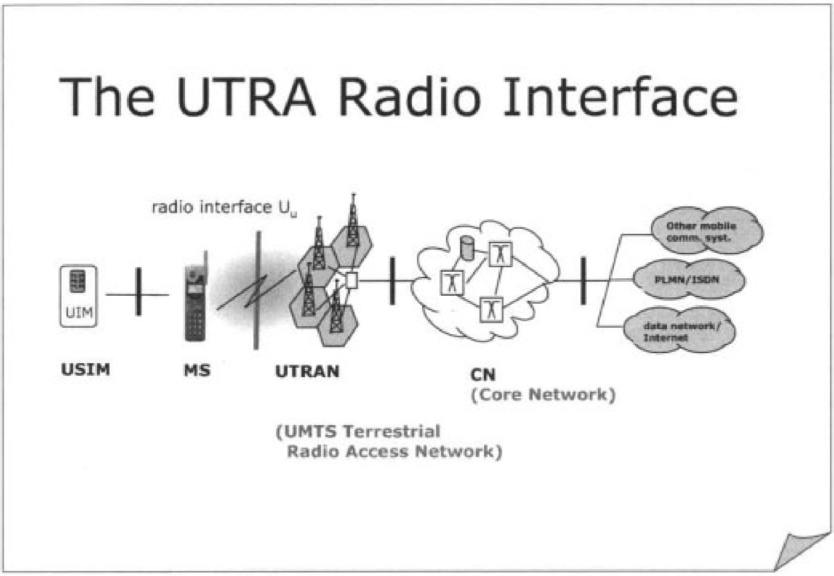


Figure 6.1: Radio interface in UMTS basic architecture

6.3 The frequency-division duplex technique

Frequency-division duplex (FDD) is a procedure that is widely used in the separation of uplink and downlink. With this technique stations transmit and receive in separate frequency bands. The transmitting band of one station then becomes the receiving band of the other station and vice versa. In cellular mobile radio the higher frequencies are normally selected as the transmitting band for the base stations and the lower frequencies as the transmitting band for the mobile stations. The reason is that electromagnetic waves of a higher frequency are more strongly attenuated by the propagation of the air than waves of the lower frequency. Therefore, the transmitting base station needs more power than the mobile station to bridge the same distance.

The distance between transmitting band and receiving band in Figure 6.3 is generally consistent within a system and is called the Duplex distance. The FDD duplex technique is particularly suited to mapping symmetric services, i.e., services that transmit about the same amount of data at the same data speed in both directions. The speech service, or video telephony, is an example of such a symmetric service. Even if the bandwidth of the transmitting and receiving bands can be selected in different ways independently of the FDD duplex technique in order, for example, to present asymmetric services.

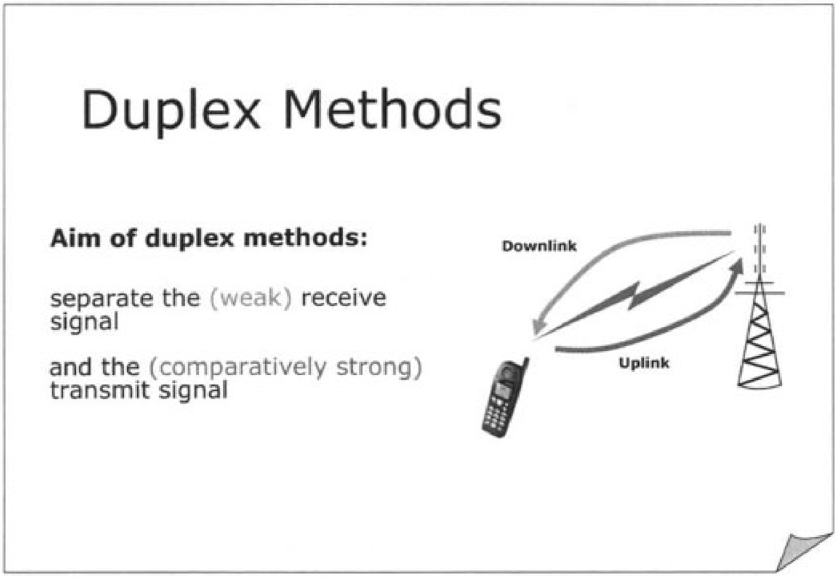


Figure 6.2: Duplex procedures

this type of spectrum separation is used in almost none of the commercially operated mobile radio systems today.

A paired frequency band is required for the implementation of an FDD system. Paired means that for each frequency block in the transmitting band a frequency block of the same size must exist in the receiving band, which generally has the same bandwidth. It is obviously more difficult to make two frequency ranges of the same bandwidth available in the spectrum than unpaired ones.

GSM is a typical representative of FDD systems.

6.4 The time-division duplex technique

Time Division Duplex (TDD) is another duplex technique used in mobile radio and particularly in UMTS. With this procedure the mobile and base stations take turns transmitting and receiving, as illustrated in Figure 6.4. The transmitting and receiving times then periodically alternate. The TDD technique thus separates the transmit and the receive signal in the time domain.

The point at which the switch is made from transmitting to receiving within a period is called the *switching point*. In principle, the switching point can be

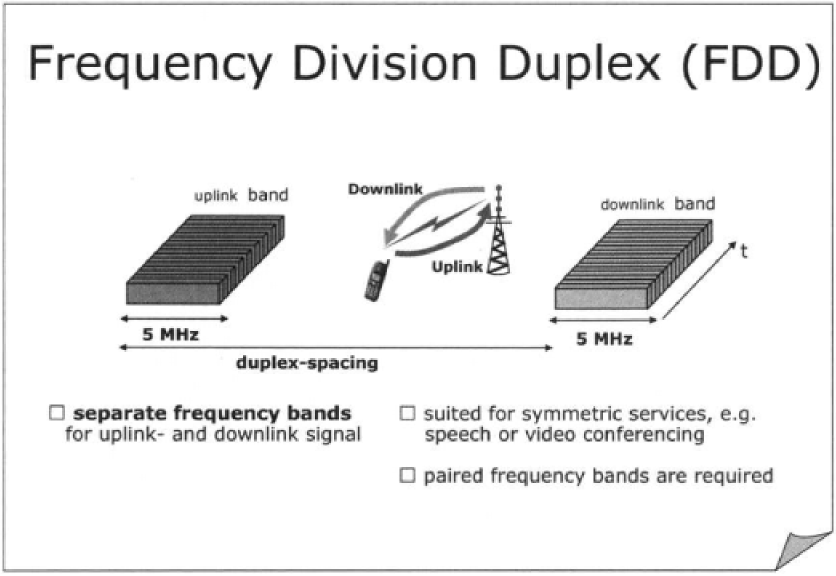


Figure 6.3: Frequency-division duplex

selected arbitrarily within a period. It is also possible for the uplink and the downlink to be switched back and forth several times during a period.

Through the possibility of varying the distribution of transmission capacity between uplink and downlink, the time-division duplex technique is suitable for the efficient mapping of asymmetric data traffic. In contrast to the FDD technique, all that is then needed in the frequency spectrum is a frequency block. It is obviously easier to find individual frequency blocks in the spectrum than paired ones. However, the potentially different signal runtimes with TDD have to be taken into account to prevent the transmitter signals of different stations overlapping at the receiver. Therefore, the time available for a transmission direction cannot usually be totally used for data transmission due to the need for guard times. These times can also be necessary for transferring the transceiver from a transmitting state to the receiving state and vice versa. A DECT system is a typical representative of a TDD system.

6.5 Multiple-access procedures

To repeat: through the purchase of a licence, a mobile radio network operator is allocated one or more frequency bands in which it can carry out its data transmission. The duplex technique of the technology used maps the com-

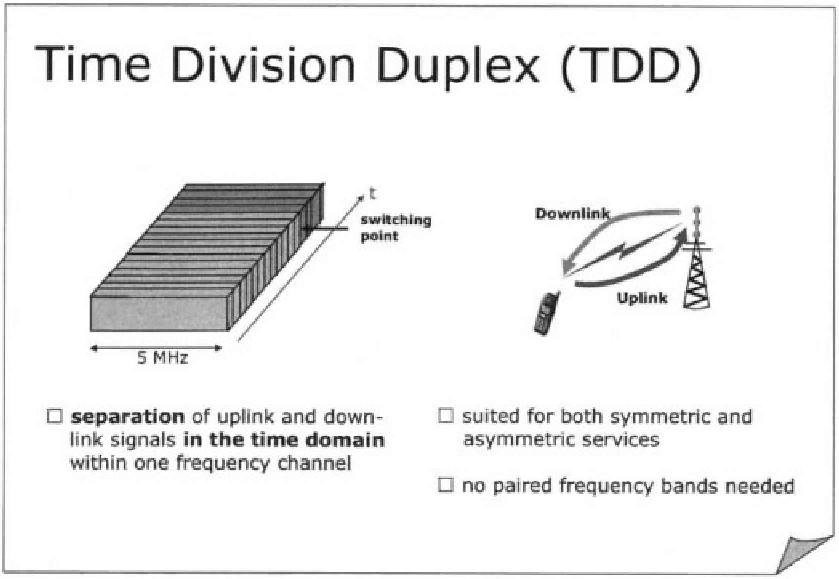


Figure 6.4: Time duplex

munication of the uplink and the downlink to these frequency bands, thereby separating the transmitting and receiving signals of the communicating stations.

Within a transmission direction a multiple access technique distributes the available transmission bandwidth among the individual users or connections. The multiple access technique defines the so-called physical channels. A physical channel is characterised by different physical parameters, e.g., by a period of time or a frequency range. Two stations communicate over a physical channel. These physical channels should be defined so that various connections do not cause mutual interference. The multiple access technique should therefore implement an efficient, flexible and equitable distribution of the total throughput of a transmission direction among many users.

A multiple access technique only divides up the available transmission capacity among physical channels. The theoretical total capacity remains unaffected by the multiple access technique (see Figure 6.5). The technique itself does not create capacity!

The three most common techniques for the separation of user signals of the same transmission direction are shown in Figure 6.6. They define the physical channels in the frequency domain, in the time domain or through the use of bipolar orthogonal carrier signals, called codes.

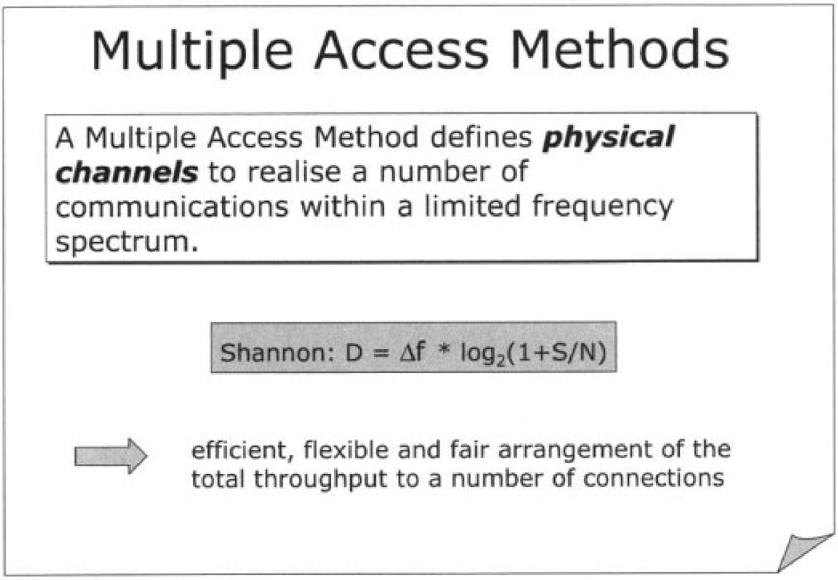


Figure 6.5: Multiple-access procedure

Frequency-division multiple access technique (FDMA) separates the user signals in a frequency range, i.e., it divides the frequency spectrum into frequency channels. A user can transmit or receive within a frequency channel. Since the bandwidth of a frequency channel is narrower than the overall bandwidth of a system, the transmission speed attainable in a channel is correspondingly lower than the overall transmission speed. Therefore, only a fraction of the overall capacity is available to each user.

Time Division Multiple Access (TDMA) is another technique that separates user signals. With TDMA, users do not transmit simultaneously in different frequency channels, as is the case with FDMA, but instead successively in the same frequency range. Users then use the entire frequency bandwidth of the system, and the transmission speed is correspondingly high. However, transmission in time segments called time slots has the effect that the effective transmission rate of a user is less than that of the overall transmission speed. Thus here too each user is only provided with a fraction of the overall capacity.

Mathematically, a multiple access technique is always based on the use of orthogonal carrier functions by which the signal of the individual participants is multiplied in order to orthogonalise the user signals, i.e., separate them from one another. With FDMA, these carrier functions are sine or cosine waves of different frequencies. In the case of TDMA, these are window functions that activate and deactivate the transmitter. It is easy to see that these window

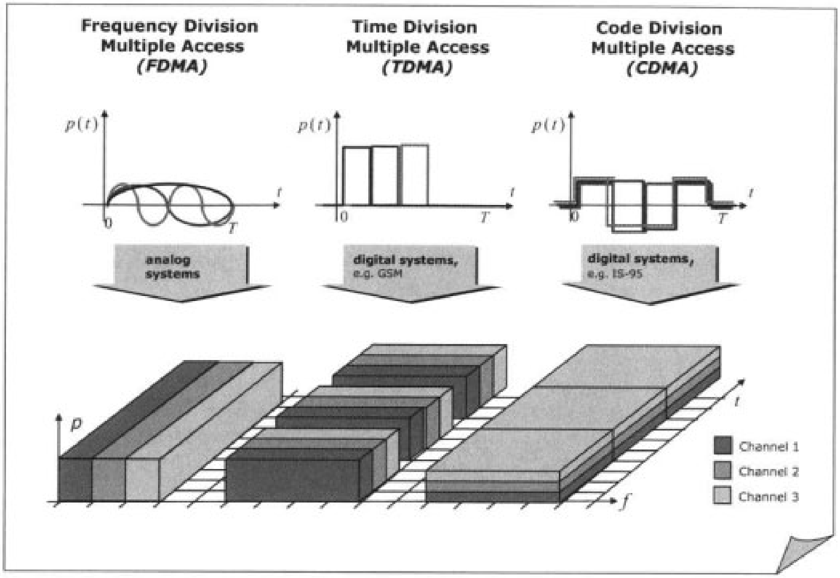


Figure 6.6: Frequency division, time division and code division multiple-access

functions are orthogonal: two periodic functions are orthogonal when the integral of the product of two functions produces a zero over a period of time. If two window functions are considered, then their product already equals zero and consequently the integral also. Thus the functions are orthogonal. It is easy to imagine that other two-valued carrier functions could also be orthogonal to one another.

Code Division Multiple Access (CDMA) is a technique that most often uses two-valued carrier functions to separate user signals. The carrier functions are bipolar sequences, called code sequences. These code sequences should be orthogonal to one another, i.e., the integral through the products of two sequences over a period of time should be zero. If such carrier functions are used, then the individual user signals can be multiplied by these sequences and transmitted in the same frequency range simultaneously. As will be explained later, the received signals in the receiver can be correlated with the code sequence, thus resulting in the recovery of the user signals.

The energy necessary for the representation and transmission of a bit is independent of the multiple access technique. In FDMA systems users transmit in a frequency channel with no time limitation. The energy is thus distributed over a relatively long period of time and the power is correspondingly low. In a TDMA system the same energy is transmitted within a time slot and the relevant power is high compared to an FDMA system. However, because a TDMA user has the entire frequency bandwidth available for transmission but the FDMA user only has a frequency channel, the spectral power density of a user signal is theoretically the same in both cases.

In a CDMA system users transmit simultaneously in the entire frequency bandwidth. Due to the long transmitting duration the transmitter power is correspondingly low; due to the large frequency bandwidth the spectral power density is also low. If the resources frequency, time and power density were applied to a diagram, one would notice that FDMA users share the frequency, TDMA users share the time and CDMA users share the power density. The respective volume of the blocks covered by the dimensions frequency, time and power density is equal, i.e., the individual multiple access techniques only share the overall transmission bandwidth available.

6.6 Direct-sequence CDMA

The most common technique in commercial mobile radio systems and used in UMTS for the implementation of code-division multiple access is called *Direct Sequence* (DS)-CDMA. With this technique the bipolar user data bit stream is multiplied by a user-specific bipolar code sequence (see Figure 6.7). The elements of the code sequence are called chips to differentiate them semantically from the bits of the user data stream. Chips are basically nothing other than bits! The multiplication of the bit stream by the chip stream in turn generates another bipolar data stream. Generally the rate of the chip stream is a multiple of the rate of the bit stream. A data stream generated through in-phase multiplication therefore has the rate of the chip stream. It is said that the multiplication of the bit stream by the code sequence will generate yet another chip stream.

If the generated chip stream is transmitted, a bandwidth is required that is larger than the bandwidth that would be necessary to transmit the user data bit stream. All users apply this procedure in order to leave the stamp of a fingerprint on their user data with the user-specific code that allows the transmitted signal to be reconstructed from the aggregate of received signals. As explained above, users in a CDMA system transmit simultaneously in the same frequency band. As a result, the signal-to-interference ratio at the receiver can easily be less than one since the power of the signal of a user is typically lower than the aggregate of the power of the signals of other users.

Figure 6.8 explains the direct sequence CDMA technique: a user would like to transmit the two bits 1 and 0. The diagram shows the bipolar representation of this bit sequence. The code sequence 10110100, which has eight times the transmission rate of the bit stream and is represented in its bipolar form in the diagram, is used to generate the chip stream. Consequently, the length of the code sequence is exactly equal to the duration of a bit. Each bit is multiplied

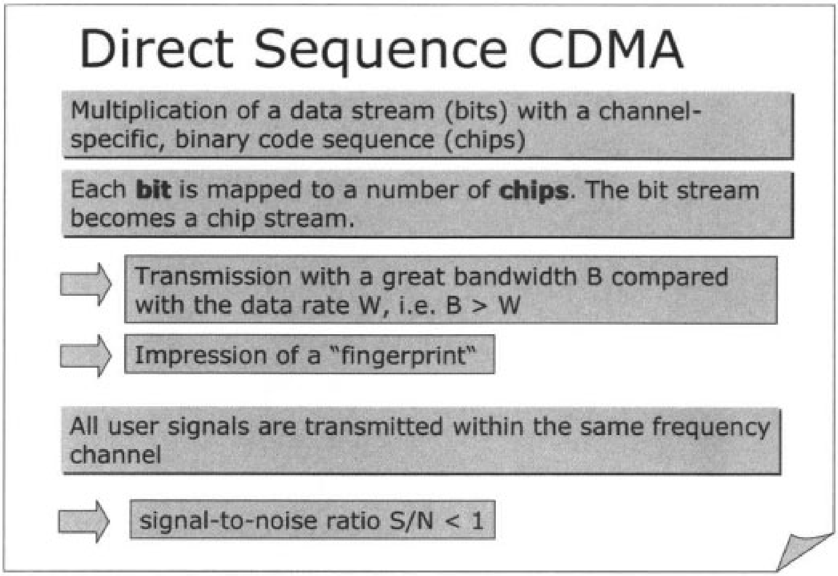


Figure 6.7: Direct-sequence CDMA

in-phase by the code sequence to generate the chip stream 1011010001001011. What is obvious is that the multiplication of the two bits has generated 16 chips that can now be transmitted over the mobile radio channel through an appropriate digital modulation procedure.

The bits of the data stream can be recovered in the receiver from the chip sequence received through a repetition of the multiplication procedure. The chip stream with the same code sequence that was already used in the transmitter is multiplied in-phase again producing the transmitted bit sequence 10 (see Figure 6.8).

6.7 Spectral characteristics of CDMA signals

The following can be established when one considers the power density spectra of signals. As illustrated in Figure 6.9, the bit stream has the rate R_b that characterises the width of the power density spectrum. A multiplication by the chip sequence again produces a chip sequence of the rate R_c, which is a multiple of the bit rate. The power density spectrum of the generated chip stream has the same form as that of the bit stream, except that it is wider by the same factor as the chip rate is higher than the bit rate. One talks about the power density spectrum of the signal being spread; this means that

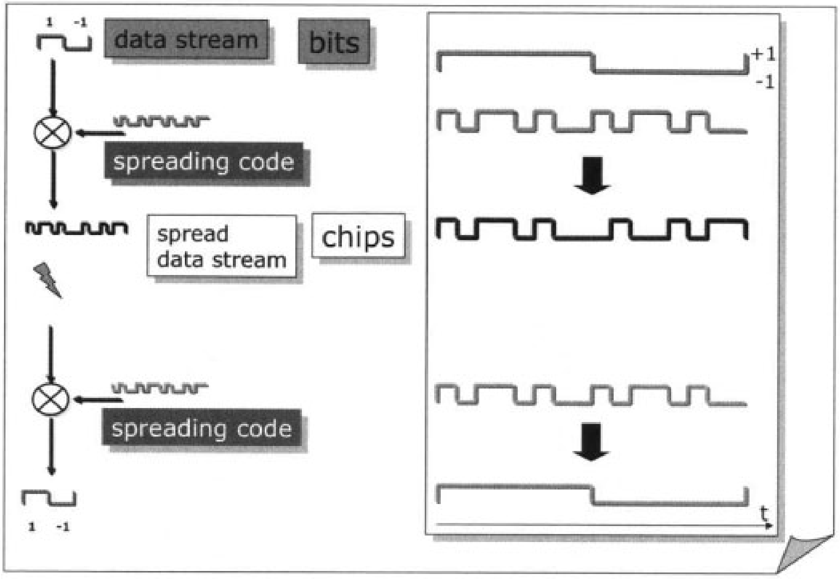


Figure 6.8: Mapping a bit stream to a chip stream and bit stream recovery

the spectrum spreads through multiplication by the bit sequence against the spectrum without multiplication. Therefore, one also talks about a ***spread spectrum*** technique in conjunction with CDMA.

The factor by which the spectrum spreads is called the spreading factor. It is calculated from the ratio of the bandwidths or the transmission rates of the bit and created chip stream. Since a bit is multiplied by the number N of chips, the chip rate is larger than the bit rate by exactly the factor N. The number of chips per bit therefore exactly corresponds to the spreading factor.

The code sequence that is multiplied by each bit is called a spreading code.

6.8 Reception of CDMA signals

So how is it possible to reconstruct a user's signal from the sum of all user signals received? The easiest way is to use the correlation filter receiver shown in Figure 6.10. As explained above, the chip stream received in the receiver has to be multiplied again by the spreading code so that the bit stream can be reconstructed. If the receiver receives the sum of several user signals, then this sum is multiplied by the spreading code of the respective user.

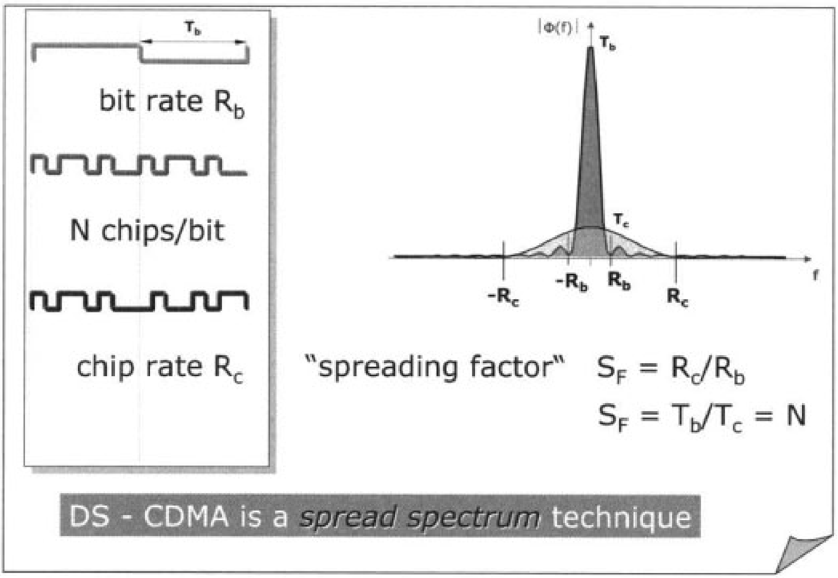


Figure 6.9: Spectral characteristics of CDMA signals

The influence of all other user signals has to be eliminated. Consequently, the product generated from the receive signal and spreading code is periodically integrated for the duration of a bit and the originating signal is sampled at the end of a period. In terms of the sampling time at the end of the duration of a bit, this architecture with multiplier and integrator represents a matched filter.

A matched filter forms the correlation function of the received signal with a pattern function, here the code sequence. The value of the correlation function is a measurement of the similarity of the received signal to the pattern function. Figure 6.10 shows that the value of the correlation function is obtained over a period through the integration of the product consisting of signal and pattern function. This mathematical operation was mentioned earlier in the introduction to multiple access techniques. There it was used to establish whether two carrier functions are orthogonal to one another. They are orthogonal when the integral is zero.

If the received signal corresponds to the pattern function, then the *Autocorrelation function* (ACF) of the pattern function is obtained at the output of a matched filter. The autocorrelation function reaches its maximum when the phase-shift between the received signal and the pattern function equals zero. If the received signal does not correspond to the pattern function, then the *Cross-correlation function* (CCF) of the pattern function appears with

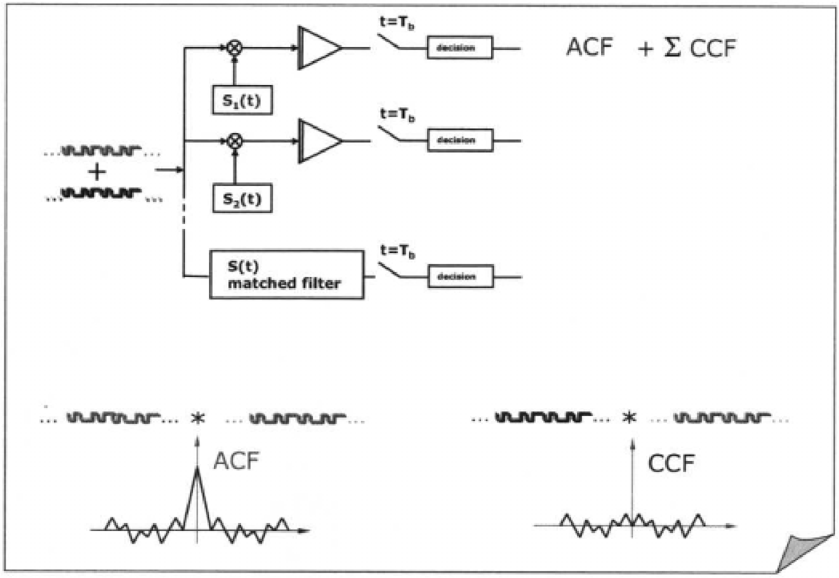


Figure 6.10: Correlation reception of CDMA signals

the received signal at the output of the matched filter. With phase-correct multiplication the integration exactly corresponds to the orthogonality test between received signal and pattern function mentioned above, in this case the spreading code.

The receiver receives the sum of all spread user signals. Consequently, the sum, which consists of the autocorrelation of the spreading code of a user and the cross-correlations of the spreading codes of all other users, is obtained along with the pattern spreading code at the end of a period at the output of the branch of the correlation filter receiver shown in Figure 6.10.

It is easy to understand that individual user signals can be received without mutual interference when the effects of cross-correlation disappear. Based on the above definition, this is the case when the spreading codes of the various users are orthogonal to one another. With CDMA, as with the other multiple access techniques presented, orthogonal carrier functions, in this case spreading codes, also ensure that the overall bandwidth can be distributed among individual users.

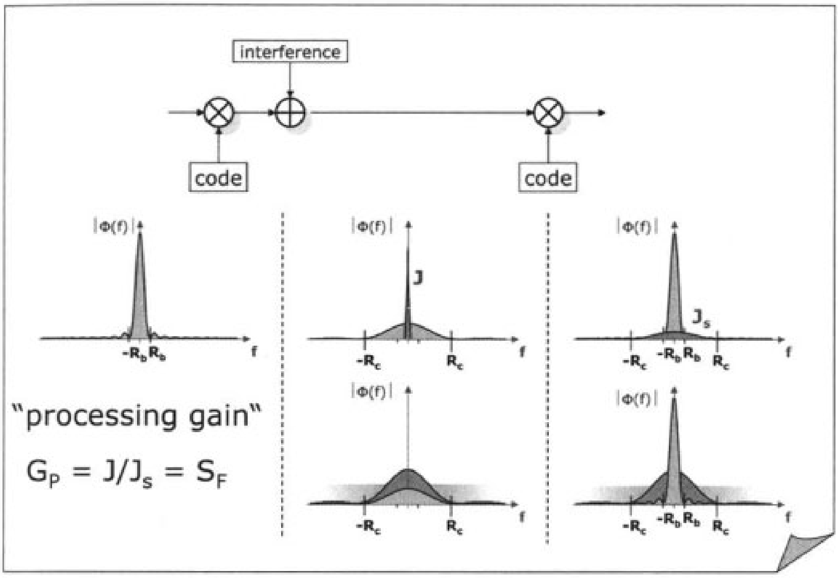


Figure 6.11: Processing gain and noisy transmission

6.9 Processing gain

The spectral spreading of user signals gives CDMA systems a special resistance to interference. Figure 6.11 shows a simplified DS-CDMA transmission system in which the spread signals experience interference when transmitted over the radio channel. Let us first look at narrowband interference of the power density spectrum which is illustrated in Figure 6.11. In this sense, narrowband means that the bandwidth of the interference signal is considerably narrower than the bandwidth of the spread user signal.

In the receiver the sum of user and interference signal is multiplied by the spreading code of the user. This enables a reconstruction of the original bit sequence of the user. The bandwidth of the generated bit signal is narrower than the spread signal by the amount of the spreading factor. Therefore, one also talks about a signal being de-spread. With respect to the interference signal, however, the multiplication by a code sequence turns out to be similar to the spreading procedure that the user signal passed through in the transmitter. The interference signal is thus spread in the receiver. The power density of the interference signal ideally decreases by the amount of the spreading factor, whereas the power density of the user signal increases by the amount of the spreading factor. CDMA systems are therefore resistant to narrowband interference signals and are thus often used in military technology. This is where the CDMA technique originated.

With a broadband interferer, e.g., other user signals, the broadband signal remains a broadband signal even after multiplication by the spreading code. The code sequence of an interference-producing user is multiplied by a different spreading code. However, this produces yet another code sequence but the rate does not change. A CDMA receiver thus makes a broadband interferer out of a broadband interferer. However, the user signal is de-spread at the same time, thus increasing the power density of the carrier signal.

The amount by which the power density of the carrier signal is increased in the receiver is called *processing gain*. The value of the processing gain corresponds to the spreading factor and therefore the number of transmitted chips per bit. The narrowband interferer is attenuated in the receiver by the amount of the processing gain.

White noise appears in the receiver as broadband interference. Broadband interference remains broadband. As shown in Figure 6.11, with a sufficiently large spreading factor, it is possible to transmit the user signal with lower spectral power density than that of thermal noise and to hide the user signal in the thermal noise so that it is only detectable with knowledge of the spreading code. This also explains why CDMA is used in military technology.

In the case of correlation filter reception, the ratio of the energy allotted to a bit during signal transmission to the noise power density equals the signal-to-noise ratio multiplied by the ratio of user bandwidth and data rate (see Figure 6.12). This relationship is generally valid, even with systems that do not use CDMA. As described above, in CDMA systems the user signal is spread to a bandwidth that is larger by the spreading factor than would be necessary in transmission without spreading.

Bandpass systems require at least one frequency bandwidth that corresponds to the data rate in order to transmit data. Thus with correlation filter receivers the bit energy-to-noise power density ratio is equal to the product of signal-to-noise interval and processing gain, with the processing gain ideally equal to the spreading factor.

In CDMA systems the E_b/N_0 is thus typically larger than the S/N by the amount of the spreading factor. This is also why communication is possible in CDMA systems even though the S/N for a connection at the receiver is typically much smaller than the value one.

In UMTS the QPSK modulation scheme is used. Since this scheme takes two bits to form a modulation symbol the signal power with respect to one bit equals half of the signal power with respect to the modulation signal.

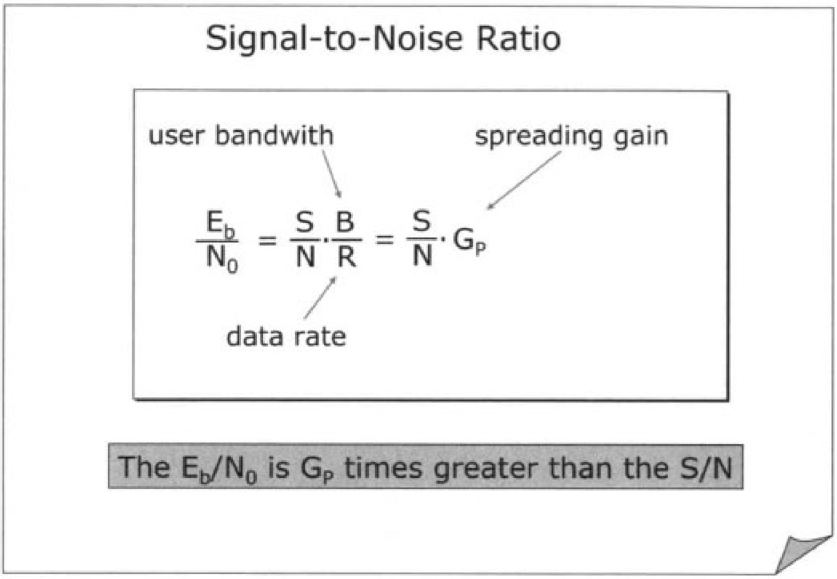


Figure 6.12: Effects of processing gain on signal-to-interference ratio

6.10 A CDMA transmission system

Let us conclude by considering the CDMA system with two channels shown in Figure 6.13. On the left is the transmitter that sends data on both channels; on the right is the receiver for the top channel. The bit sequence (A) is spread with the DS-CDMA technique, i.e., chip sequence (B) results from chip sequence (A) through multiplication with the code sequence of the channel at the top. The rate of the chip sequence is a multiple of the original bit sequence. The same applies to the bit sequence that is transmitted over the second channel. Both chip sequences are added and transmitted together over a noisy channel.

The noisy sum signal (C) of both channels is thus at the receiver. For recovery of bit sequence (A) the received signal is multiplied again by the code sequence of the first channel and integrated or low-pass filtered (an integrator is also a low-pass filter). The low-pass filtered signal (E) now only has to be sampled at the right times so that the bit sequence originally sent can be reconstructed.

The assumption with this example is that the two spreading codes are orthogonal to one another, i.e., the proportion of the second channel disappears through multiplication of the sum signal (C) by the code sequence of the first channel and the subsequent low-pass filtering. A receiver for the second chan-

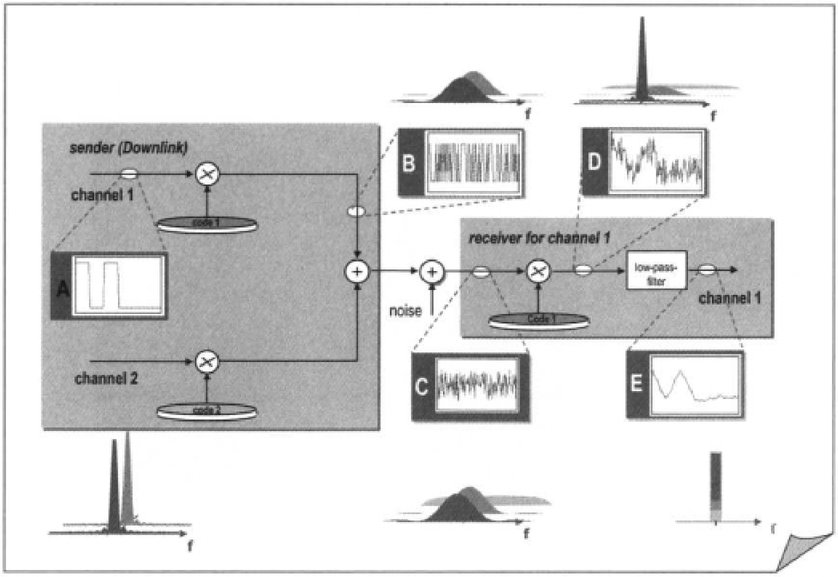


Figure 6.13: An example of a CDMA system on the downlink

nel would be structured the same way but would use the second spreading code in the multiplication.

The power density spectrum of the overlapping signals indicated in Figure 6.13 shows that the transmitter uses the spreading process to produce broadband signals from the original narrowband signals. Both signals are transmitted together at the same time in the same frequency band; the S/N at the receiver input is therefore smaller or equals one. Due to de-spreading the power of the signal of the first channel is again concentrated on a narrower bandwidth; the power density of the de-spread signal within this narrower bandwidth is higher by the spreading factor than the power density in a spread situation. The signal part of the second code channel appears as a broadband interferer to the receiver for the first code channel. Broadband interferers remain broadband in character so that the ratio E_b/N_0 is finally greater by the spreading factor than the S/N at the receiver input.

6.11 Spreading codes

As already explained, spreading codes have the property that they are orthogonal to one another, i.e., their cross-correlation disappears. In some cases it is sufficient if the cross-correlation does not disappear but is very low so that the different user signals interfere very little with one another. In this case the codes are called quasi-orthogonal codes.

The codes most suitable for detection, synchronisation and channel evaluation are those that have a distinctive impulse form in the zero point of their autocorrelation function and are low in all other places. A set of codes, the elements of which are all orthogonal or quasi-orthogonal to one another, is also called a code family. Large code families, i.e., families with a large number of spreading codes, are suitable for mobile radio use.

When the chip rate on the radio transmission link is constant, the bit rate of a user signal represented in the chip sequence is only dependent on the spreading factor, since the chip rate is higher by the spreading factor than the bit rate. At the same time the spreading factor corresponds to the number of chips per bit. If the same code sequence is used to spread each user bit, it then makes sense to use code families that contain code sequences of different lengths to enable the implementation of variable data rates. An example of this can be seen in Figure 6.15 (see Section 6.12).

UMTS uses the DS-CDMA technique in which a distinction is made between spreading individual bit streams and scrambling the sum signal of a station. Different bit streams that are to be transmitted simultaneously from the transmitter are multiplied by different, orthogonal spreading codes and then added together, shown already as a general procedure for CDMA systems.

The aggregate signal is then scrambled, which occurs through a chip-by-chip multiplication of the aggregate signals by a scrambling code (see Figure 6.14). The scrambling code has the same rate as the spread chip streams of the individual channels. Therefore, the scrambling does not cause the user signal to be spread again, and the chip rate remains the same. Instead the preliminary sign of the aggregate signal is systematically changed at certain places through the scrambling.

Scrambling codes are basically transmitter-specific. Through scrambling the chip sequences of different transmitters lose their orthogonality to one another and become only quasi-orthogonal. Although this is accompanied by a talkover of the various code channels, the signals of the different transmitters do not have to be synchronised simultaneously or arrive chip-synchronous in the receiver. The quasi-orthogonal signals remain quasi-orthogonal even with time shift.

In the case of asynchronous reception, orthogonal codes lose their orthogonality. A CDMA receiver therefore uses orthogonal codes to compensate for the runtime-associated delay of the individual user signals. The amount of decoding required increases with the rise in the number of user signals.

As a result of scrambling, in UMTS the entire family of orthogonal codes is available to each transmitter that uses its own scrambling code. Two different transmitters can use the same spreading codes, because the different

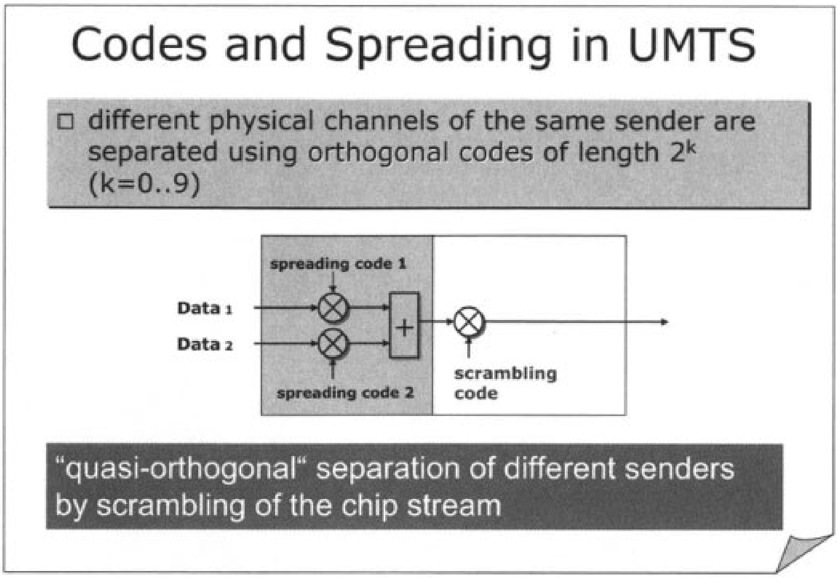


Figure 6.14: Spreading and scrambling

scrambling codes ensure that the signals on the radio transmission link are quasi-orthogonal. Since the quantity of orthogonal codes is basically limited, a system would lose its flexibility if the spreading codes used in a transmitter could not be used simultaneously in another station.

In FDD mode of UMTS the scrambling code is station-specific, i.e., each base station and each mobile station uses a different scrambling code. In TDD mode the scrambling code is cell-specific, i.e., mobile stations in the same cell use the same scrambling code as the base station. Different base stations use different scrambling codes. The data streams in a cell are thus transmitted with the same scrambling code on the downlink and are therefore orthogonal.

This difference essentially has to do with the fact that in TDD mode only up to sixteen mobile stations per cell can transmit at the same time, but a considerably higher number is possible in FDD mode (see Section 6.14).

6.12 Orthogonal spreading codes in UMTS

As explained earlier, the code families that are particularly appropriate for mobile radio are those that consist of orthogonal code sequences of varying lengths. The orthogonal codes with a variable spreading factor (*Orthogonal*

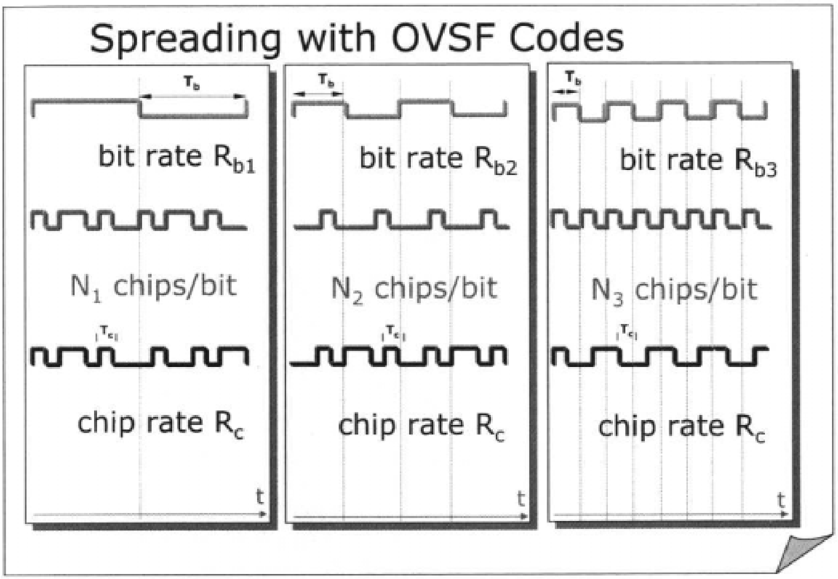


Figure 6.15: Implementation of variable bit rates using OVSF codes

Variable Spreading Factor (OVSF) codes) used in UMTS form such a family. This family will be discussed in detail later.

The principle covering the implementation of different transmission rates is illustrated in Figure 6.15. Three data streams are transmitted simultaneously in three different code channels. At the same time the bit rates of the data streams are different. The bit streams are spread so that the resulting chip streams have the same rate. This involves the multiplication of each bit of the first data stream by a spreading code of eight chips. The transmission rate of the resulting chip stream is therefore eight times that of the bit stream. The same thing happens with the second and third data streams, the only difference being that each bit is multiplied by a spreading code of four or two chips, respectively. The chip rate is then four times or twice as great as the corresponding bit rate.

The receiver that receives the sum of all chip streams must be in a position to reconstruct through correlation the transmitted bit stream as shown. This is only possible if the code sequences of different chip streams are orthogonal to one another. Since the third code sequence is used exactly four times during the first code sequence, four copies of the third code sequence have to be orthogonal to the code sequence of the first channel. The same applies to two copies of the second code sequence. In the end two copies of the third code sequence must be orthogonal to a code sequence of the second code channel.

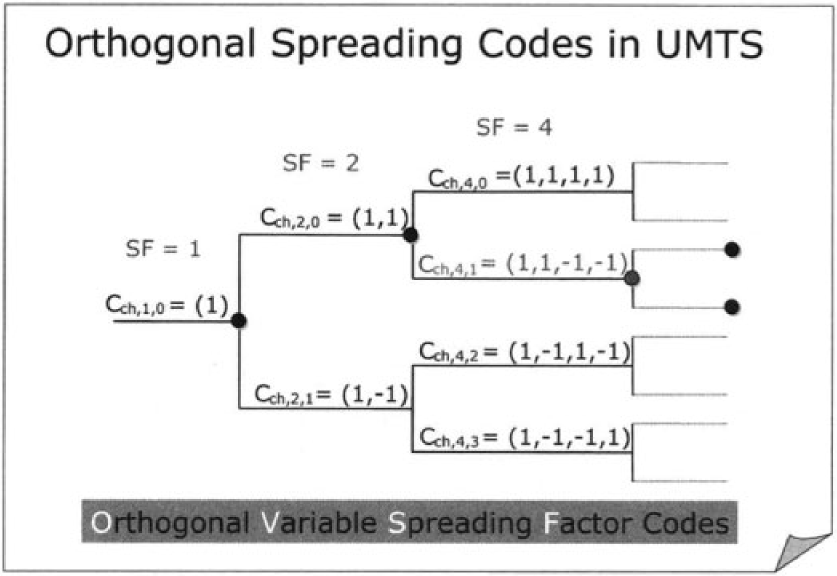


Figure 6.16: OVSF code tree

Codes that meet the conditions described and are used in UMTS are the ones with a variable spreading factor (*Orthogonal Variable Spreading Factor* (OVSF) codes). As shown in Figure 6.16, these codes can be created through the use of a code tree. Each node of the tree exactly has two branches, each representing a double-length code. The codes of a level (vertical) have the same length N and thus the same spreading factor. Each code with a spreading factor N is created from a code with the spreading factor $N/2$. Consequently, a set of 2^k spreading codes with a length of 2^k chips are available at the k-th level. For example, there are four codes with the spreading factor four and eight codes with the spreading factor eight.

A code is basically created through the multiplication of a code of the next lower level of a code tree. The code being multiplied is called the mother code. Exactly two double-length codes are created from a mother code through the chaining of two copies of the mother code or the chaining of the mother code with a copy multiplied by -1.

Codes of different levels are only orthogonal if the shorter one is not found again in the longer one, which, based on the construction rule described above, can happen. This means that two codes of different levels of the code tree are orthogonal to one another as long as one of the two codes is not the mother code of the other one. Because of this limitation the number of simultaneously usable codes depends on the bit rate and spreading factor.

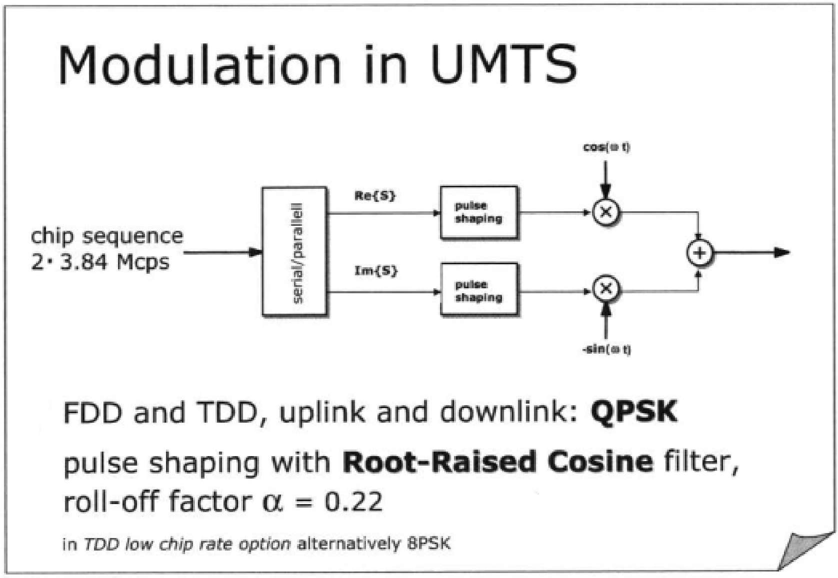


Figure 6.17: Modulation in UMTS

If a connection uses the spreading factor one, then there is no other code that does not have this code as the mother code. In this case, the connection has exclusive use of the channel. However, if, for example, a connection uses the spreading factor two, another participant can use the spreading factor two or two other connections can use the spreading factor four.

6.13 Modulation in UMTS

UMTS uses the modulation technique *Quaternary Phase Shift Keying* (QPSK) in which two chips are transmitted in each symbol (see Chapter 1). QPSK is used in FDD and in TDD mode in both transmission directions. For the low chip rate option of TDD an 8-ary *Phase Shift Keying* (PSK) can be used alternatively, i.e. 3 bits are transmitted within one modulation symbol. On the uplink in FDD mode the chips of different physical channels are either mapped to the first or the second chip of the modulation symbol. Consequently, use of a *Binary Phase Shift Keying* (BPSK) modulation technique on the uplink in FDD mode is often mentioned. However, this only applies to one single physical channel.

The modulation rate is 3.84 Mchip/s, with two chips being transmitted per modulation step due to the four-value modulation technique (see Figure 6.17).

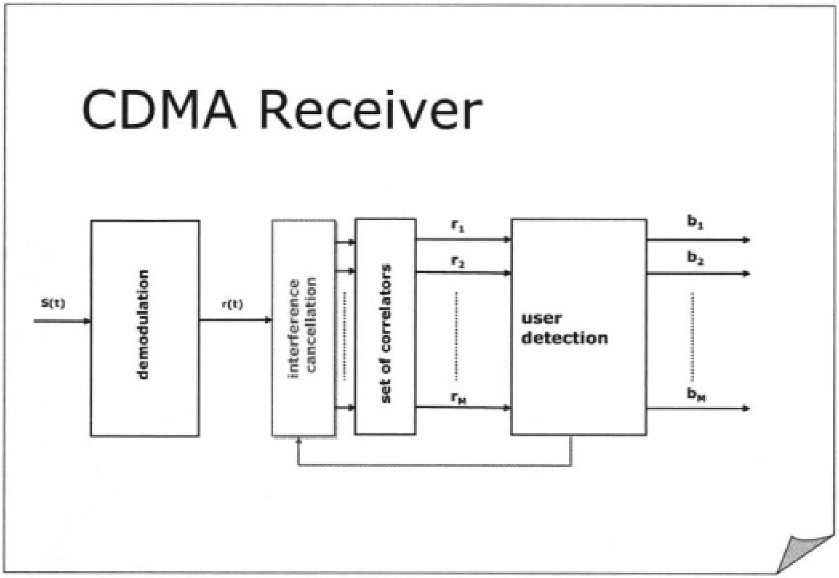


Figure 6.18: Structure of a CDMA receiver

The gross chip rate is therefore double the modulation rate. For the low chip rate option of TDD the modulation rate is 1.28 Mchip/s. A root-raised cosine is used for pulse forming.

6.14 CDMA receivers

The task of a CDMA receiver is to reconstruct the individual bit streams from the sum of individual spread data streams. There are basically three detection types: single detection, joint detection and interference cancellation.

As explained in Section 6.8, a conventional single detector consists of a bank of correlators. One correlator is required for each code channel. In the correlator bank the received signal is correlated with each specific carrier signal in separate detection branches (see Figure 6.10). This correlation receiver can equally be implemented with matched filters. The output signals of the individual branches are sampled. Based on the preliminary signs of the sampling values, the subsequent decision levels identify the detected bit values. Each branch of the receiver thus represents a detector for a particular user; this detector ignores the presence of other users.

No multiple-access interference exists as long as the spreading codes of the individual chip streams are orthogonal to one another. However, in most

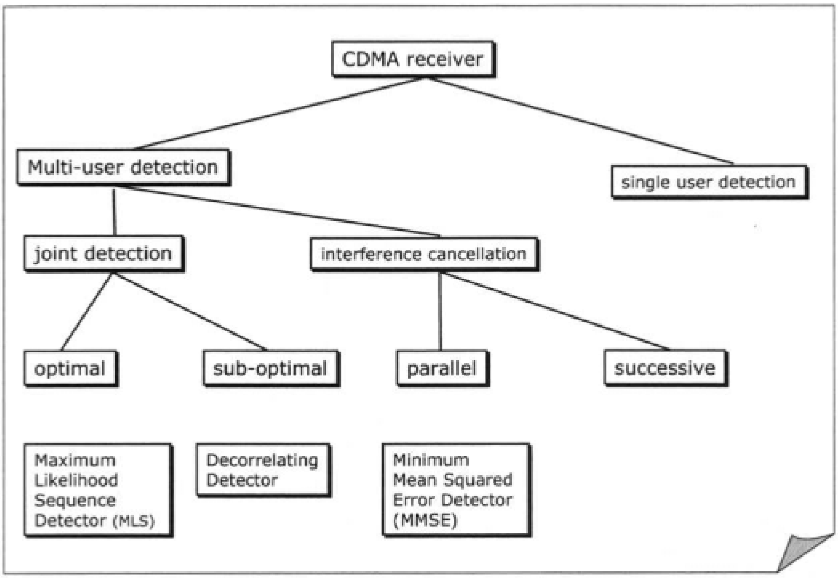


Figure 6.19: Classification of CDMA receivers

cases transmission is asynchronous and, moreover, the spreading codes are only quasi-orthogonal. Consequently, multiple-access interference caused by channel talkover has a considerable bearing on the capacity and the efficiency of CDMA systems. If the number of users rises, the noisy contribution of multiple-access interference will also increase.

In the case of reception with joint detection, the data of all users is detected in one single step. All specific code sequences in the receiver have to be known *a priori*. Depending on the detection algorithm, other parameters, such as signal energy or amplitude and delay time, must also be known. The basic principle covering receivers with joint detection is shown in Figure 6.18. A joint detector follows a bank of correlators. This joint detector applies the respective detection algorithm to the sampled output signals of the correlators and determines the estimated values of the data bits. Joint user detection thus uses knowledge about other user signals being received at the same time in order to suppress multiple-access interference.

According to the principle of interference cancellation, an estimated value is produced from each user's contribution to multiple-access interference so that it can be subtracted from the received signal and thus reduce the overall interference. This can either take place step-by-step in several detection stages switched one after the other or be carried out for all users simultaneously. The interference cancellation approach is not optimal, because the effects not yet

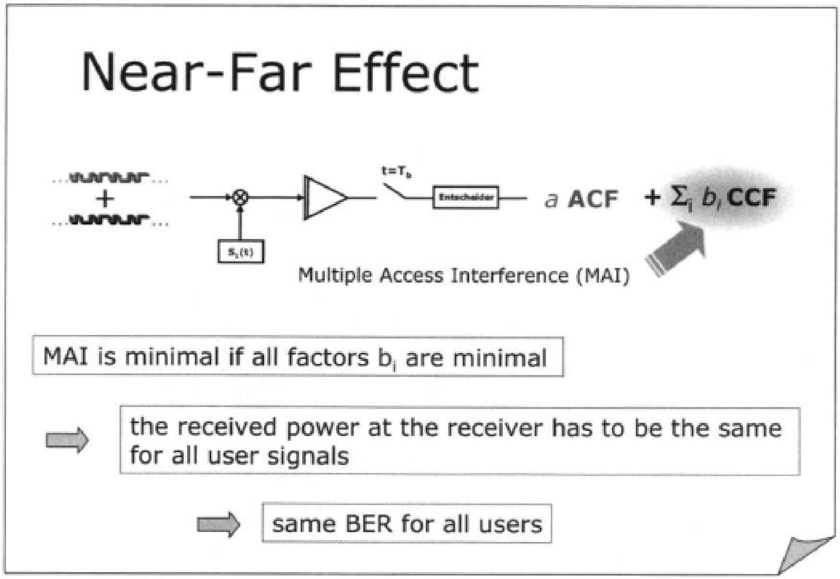


Figure 6.20: The near-far effect

cancelled when some of the transmitted data symbols are detected are treated as noise.

In summary, CDMA receivers can be classified as follows. If the presence of other code channels is ignored during the decoding of a code channel, this is referred to as single detection; otherwise it is called multi-user detection. Single detection is currently used in the FDD mode of UMTS, because multi-user detection would require a considerable amount of computing due to the length of the scrambling codes, the asynchronicity and the large number of simultaneously active users. Multi-user detection can be used in TDD mode due to the shorter scrambling codes and low number of simultaneously active users.

6.15 The near-far effect

The near-far effect occurs in mobile radio systems that use single detection. As described earlier, the value of the autocorrelation function of the code sequence and the sum of the cross-correlation functions are sampled with all other code sequences at the input to the correlation filter receiver. In ideal circumstances, thus with orthogonal codes and synchronous transmission, this part, the multiple-access interference, equals zero. Generally, however, the

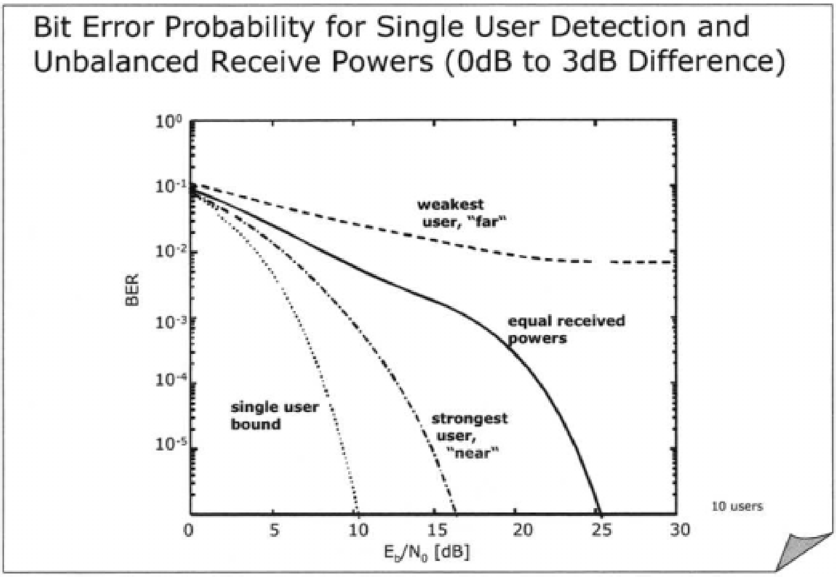


Figure 6.21: The near-far effect: effects on bit-error ratio

noisy contribution of multiple-access interference increases as the number of users rises. This negative influence is clearly increased when the received signal strength of other users is considerably higher than the signal strength of those users being considered.

In Figure 6.20 the different received signal strengths are indicated by the weighting factors a and b. If all code channels in a system are considered, it can be determined that the a of one is the b of the other. As a result, if the cross-correlation is the same, the multiple-access interference for all participants is equal. In other words, the multiple-access interference is distributed fairly when the received signal strength of all user signals at the receiver is equal. This can only be guaranteed through control of the transmitter power.

When mobile stations have the same transmitter power, the signal of one of the users close to the base station is received considerably stronger than the signal of a user who is further away. Hence, the bit error ratio of the remote user is much greater than that of the closer user. This is called the near-far effect. It only occurs if channel talkover also exists between code channels - for example, through a lack of synchronisation or only quasi-orthogonal codes. Multiple-user detectors can eliminate multiple-access interference or reduce it and in ideal circumstances are resistant to the near-far effect.

Figure 6.21 shows the significant impact the near-far effect can have on system performance. Here the frequency of detection errors on the uplink is shown as a function of the bit energy-to-noise power density ratio for a cell with ten users. The solid line represents the bit error ratio for a situation in which the received signal strength of all user signals is exactly the same.

An extreme difference in the bit error ratios for near and far users becomes evident when the assumption is made that the received signal strength of user signals can differ up to 3 dB, i.e., the weakest signal has exactly half the power of the strongest signal. The bit error ratio for the far user can be magnitudes greater than that of the near user.

With the bit error ratio shown for a weak user in the example, communication is basically no longer possible. CDMA systems with a single detector therefore absolutely require power control for a fast and precise adaptation of the received signal strength. This is the reason why CDMA systems often incorporate their own physical channels that mainly transmit power control commands at a high rate.

FDD mode in UMTS uses single detectors. The transmission rate for the power control commands is at 1500 commands per second.

6.16 Questions

6.1 Explain the difference between a duplex and a multiple-access technique.

6.2 What is an asymmetric service and why is it easier to implement asymmetric services with a TDD duplex technique than with an FDD duplex technique?

6.3 What is the maximum transmission speed D that can be achieved in a bandwidth $B = 5\,\text{MHz}$ with a signal-to-noise ratio of $S/N = 15$? [Tip: $log_2(x) = \frac{log(x)}{log(2)}$]

6.4 Which multiple-access technique basically enables the highest capacity in a mobile radio system?

6.5 You want to distribute the total transmission rate D among 10 users. How is this done in a FDMA, a TDMA and a CDMA system? How should the parameters transmission rate D_T, bandwidth B_T and signal-to-noise ratio S/N_T be selected for each user and why?

6.6 What effect does the modulation scheme have on the transmission rate?

6.7 Why is it that the signal-to-noise ratio S/N at the receiver can be $S/N \leq 1$ in a CDMA system?

6.8 What is the difference between a *bit* and a *chip*?

6.9 Why is DS-CDMA a "spread spectrum" method?

6.10 What does "orthogonality" stand for in CDMA?

6.11 How is orthogonality ensured in TDMA and FDMA?

6.12 What is the difference between spreading and scrambling?

6.13 How are variable transmission rates achieved with a constant bandwidth?

6.14 A system has an overall throughput of 2 Mbit/s. Orthogonal Variable Spreading Factor (OVSF) codes are used. Two subscribers are using the spreading factor SF = 4. How many subscribers with SF = 16 can transmit data simultaneously? What is the transmission rate for the subscribers when a constant chip rate is assumed?

6.15 What is Multiple Access Interference (MAI)?

6.16 How does the transmitter power of users have to be controlled to keep multiple-access interference to a minimum?

6.17 What is the near-far effect?

7 The Physical Layer at the Radio Interface

7.1 The physical layer in the UTRA protocol stack

The physical layer of the UTRA protocol stack provides transport services for the *Medium Access Control* (MAC) layer above it (see Figure 7.1). These services are characterised by the way in which data is transmitted and the quality with which it is transmitted. The services of the physical layer are also called transport channels.

The physical layer maps the transport channels to the physical channels that, as will be shown later, are distinguished by parameters such as spreading codes. Data of different transport channels can be transmitted simultaneously over one physical channel. A physical channel that is used to transmit user data therefore cannot directly follow a transport channel. Each transport channel is allocated a transport format or a set of transport formats that define the way the transport channels are mapped to the physical channels. The transport format specifically defines the channel coding, the interleaving and the bit rate. The different possible transport formats are determined by the *Radio Resource Control* (RRC) layer (see Section 5.10).

The physical layer essentially is responsible for the following tasks: error protection and detection, measurement of transmission channel characteristics, reporting of measurement results to RRC layer, duplication and merging of data streams for soft handovers, mapping transport channels to physical channels, spreading and modulation as well as synchronisation and power control.

7.2 Mapping transport channels to physical channels

Figure 7.2 shows how transport channels are mapped to physical channels. Periodically one or more data packets, called transport blocks, are transmitted simultaneously over the transport channels. Each transport block passes through a string of algorithms in the physical layer for the implementation of error protection and matching the data to the transmission capacity of the physical channels.

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff
 ISBN: 0-470-84557-0

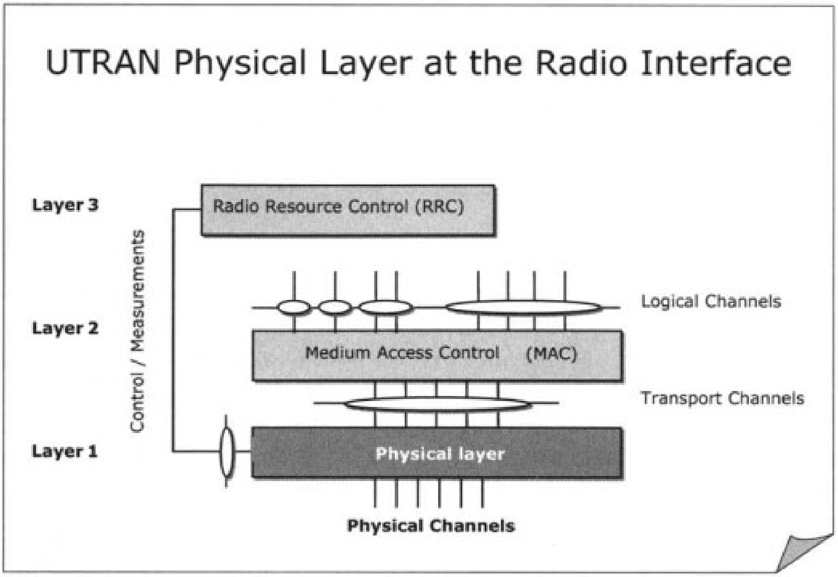


Figure 7.1: The physical layer in the UTRA protocol stack

Within the physical layer several transport channels, the transport blocks of which require the same coding, interleaving and puncturing within the physical layer, can be multiplexed to an internal channel of the physical layer, called the *Coded Composite Transport Channel* (CCTrCH). The data stream that is formed this way is then transmitted over one or over several physical channels simultaneously.

Only one CCTrCH is permitted on the uplink, i.e., the transport blocks of different transport channels are always transmitted at the same time and with the same channel coding, and so forth.

Compared to second-generation systems (e.g., GSM), an UTRAN enables very flexible and efficient data transmission of different data streams due to this concept of mapping transport channels to physical channels. Since the transmission capacity of the physical channels can be changed every 10 ms, e.g., through a change of the spreading factor, the transmission rate can be adapted to cope with bursty-type or fluctuating traffic. The transmission capacity of the radio channel is then used efficiently although intelligent algorithms are needed for allocating and changing transmission capacities.

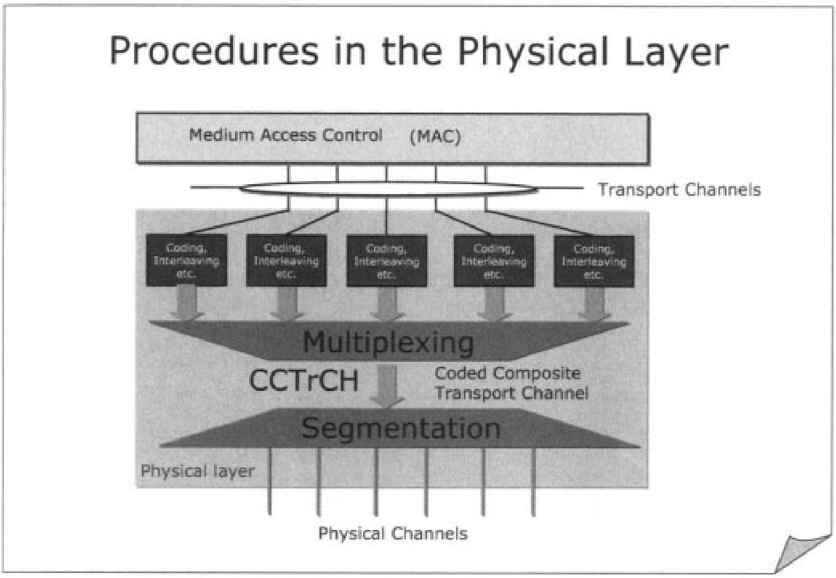


Figure 7.2: Mapping transport channels to physical channels

7.3 Multiple access in UMTS

Two operating modes have been standardised for UMTS, one for frequency duplex operations (FDD) and one for operation in an unpaired frequency band in time duplex (TDD). UMTS-FDD and UMTS-TDD differ not only in the duplex techniques they use but also in their multiple-access techniques. It can generally be said that the only place where UMTS-FDD and UMTS-TDD differ is in the physical layer of the UTRA protocol stack. All other protocols and system components are nearly the same.

Multiple access of both modes is based on the same time scheme, which is shown in Figure 7.3. For the realisation of periodic functions time is divided into time frames each of 10 ms duration. The duration of a frame corresponds to the duration of 38400 chips, the modulation rate is thus 3.84 Mchip/s. Due to the four-value QPSK modulation used, a chip carries the information of two chips - one for the in-phase branch of the modulation and one for the quadrature branch (see Chapter 1).

Each frame is divided into 15 time slots, each corresponding to the duration of 2560 chips. The chip duration is around 0.2604 μs. Optionally, the TDD mode can be operated at a lower chip rate of 1.28 Mchip/s. In this case the structure of a radio frame is different. A radio frame still has a length of 10 ms but is

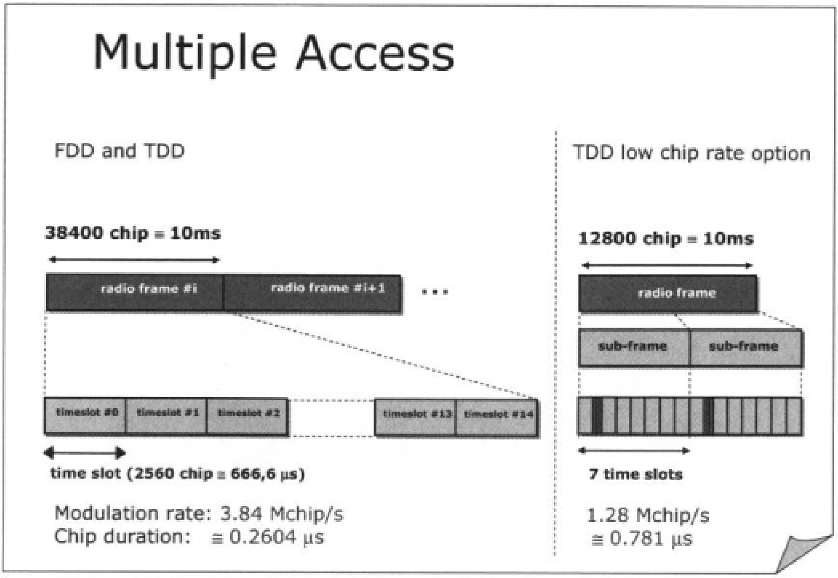


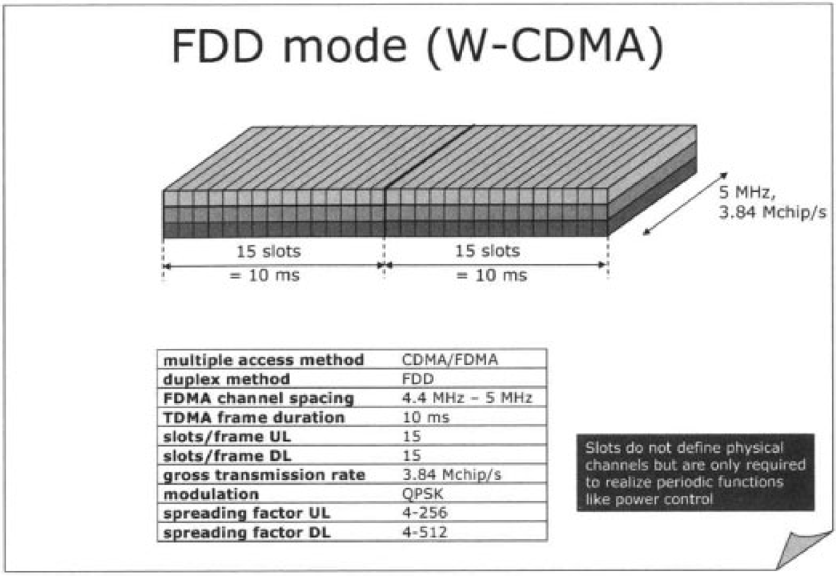
Figure 7.3: Time structure in multiple access of UMTS

subdivided into 2 sub-frames. Each sub-frame contains 7 normal timeslots and additionally 3 time slots for synchronisation purposes, see Section 7.3.3.

7.3.1 Multiple access in FDD mode

In FDD mode the CDMA technique is used as a multiple-access procedure. Individual user signals or physical channels are separated through the use of different spreading codes. As illustrated in Figure 7.4, different users transmit in FDD mode simultaneously in the same frequency band but using different spreading codes. The duration of a user signal is typically a multiple of a frame length; a user thus occupies all time slots of several sequential time frames. On the downlink spreading factors between 4 and 512 can be used; on the uplink, between 4 and 256.

The FDMA technique can be used at the same time as the CDMA technique if the network operator is permitted use of more than one frequency channel. In this case all spreading codes in each frequency channel are available once. The spacing between frequency channels can be varied between 4.4 MHz and 5 MHz in 200 kHz steps. This essentially exists for regulatory purposes so that systems of different operators working alongside each other in a spectrum are mutually protected from interference caused by neighbouring channel transmission of the other system.



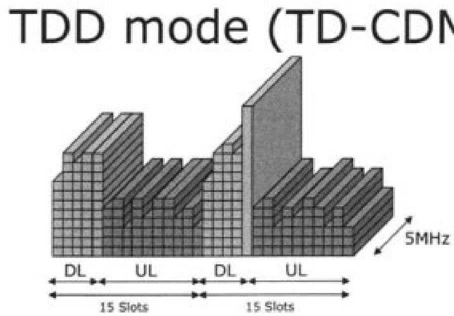
multiple access method	CDMA/FDMA
duplex method	FDD
FDMA channel spacing	4.4 MHz - 5 MHz
TDMA frame duration	10 ms
slots/frame UL	15
slots/frame DL	15
gross transmission rate	3.84 Mchip/s
modulation	QPSK
spreading factor UL	4-256
spreading factor DL	4-512

Figure 7.4: Multiple access in FDD mode

In FDD mode, therefore, the division of time frames into time slots is not designed to separate user signals but instead exclusively to implement periodic functions, such as power control. As explained in Section 6.15, a CDMA system requires fast and precise power control in order to prevent near-far effects. In FDD mode, therefore, a power control command is transmitted in each time slot.

7.3.2 Multiple access in TDD mode

In TDD mode the uplink and the downlink are implemented in the same frequency channel. Each time frame contains a minimum of one time slot for the uplink and at least one time slot for the downlink. An example of multiple access in TDD mode is shown in Figure 7.5. The time axis is represented vertically, in this case for the duration of two time frames, thus 20 ms. In each time frame the first five time slots are used for the downlink and the following ten time slots for the uplink. In contrast to FDD mode, in TDD mode a physical channel is not only characterised by the spreading code but also by a time slot. A cube in Figure 7.5 thus represents an occupied physical channel. Because of the CDMA component several physical channels can be implemented simultaneously through the use of suitable spreading codes in a time slot.



multiple access method	CDMA/FDMA/TDMA
duplex method	TDD
FDMA carrier spacing	4.4 MHz - 5 MHz
TDMA frame duration	10 ms
slots/frame UL	1-14 (of 15)
slots/frame DL	1-14 (of 15)
gross transmission rate	3.84 Mchip/s
modulation	QPSK
spreading factor UL	1-16
spreading factor DL	1 or 16

Figure 7.5: Multiple access in TDD mode

Whereas in FDD mode the entire duration of one or more time frames is available to a physical channel for data transmission, a physical channel in TDD mode only receives the duration of one time slot per time frame. Since the bit rate drops as the spreading factor increases, in TDD mode only spreading factors between 1 and 16 are used so that about the same transmission rates can be achieved for a physical channel as with FDD mode.

As is the case with FDD, TDD can also use an FDMA component when the network operator has more than one frequency channel available. The modulation type, modulation rate, frame duration and frequency channel distance are the same in FDD mode and in TDD mode.

Whereas in FDD mode the distribution of traffic capacity to the uplink and the downlink cannot be changed because of the fixed frequency channel bandwidths, in TDD mode the portion of the time frame available to the uplink and the downlink can be varied. This occurs through the selection of a switching point within a time frame.

A distinction is essentially made between four different types of configuration, shown in Figure 7.6. Arrows pointing upwards indicate a time slot that is reserved for the uplink; arrows pointing downwards indicate a time slot for the downlink. Basically it is possible to switch several times between the uplink and the downlink within a time slot; in an extreme situation, after each time slot. Also, the number of time slots configured within a frame

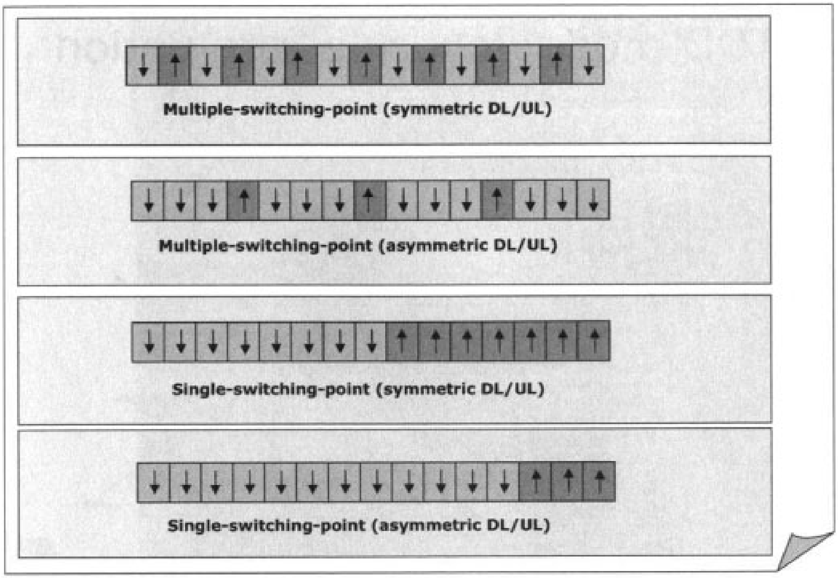


Figure 7.6: Different options for the selection of a switching point

does not have to be the same for both the uplink and the downlink. This enables an efficient implementation of asymmetrical services, for example. A configuration of several switching points within a time frame reduces the time between two transmission possibilities in one transmission direction. This can minimise the transmission delay since data packets for transmission have to wait for the next transmission possibility. Quick switching between uplink and downlink places high demands on the transmitting and receiving units of the base and mobile stations. It is therefore advisable to select only one switching point within a time frame if the transmission delay that occurs is tolerable, which is the case with most services in UMTS. However, even a configuration with one switching point offers the possibility of selecting a different capacity for the uplink and the downlink.

7.3.3 Multiple access in the TDD mode low chip rate option

In Release 4 of the UTRA standard a so-called *low chip rate option* with a modulation rate of 1.28 Mchip/s is specified for the TDD mode. As in the normal TDD mode the time frame has a length of 10 ms but is divided into 2 sub-frames of 5 ms with 7 traffic time slots each. The uplink and downlink directions are separated by two switching points per sub-frame. Between the first and the second time slot of a sub-frame two synchronisation slots

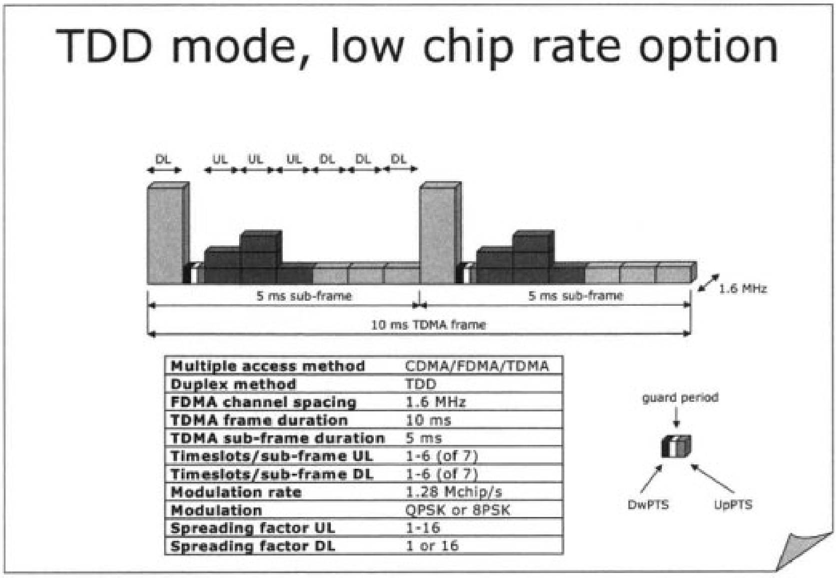


Figure 7.7: Multiple access in the TDD mode low chip rate option

Downlink Pilot Tone Slot (DwPTS) and *Uplink Pilot Tone Slot* (UpPTS) as well as one guard period are located. Figure 7.7 shows the multiple access scheme of the TDD low chip rate option. The first slot of each sub-frame is always allocated to the downlink while the second time slot is always allocated as uplink. The position of the second switching point can be arbitrarily chosen from the remaining time slot boundaries. Thus, the slot configuration can be adjusted to asymmetric traffic as in the high chip rate TDD mode.

Since more than one frequency channel can be used by one operator, the multiple access of low chip rate TDD comprises FDMA with a carrier spacing of 1.6 MHz.

7.4 Power control

7.4.1 Power control in FDD mode

As explained earlier, power control in UMTS is a particularly important mechanism for preventing near-far effects . Figure 7.8 illustrates the principle behind power control in FDD mode. The control mechanism basically consists of two loops, an outer one and an inner one. The inner loop controls on the time basis of time slots. A power control command can be transmitted in

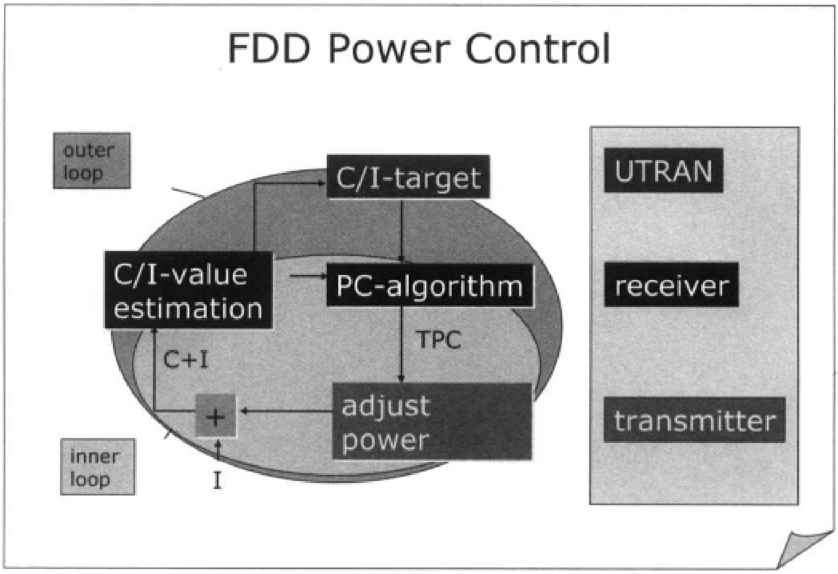


Figure 7.8: Power control in FDD mode

each time slot, i.e., 1500 commands are transmitted per second. An appropriate signalling channel is used for this purpose. The outer loop works on the time basis of frame lengths, thus 10 ms intervals. The outer loop is always implemented in the *Radio Network Controller* (RNC) of the UTRAN. It is responsible for establishing the target value for the inner loop.

The controlled variable of the power control in UMTS is the C/I (Carrier-to-Interference Ratio). The RNC sets a C/I target value and can change it at 10 ms intervals. The receiver, irrespective of whether it is a base station or a mobile station, estimates the C/I actual value and using a prescribed algorithm generates a power control command for the transmitter. Taking into account conditions such as technological limitations, the receiver then changes its transmitting power. This mechanism is executed for all connections within the network. The interference power experienced by the receiver depends on many parameters, particularly on the traffic load in the network. The changing interference power and changing transmitter power of the transmitter produce a changeable C/I ratio that in turn leads to appropriate power control commands.

The estimation of the C/I value uses a characteristic of CDMA systems, which can extract the desired signal from the aggregate signal of user and interference signals through correlation with the associated code sequence. This enables a relatively exact identification of the signal-to-interference ratio. Second-

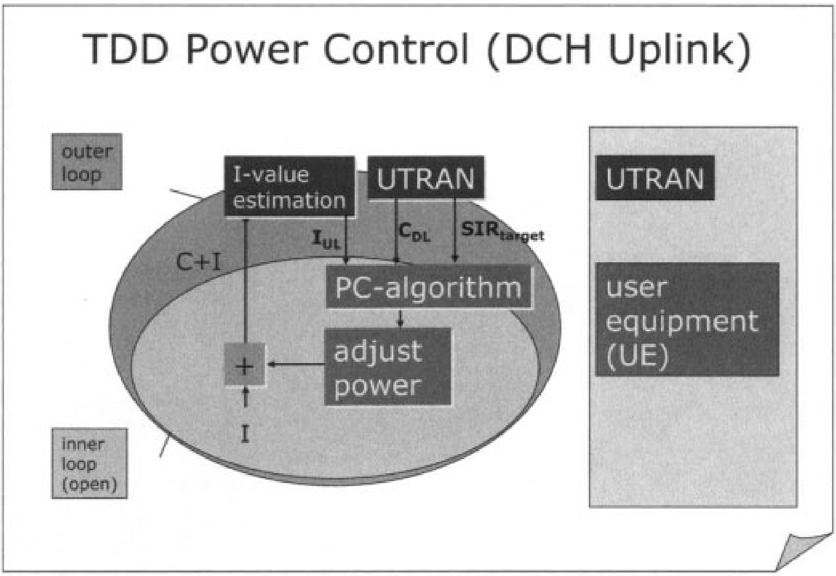


Figure 7.9: Power control on the uplink in TDD mode

generation TDMA/FDMA systems, such as GSM, do have this possibility. A GSM receiver cannot decide how much of the received power is a user signal or how much is an interference signal. Power control in these systems is therefore usually based on an estimation of the bit error rate.

The C/I target value depends on a multitude of factors and is determined by the RNC. The bit error rate, along with the spreading factor, the channel coding, the speed of the users and many other factors, has a considerable influence on the C/I target value. The capacity of the physical channels that transmit user and control data is always controlled at the same level, i.e., only one power control exists for each connection, even if several different physical channels are used. Although the C/I target value in the outer loop is adjusted in 10 ms intervals, the transmitter power can be altered in intervals of one time slot. On the uplink a period of three time slots, on the downlink a period of five time slots, is also possible.

The transmitter power is matched in fixed steps, typically in steps of one decibel. Since the mobile station generates commands for changing the transmitter power of the base station, reference is also made to ***mobile-controlled power control.*** This procedure differs from GSM power control where the base station controls its own transmitter power on the basis of measured values supplied by the mobile station.

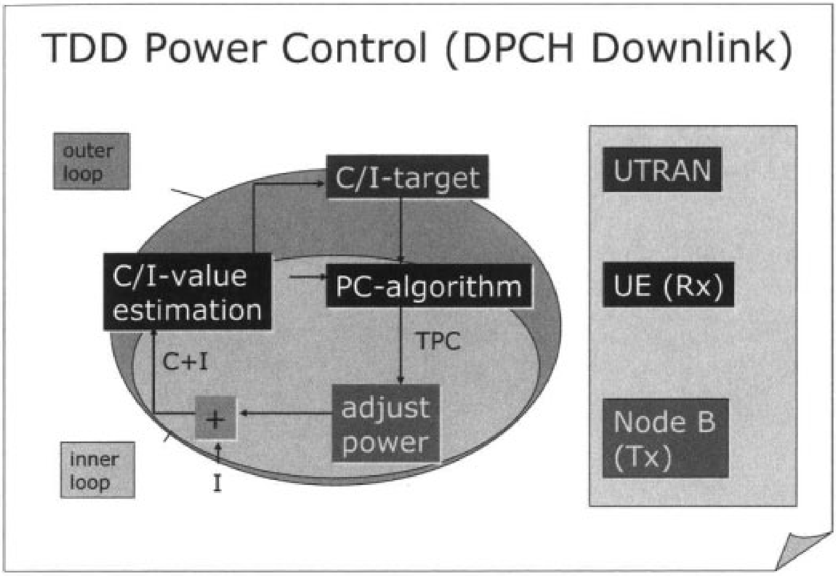


Figure 7.10: Power control on the downlink in TDD mode

7.4.2 Power control in TDD mode

The power control used in TDD mode is slower compared to the one in FDD mode. Although such a power control is basically less specific, it makes sense because of the TDMA components in the multiple-access technique of TDD mode. In TDD mode a channel is identified not only by the spreading code but also by a time slot. Consequently, only a part of the transmission capacity of a 10 ms long time frame is allotted to such a channel. Therefore, short codes with a small spreading factor are necessary for achieving sufficiently high bit rates. As already shown, the number of short, orthogonal codes is low, and, consequently, only a small number of users can be active simultaneously within a time slot. These two conditions - short code sequences and a small number of users at any one time - favour the implementation of receivers with joint detection that are resistant to near-far effects. Fast, precise power control is needed in FDD mode particularly for the prevention of near-far effects.

It should be noted that fast power control is difficult to implement in TDD mode. The reason is that mobile stations only transmit for a fraction of the duration of a time frame and the channel state can change significantly when a mobile station moves to the next transmission point.

Figure 7.9 shows the two interleaved loops for power control on the uplink, with the inner loop open. The received interference power is estimated in Node

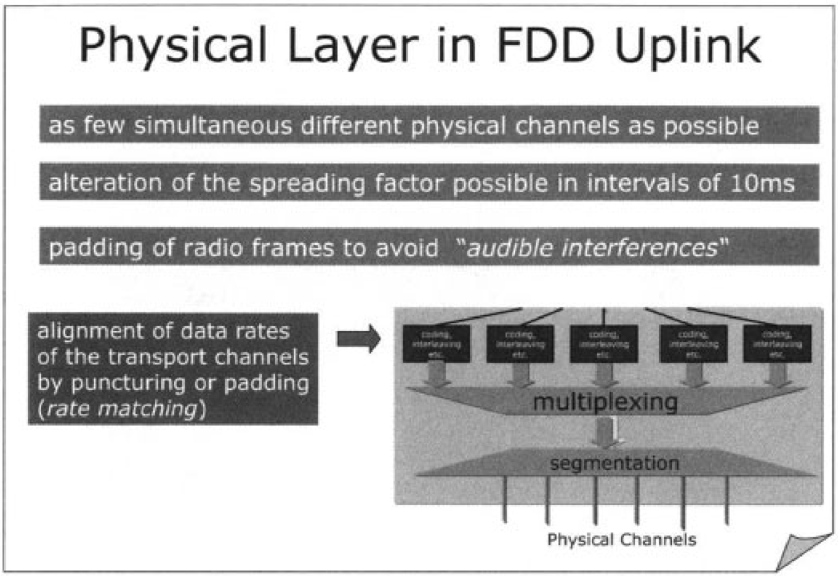


Figure 7.11: Structure and principles of the physical layer on the FDD uplink

B, and the mobile station is notified of the amount of interference power and the C/I target value. The mobile station measures the carrier signal power and, using an instruction described in the standard, calculates the transmitter power it needs. Depending on the switching points selected within a time frame between uplink and downlink, the transmitter power can be changed once or even several times in a time frame, i.e., within 10 ms.

Power control on the downlink in TDD mode follows the same principle as in FDD mode with two closed loops interleaved in one another (compare Figure 7.10). For the reasons mentioned above, power control on the TDD downlink is considerably slower than on the uplink, for example, with a period of 10 ms.

To summarise: on the TDD uplink the inner loop is open, the transmitter power is calculated by the mobile station on the basis of measured and received parameters. The downlink uses what is referred to as ***mobile-controlled closed loop power control***, the same as in FDD mode. The control interval varies, depending on the division of timeslots on the uplink and the downlink. The power control stepsize on the downlink is 1 dB, 2 dB or 3 dB; no spacing is used on the uplink.

According to the standard, the difference between the lowest and the highest transmitter power of the dedicated physical channels of a downlink time slot is

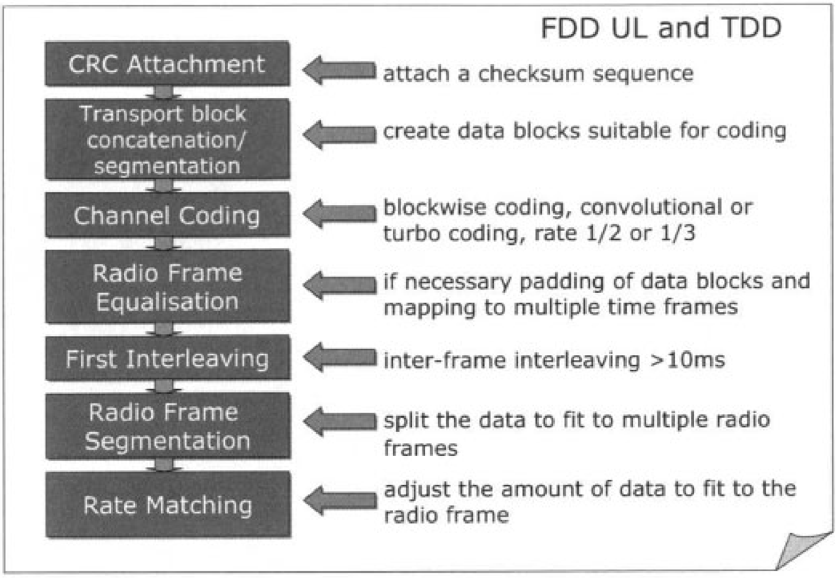


Figure 7.12: Handling of transport blocks on the FDD uplink and in TDD mode

permitted to be a maximum of 20 dB to guarantee the functioning of receivers with joint detection.

7.5 Channel coding, multiplexing and interleaving

As explained earlier, the physical layer provides transport channels over which the MAC layer can transmit data blocks of different sizes. For error protection each data block is subject to channel coding and interleaving. In addition, the data blocks of different transport channels are matched so that they can be distributed and transmitted over one or more physical channels based on serial multiplexing (see Figure 7.2). Thus a transport channel cannot immediately follow a physical channel as long as the parameters for rate matching, multiplexing and segmentation are not known.

7.5.1 TDD mode and FDD uplink

The transmission capacity of transport channels is matched to that of the physical channels in FDD mode on the uplink through puncturing, a systematic removal of coded bits, or through padding, a systematic enlargement of data blocks. As a result, a transmitting mobile station always transmits

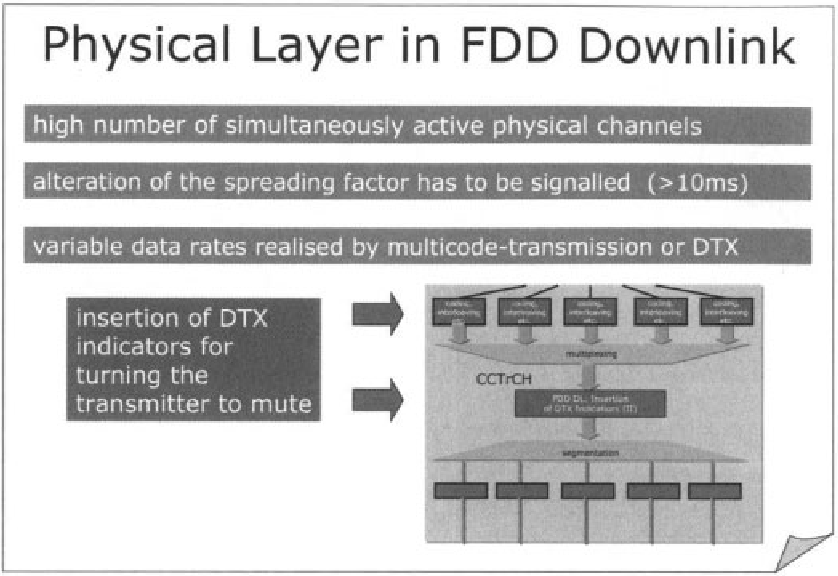


Figure 7.13: Structure and principles of the physical layer on the FDD downlink

for the duration of an entire time frame, which helps to prevent the type of audible interference that occurs in GSM [10].

Since each mobile station uses a specific interleaving code in FDD mode, the entire family of OVSF codes is available to it. Spreading codes and the spreading factor can therefore be changed at short intervals, i.e., every 10 ms. The aim is that as few physical channels as possible should be occupied simultaneously (see Figure 7.11).

In FDD *Uplink* (UL) and in TDD mode each transport block passes through the chain of protection and matching functions shown in Figure 7.12. First a check sum is added to each block for the detection of transmission errors that cannot be corrected by forward error correction. Depending on the size of the transport block, these checksums are 8 bits, 12 bits, 16 bits or 24 bits long; there is also the option of dispensing with the checksum.

Data blocks of a certain size are then created through segmentation, attachment or padding as demanded by the appropriate channelling coding scheme. The channel coding technique can either be convolutional coding with the rate 1/2 or turbo coding with the rate 1/3. Uncoded transmission is also possible as an option.

A data block is always transmitted in one or more time frames with the same rate. The number of bits in the coded block must therefore be a multiple of

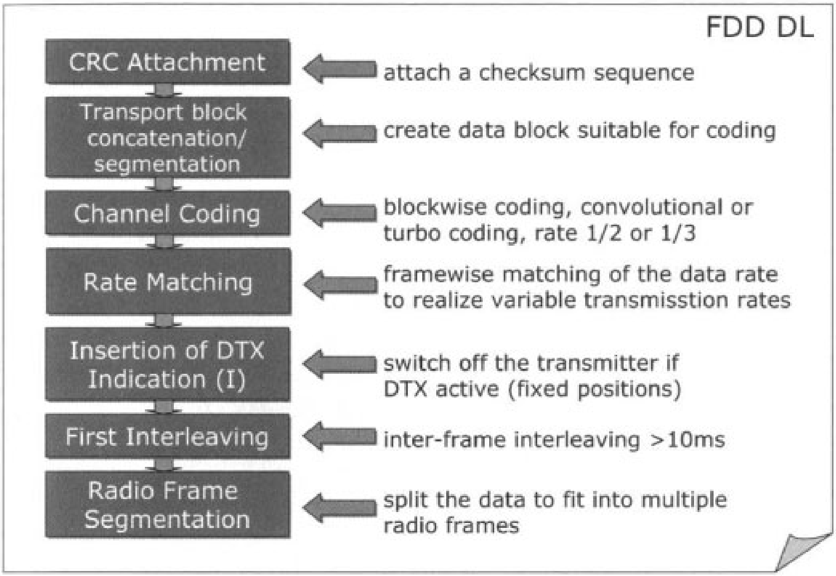


Figure 7.14: Handling transport blocks on the FDD downlink

the data set that can be transmitted within a time frame. If the length of the coded data block is not appropriate, the data block is equalised. The coded data blocks are then interleaved with a depth of at least one frame length.

Each transport block of a transport channel passes through this chain of protection and matching functions. The coded data blocks are systematically increased or reduced through puncturing to match the resulting data streams to the transmission capacity of the physical channel(s). In the latter case of puncturing, individual bits of the coded data block are removed according to a prescribed algorithm. This lowers the effective coding rate.

7.5.2 FDD downlink

The problem of audible interference is not relevant on the downlink. This is essentially due to two reasons: first, the distance between a base station and a noisy device is typically much greater than the same distance in an uplink situation. Second, there is less of a variance in the transmitter power of a base station because several users are typically served at the same time [10].

A base station uses a specific scrambling code (according to the standard up to three different scrambling codes are permitted). Consequently, the family of OVSF spreading codes is available only once to the physical layer on the

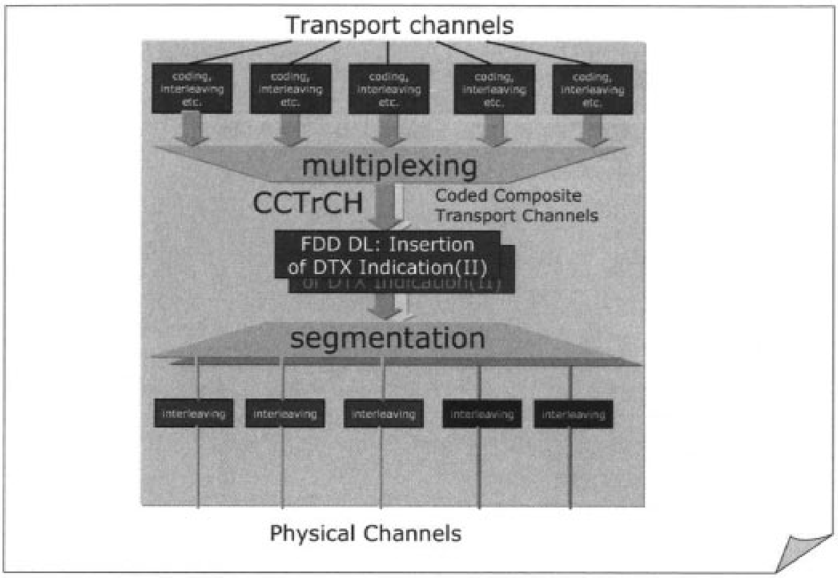


Figure 7.15: Structure of the physical layer in Node B (downlink)

downlink. Since a base station communicates with more than one user, the number of codes available for communication with a user is limited. The selection or the changing of a spreading code directly affects the quantity of spreading codes available for other communication. Consequently, on the downlink the spreading code and, as a result, the spreading factor, of a physical channel is changed at intervals of more than 10 ms based on negotiation by *Radio Resource Control* (RRC).

Since there is no commitment to transmit a burst in each slot of a downlink frame, the transmitter can be switched to mute at times when no data has to be transmitted over a transport channel. This technique, called *Discontinuous Transmission* (DTX), reduces interference and is used for voice transmission because users typically only really talk during half the talk time and otherwise are listening to the other person. Since, as a general rule, a physical channel is not exclusively allocated to a transport channel, DTX occurs through the insertion of so-called DTX indicators in the data stream at an appropriate place in the chain of channel protection and matching functions (see Figure 7.13). These DTX indicators are characters that exist in a data stream but are not transmitted. The data stream can be envisaged as a bipolar representation in this case. The insertion of zeros at the appropriate places in the data stream ensures that a modulator anticipating either a +1 or a -1 will not transmit anything in the case of a zero.

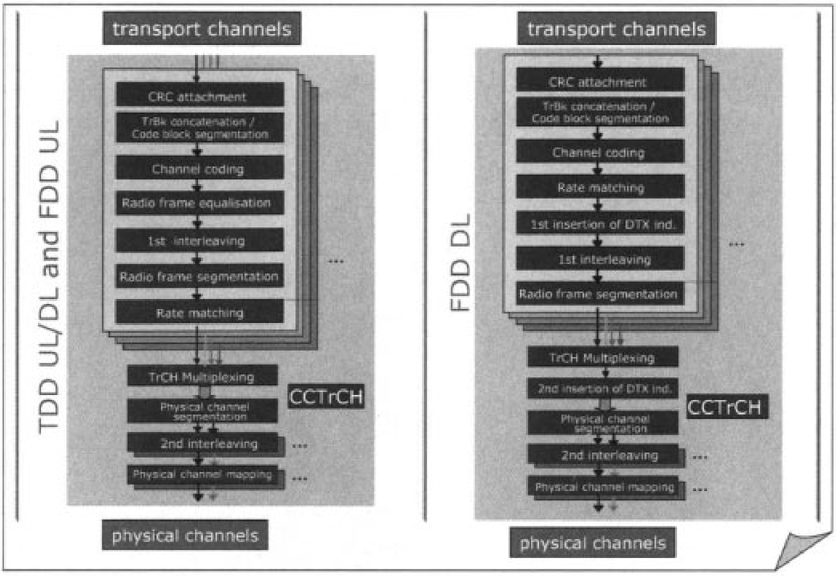
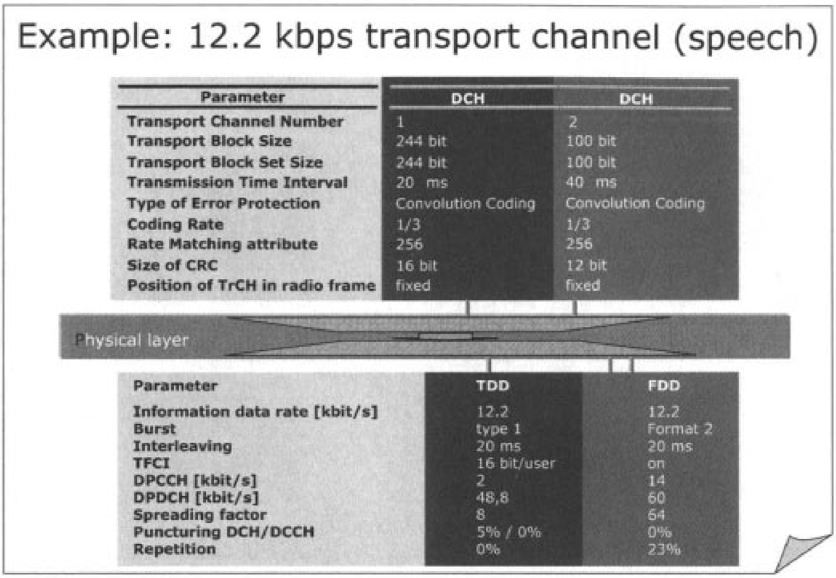


Figure 7.16: Mapping of transport channels to physical channels in FDD mode and TDD mode

The chain of protection and matching functions that each transport block passes through on the downlink is, as illustrated in Figure 7.14, very similar to the chain for the uplink. The attachment of the checksum and the channel coding are the same; the matching of the data set to the transmission capacity of the physical channels occurs on the downlink before the addition of the DTX indicators, the interleaving and the segmentation in equal-sized data blocks for transmission within a time frame.

On the downlink the rate matching function is also responsible for the implementation of variable transmission rates when the spreading factor cannot be changed or cannot be changed quickly enough. On the uplink this is possible every 10 ms; on the downlink it is only possible at considerably larger intervals.

The physical layer on the downlink, which is implemented in Node B, contains partner instances of entities in the physical layers of several mobile stations. A *Coded Composite Transport Channel* (CCTrCH) exists for each dedicated connection on the downlink. Segmentation, second interleaving as well as mapping to physical channels occur separately for each connection (see Figure 7.15). A dedicated physical channel therefore also only contains the transport blocks of one connection.



Example: 12.2 kbps transport channel (speech)

Parameter	DCH	DCH
Transport Channel Number	1	2
Transport Block Size	244 bit	100 bit
Transport Block Set Size	244 bit	100 bit
Transmission Time Interval	20 ms	40 ms
Type of Error Protection	Convolution Coding	Convolution Coding
Coding Rate	1/3	1/3
Rate Matching attribute	256	256
Size of CRC	16 bit	12 bit
Position of TrCH in radio frame	fixed	fixed

Parameter	TDD	FDD
Information data rate [kbit/s]	12.2	12.2
Burst	type 1	Format 2
Interleaving	20 ms	20 ms
TFCI	16 bit/user	on
DPCCH [kbit/s]	2	14
DPDCH [kbit/s]	48,8	60
Spreading factor	8	64
Puncturing DCH/DCCH	5% / 0%	0%
Repetition	0%	23%

Figure 7.17: Implementation of 12.2 kbit/s voice connection (transport channel)

7.5.3 Summary

Figure 7.16 summarises the functions of the physical layer in FDD and TDD modes. It is noticeable that the physical layers in TDD mode are structurally the same as those on the uplink in FDD mode, whereas individual functions, such as mapping to physical channels, are different in TDD and FDD mode based on the respective multiple-access technique used.

7.6 Mapping of 12.2 kbit/s voice transport channel

The parameters for mapping transport channels to physical channels for typical voice transmission on the uplink are shown in Figure 7.17. The data blocks of two different transport channels, here Dedicated Channels (DCH), are transmitted over the physical layer. The characteristics of data transmission are stipulated in the transport formats.

Transmission over the first transport channel is at intervals of 20 ms per each 244-bit transport block. This corresponds to a net data rate of 12.2 kbit/s. Each data block receives convolutional coding at the rate 1/3 in the physical layer. At the same time signalling in the higher layers is transmitted over a second transport channel at intervals of 40 ms in data blocks the length of 100 bits. In the physical layer the transport blocks pass through the chain of protection measures described above and are multiplexed and segmented accordingly so they can be transmitted over the same physical channel. Half a transport block of the first transport channel and a quarter of the second transport channel are transmitted per 10 ms radio frame.

Depending on the operating mode, TDD or FDD, physical channels that provide the required transmission capacity are selected. In FDD mode this could be a physical channel with the spreading factor 64, which transmits at a gross rate of 60 kbit/s. Since this transmission rate of the physical channel is higher than is necessary for the transport of coded transport blocks, the data set is artificially increased by 23% through repetition in the physical layer. On the other hand, a physical channel with the spreading factor 8 and one time slot per time frame could be selected in TDD mode. This channel can transmit at 48.8 kbit/s, which is too little for the coded transport blocks. Therefore, in this case 5% of the bits are removed systematically from the coded code blocks in the physical layer in order to match the gross data rate.

The physical channels and corresponding gross transmission rates will be covered in detail in a later chapter.

7.7 Questions

7.1 Sketch multiple access in FDD and TDD mode of the UTRA radio interface (if possible, using numerical values). Mark the resource used by a subscriber.

7.2 What are the transmission bandwidth, the chip rate and the maximum and minimal data rates on the UTRA radio interface (in FDD and TDD)?

7.3 What are advantages and disadvantages of a multiple switching point configuration in TDD mode?

7.4 What are the functions of the physical layer in the UTRA protocol stack?

7.5 What are the reasons for using *power control* (PC) methods?

7.6 How do the PC methods for UL and DL differ in FDD mode?

7.7 How does power control differ between TDD and in FDD mode?

7.8 What is the purpose of the outer control loop of FDD power control?

7.9 What is a CCTrCH and what are its distinguishing characteristics?

7.10 What does DTX stand for?

7.11 How is the DTX function realised on the FDD UL?

7.12 How is the DTX function realised on the FDD DL and why are there differences compared to the FDD UL?

7.13 In what way does the physical layer basically ensure that data can be transmitted over transport channels with variable data rates?

8 Physical Channels and Procedures at the Radio Interface

8.1 Physical channels in the UTRA protocol stack

Figure 8.1 shows the position of the physical channels in the structure of the UTRA protocol stack. A distinction is generally made between logical channels, transport channels and physical channels. Logical channels are the services of the *Medium Access Control* (MAC) layer and are characterised by the type of data transmitted. Transport channels are the services of the physical layer and are characterised by the parameters of the data transmission, for example, the transmission rate. The physical channels are used for communication between physical layers at the radio interface. Since UMTS is also a CDMA system, spreading codes are an important property of physical channels. The configuration of the physical layer and, as a result, also the physical channels is carried out by the *Radio Resource Control* (RRC) layer that maintains a direct communication relationship with the physical layer for this purpose.

8.2 Physical channels in FDD

A physical channel in FDD mode is basically characterised by a spreading code and the frequency channel. On the uplink in FDD physical channels are distinguished in addition by the phase position of the carrier signal; physical channels therefore use either a cosine or sine wave as the carrier signal. This is implemented by a different physical channel being transmitted over the one branch of QPSK modulation rather than over the other branch. Therefore one also hears of BPSK modulation being used on the uplink of FDD mode, which is correct if one is only referring to one physical channel. In TDD mode a physical channel is defined by the spreading code, the time slot and the frequency channel.

A general distinction is made between *dedicated physical channels* and *common physical channels* (see Figure 8.2). A dedicated channel is used exclusively by one connection and is allocated separately for each connection set-up and sometimes during a connection. Common channels are used simultaneously or alternately by several connections. For instance, the system information sent out by each base station is transmitted over common physical

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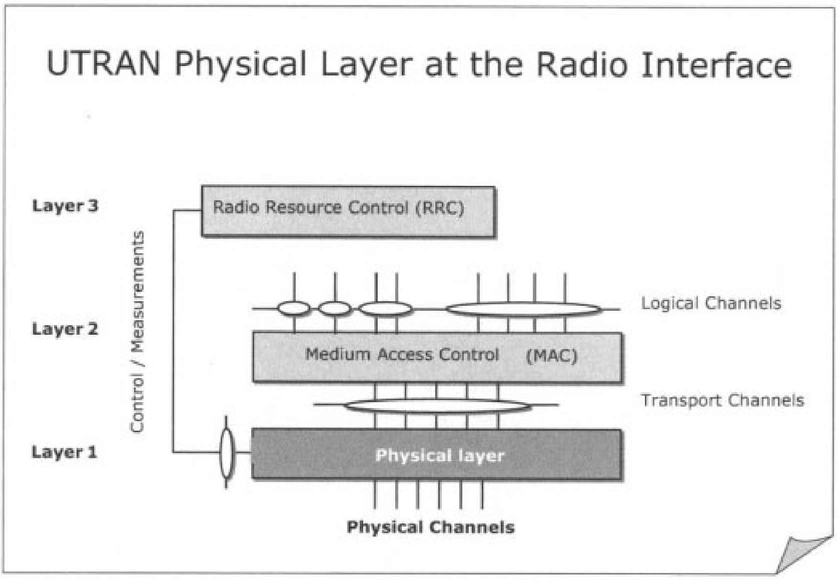


Figure 8.1: Physical channels in the UTRA protocol stack

channels that every mobile station can receive. There are also common physical channels that are used for packet data transmission. The use of a common channel always requires separate addressing for the identification of sender and receiver.

The physical layer maps transport channels to physical channels. Figure 8.3 shows the transport channels and their mapping to physical channels in FDD mode. In addition to physical channels, so-called indicators also exist in FDD mode. These are one or two-bit messages that are spread with a code sequence and transmitted at a specific time. Code, frequency and time characterise an indicator. Indicators are used for the notification and indication of certain events. For instance, a user is paged over an indicator. With 10 ms long time frames per each 15 time slots, indicator channels are generally subject to a different time structure than dedicated physical channels.

As is evident in Figure 8.3, not every transport channel can be mapped to a physical channel. The physical channels are as follows:

- **Dedicated Physical Data Channel (DPDCH):** The *Uplink Dedicated Physical Data Channel* (DPDCH) only exists on the uplink and is used to transmit user and signalling data from the higher layers. A layer 1 connection can have one, several or no DPDCH.

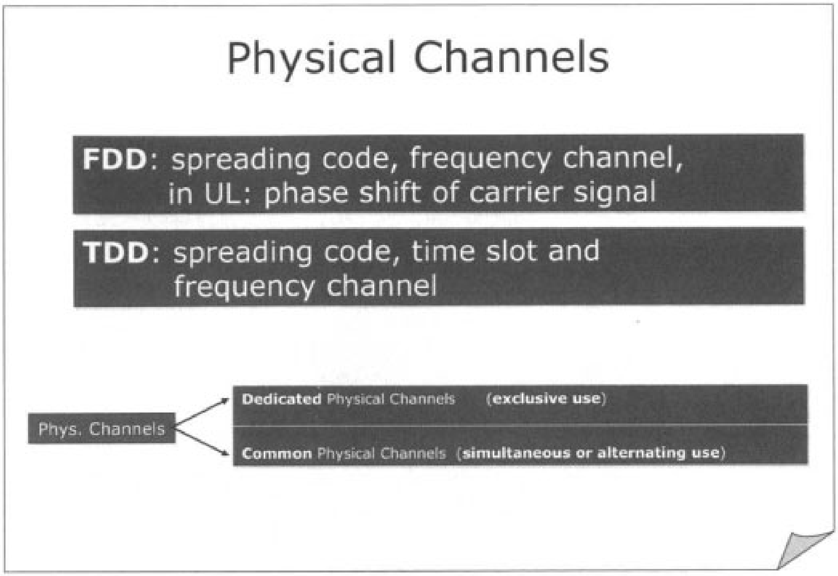


Figure 8.2: Definition of physical channels in FDD and TDD mode

- **Dedicated Physical Control Channel (DPCCH):** The *Dedicated Physical Control Channel* (DPCCH) is a physical channel that controls data transmission between partner entities of the physical layer for the uplink. Only information of the physical layer is transmitted over this channel. This includes power control commands, transport format indicators or pilot bits. Each layer 1 connection has exactly one DPCCH.
- **Dedicated Physical Channel (DPCH):** On the downlink the DPDCH and the DPCCH are implemented in a physical channel, the *Dedicated Physical Channel* (DPCH).
- **Physical Random Access Channel (PRACH):** Messages of the random access channel (RACH) are transmitted over the *Physical Random Access Channel* (PRACH). The RACH can be used for call set-up as well as for the transmission of small data packets.
- **Physical Common Packet Channel (PCPCH):** Packet data of the CPCH is transmitted over the *Physical Common Packet Channel* (PCPCH) through use of a *Carrier Sense Multiple Access with Collision Detection* (CSMA/CD) technique.
- **Common Pilot Channel (CPICH):** A *Common Pilot Channel* (CPICH) supports macro diversity on the downlink.

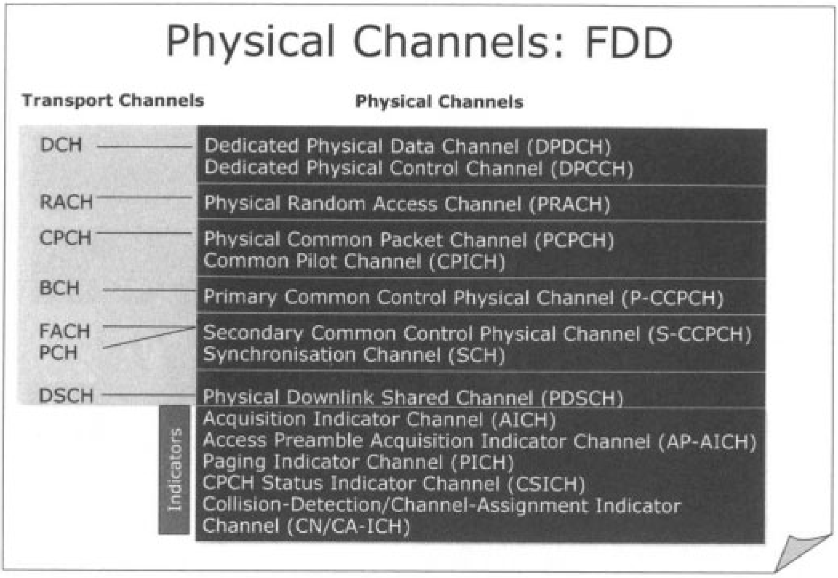


Figure 8.3: Overview of physical channels in FDD mode

- **Common Control Physical Channel (CCPCH):** Distribution services are implemented over a *Common Control Physical Channel* (CCPCH) on the downlink. The CCPCH separates into two subchannels: information of the BCH is transmitted over the *Primary Common Control Physical Channel* (P-CCPCH) and the FACH and PCH are mapped to the *Secondary Common Control Physical Channel* (S-CCPCH).
- **Synchronisation Channel (SCH):** The *Synchronisation Channel* (SCH) is a downlink channel and is used in the search for cells and the synchronisation of mobile stations. It is divided into two subchannels, see Section 8.2.5.
- **Physical Downlink Shared Channel (PDSCH):** The *Physical Downlink Shared Channel* (PDSCH) is used to transmit data over the DSCH on the downlink. A DPCH is always allocated to the PDSCH. Several mobile stations share this channel.

The four most important physical channels are explained in detail in the following sections: DPDCH, DPCCH, DCH and PRACH.

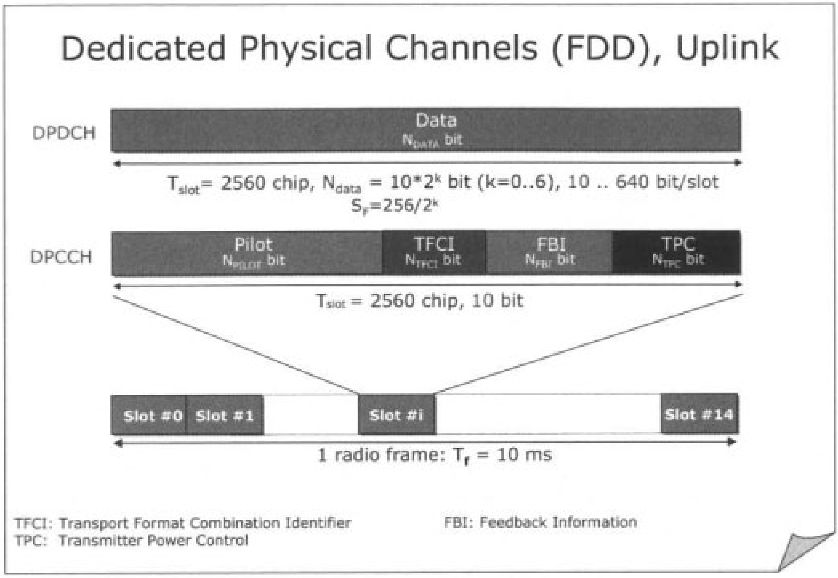


Figure 8.4: Structure of DPDCH and DPCCH

8.2.1 Dedicated transmission on the FDD uplink

Figure 8.4 shows the burst structure of the DPDCH and the DPCCH on the FDD uplink. A burst has the length of 2560 chips, equivalent to 10/15 ms. Thus exactly 15 bursts fit into one time frame. Since a time frame on the FDD uplink is always filled, 15 bursts per frame are continuously transmitted, i.e., without a guard time. The DPDCH only carries coded and interleaved user data of the higher layers, transmitting at between 10 bits and 640 bits per burst depending on the spreading factor. A DPDCH can therefore transmit a maximum of 9600 bits in 10 ms, which with a minimal spreading factor of four corresponds to a transmission speed of 960 kbit/s. The DPCCH exclusively transmits signalling between the physical layer in the mobile station and the physical layer in the base station. The spreading factor of the DPCCH is always 256, i.e., a DPCCH burst carries exactly 10 bits. The significance of these bits is represented in Figure 8.4: each burst contains a number of pilot bits used for channel assessment. In addition, a *Transport Format Combination Identifier* (TFCI) that indicates the transport formats of the multiplexed transport channels on the DPDCH is transmitted. With this information the physical layer on the receiver side is able to reconstruct the transport blocks of the individual transport channels from the data stream of the physical channel. The *Feedback Information* (FBI) field is used for signalling with soft handovers and the *Transmitter Power Control* (TPC) field carries a power control command (increase, maintain or reduce power). There are six different possible configurations of a DPCCH burst, each differing by the length of the individual fields. For instance, the TFCI is valid for the duration of an entire frame and therefore only has to be transmitted once every 10 ms.

There is exactly one DPCCH for each connection on the uplink. One or several DPDCH can be used simultaneously for transmission. A DPCCH always has the spreading factor 256 and is always transmitted over the quadrature branch of modulation. If more than one DPDCH is used simultaneously, then all DPDCH must have the same spreading factor between four and 256, with a maximum of six DPDCH able to transmit at the same time. The DPDCHs are distributed as evenly as possible over the in-phase and quadrature branches. As pointed out in Section 6.13, BPSK modulation is also mentioned in connection with a DPDCH.

Variable transmission rates can thus be implemented on the uplink by a variation of the spreading factor and through multicode transmission (see Figure 8.5). The coded, interleaved and multiplexed transport blocks are segmented in the physical layer and distributed over the DPDCHs. Each DPDCH data stream is spread with a spreading code C_d and then, based on their spreading factors, the individual physical channels are weighted against each other with the factor β. The resulting chip streams are added, scrambled and transmitted over the in-phase or the quadrature branch of QPSK modulation.

The gross bit transmission rates realisable with the various spreading factors for an individual DPDCH are shown in Figure 8.6. Depending on the spreading factor, a DPDCH burst transmits between 10 bits and 640 bits per slot. Based on 15 slots per frame and a frame duration of 10 ms, with a maximum spreading factor of 256 this equates to a minimal transmission rate of 15 kbit/s. The maximum transmission rate of a DPDCH is 960 kbit/s; at 12.2 kbit/s gross the standard-compliant reference channel for voice transmission has a spreading factor of 64 and a gross transmission rate of 60 kbit/s.

The spreading factor 256 is used for the DPCCH. It is noticeable that the DPCCH, which is solely necessary for signalling between the physical layers, requires a transmission capacity of 15 kbit/s. Of this amount, at least one bit per burst are power control commands, i.e., the power control constantly requires a transmission rate of at least 1.5 kbit/s.

Data rates of more than 960 kbit/s are only possible through the parallel use of several DPDCH. The set of OVSF codes is available one time only in the in-phase branch as well as in the quadrature branch of modulation, because a DPDCH is determined not only by code and frequency but also by the phase position of the carrier wave. A maximum of six DPDCH can be implemented simultaneously, which theoretically results in a maximum transmission rate of 5740 kbit/s gross. However, the standard only describes mobile stations that can achieve a maximum gross transmission rate of 1920 kbit/s.

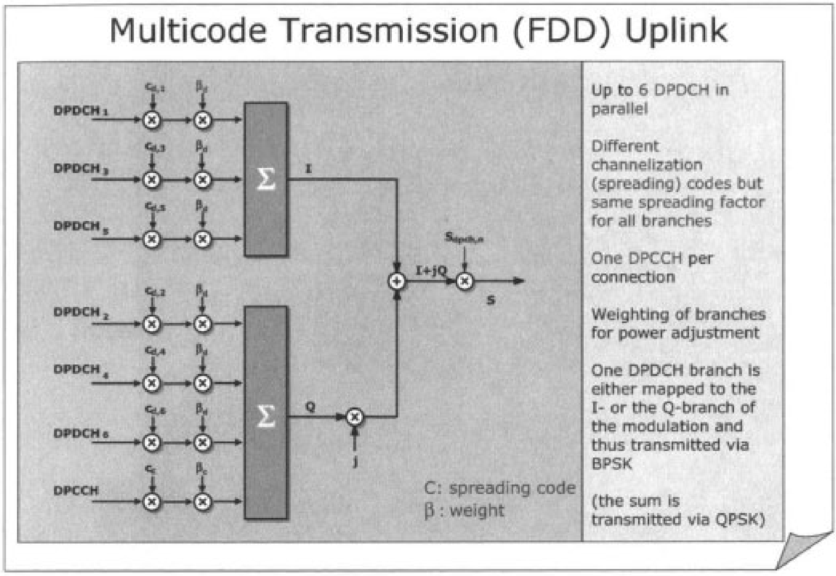


Figure 8.5: Multicode transmission on FDD uplink

8.2.2 Dedicated transmission on the FDD downlink

Physical channels on the downlink are not separated by the phase of the carrier wave. Layer 1 and layer 2 information, mapped on the downlink to DPDCH and DPCCH, is therefore transmitted in time multiplex on the downlink. There is only one dedicated channel, the *Dedicated Physical Channel* (DPCH), which takes over the tasks of the DPDCH and the DPCCH.

The structure of a DPCH burst is shown in Figure 8.7. A burst contains 2560 chips, corresponding to the duration of 10/15 ms. The number of bits per burst is determined by the spreading factor, which can be between four and 512. The two data fields of the DPCH burst transmit the information of the higher levels; the other fields, thus the TPC, TFCI and pilot, are used for the communication of the physical layers, equivalent to the DPCCH on the uplink.

The size of the individual DPCH burst fields varies; in total there are 17 different possible burst configurations. The burst configuration has to be negotiated at connection set-up and can be renegotiated during a connection. The TFCI field carries information about the transport formats of the transport blocks multiplexed to the DPCH.

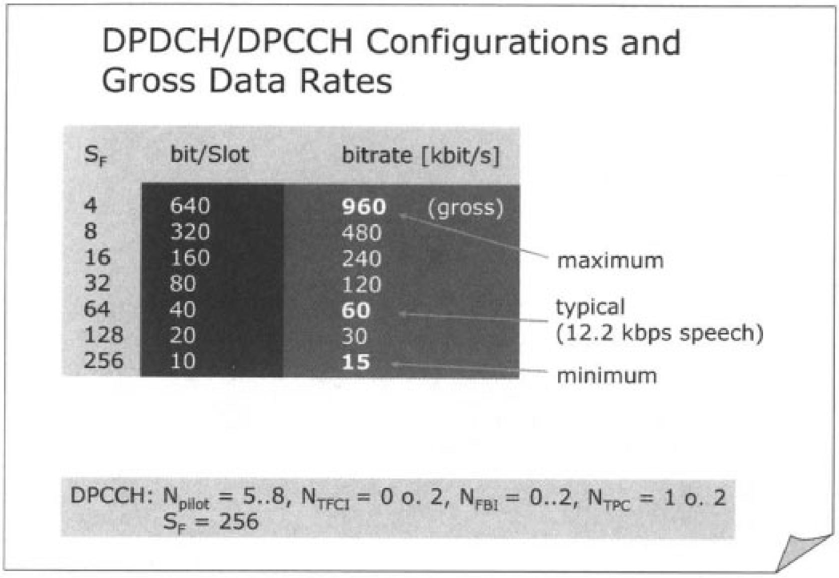


Figure 8.6: Some possible transmission rates for a DPDCH

Since two bits per symbol are transmitted on the downlink, the symbol rate always corresponds to half the bit rate. Figure 8.8 shows two bits being combined for each bit pair as well as the subsequent code spreading with the code C_{ch} and scrambling with the scrambling code S_{dl}. If the transmission speed of an individual DPCH is not sufficient for transmitting a CCTrCH, then several DPCH can be operated in parallel. Such a parallel operation is also necessary on the downlink for the simultaneous communication with different mobile stations.

With multicode transmission, which is the parallel use of several DPCH for a connection, layer 1 information between the physical layers only has to be exchanged once per connection and not once per DPCH. The respective fields of the DPCH burst are therefore only occupied in one physical channel; in all other channels they remain empty. In these time periods the transmitter power for the channel is zero (see Figure 8.9). If a CCTrCH is mapped to several DPCH, then these must have the same spreading factor.

Since each physical channel also represents a transmitter power, the total power available to a connection using multicode transmission is higher according to the number of DPCH channels. For achieving even transmitter power timewise, the DPCCH fields of the DPCH burst can be sent with increased power (see Figure 8.9). Another reason is to increase the detection

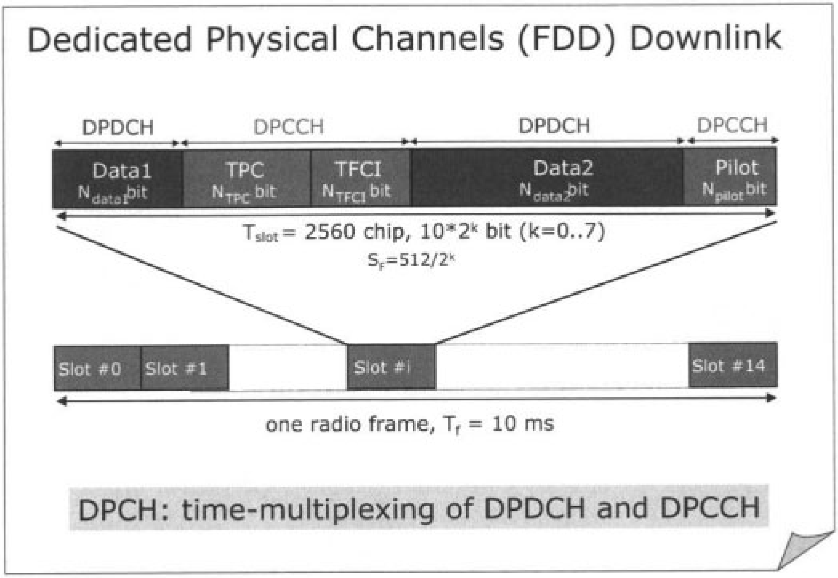


Figure 8.7: Structure of DPCH

security of the important layer 1 information, particularly of the power control commands.

The maximum achievable transmission rates are shown, along with each appropriate burst configuration, in Figure 8.10, for each spreading factor of a DPCH on the FDD downlink. The spreading factor 4 results in a gross transmission rate of 1920 kbit/s for the DPCH, with only 1872 kbit/s allotted to transmission of the transport blocks. The minimum gross user data throughput of 6 kbit/s is achievable with the spreading factor 512; the reference configuration for voice transmission at 12.2 kbit/s gross provides a spreading factor of 128, corresponding to a gross user data rate of 51 kbit/s.

The maximum achievable transmission rate for a connection depends among other things on a mobile station's ability to decode several DPCH at the same time. Since some spreading codes of the OVSF code are used by other physical channels, e.g., the CPICH, and since all DPCH must have the same spreading factor with multicode transmission, a maximum transmission rate is achievable with three parallel DPCH, each with a spreading factor of four. Such a configuration is also planned in the standard for the implementation of a 2 Mbit/s bearer service.

It is easy to understand that a 2 Mbit/s bearer service, therefore a net user data rate of 2 Mbit/s, can only be realised with a code rate far higher than 1/3. The reason is that even with multicode transmission the gross rate is

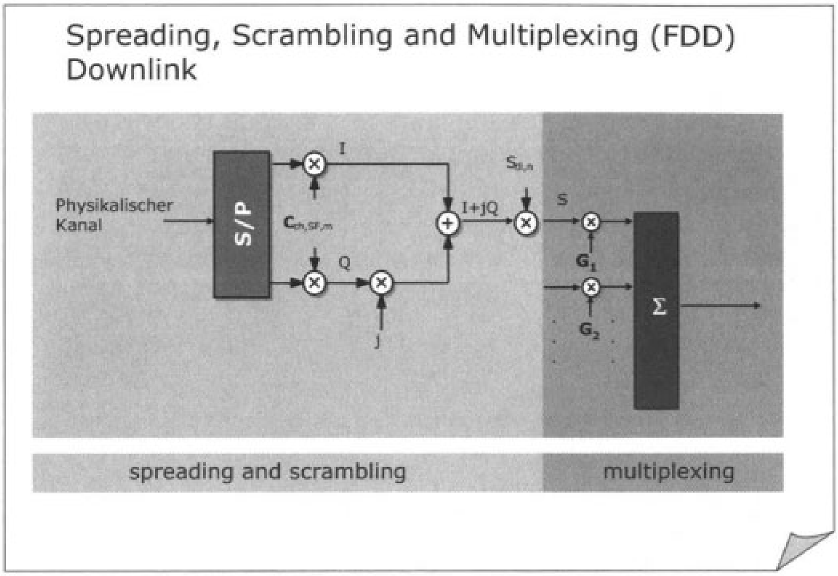


Figure 8.8: Spreading, scrambling and multiplexing on the FDD downlink

lower than 6 Mbit/s, and signalling also has to be transmitted along with the user data. A higher code rate automatically means less error protection, which only suffices with good propagation and interference conditions.

The set of OVSF spreading codes is available only once to base stations that use a scrambling code. It is clear from the restrictions described in the selection of a spreading code from the code tree that the maximum overall transmission rate of a base station for user data is the same as the maximum transmission rate per connection, which is based on the mobile station having appropriate multicode capabilities. This means that all mobile stations in a cell share the maximum user data rate of around 2 Mbit/s.

It is possible for more than one scrambling code to be used simultaneously on the downlink. Although this would increase the number of available OVSF codes, the orthogonality of the signals with different interleaving codes would be lost.

8.2.3 Compressed mode

In FDD mode, if a mobile station wants to switch to other frequency channels or to GSM, it has to receive and decode pilot signals on other frequency channels. A mobile station should not receive on different frequency channels

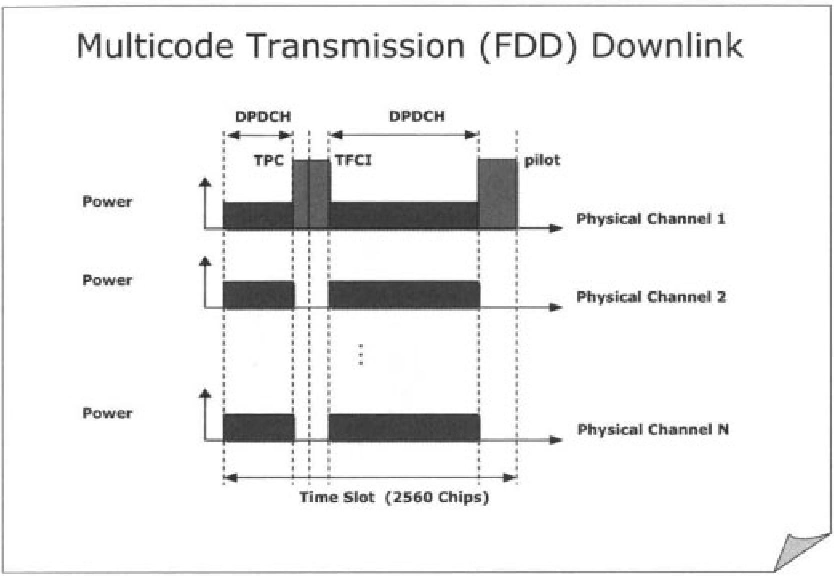


Figure 8.9: Multicode transmission on the FDD downlink

simultaneously. Therefore, it must be able to interrupt the reception of its own UMTS signals without losing any information for the period while it is receiving the pilot signals on other frequencies. Here, the so-called *compressed mode* is used to create a transmission pause, as shown in Figure 8.11. The transmission pause can be up to seven time slots long and occur either within a time frame or symmetrically at the end of one frame and at the beginning of the next frame (see Figure 8.11). There are two possibilities to ensure that no information is lost in *compressed mode*: One is to halve the spreading factor before and after the transmission pause; the other is to reduce the data being transmitted through puncturing or through disposition in higher layers.

8.2.4 Random access procedure in FDD

The *Physical Random Access Channel* (PRACH) is used for random access on the uplink. Random access can occur at defined times called *access slots*. An access slot corresponds to the duration of 5120 chips, which means that an access slot is twice as long as a normal time slot - for instance, for a DPDCH burst. Fifteen access slots each defining an *access channel* exist within 20 ms (see Figure 8.12).

Random access is divided into a contention phase and a transmission phase. In the contention phase, using a Slotted ALOHA technique, mobile stations

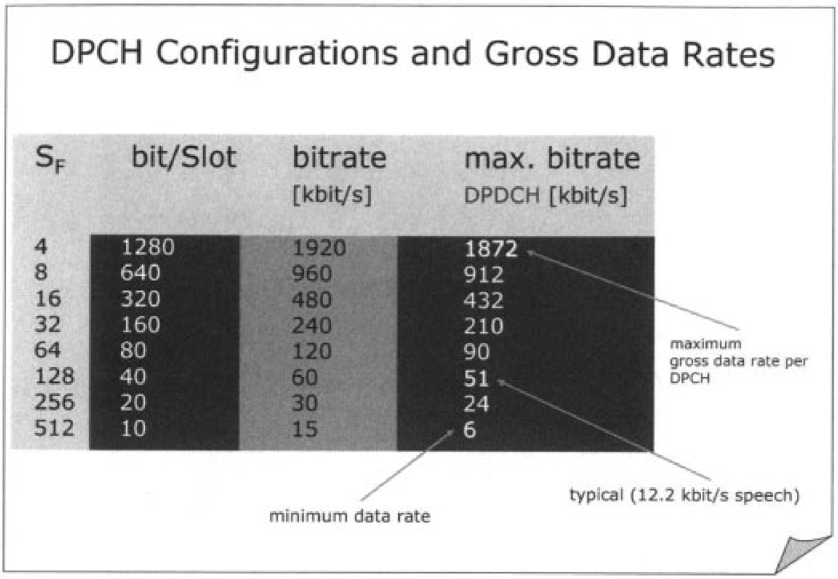


Figure 8.10: Some possible transmission rates for a DPCH

access a channel within an access slot by sending a code sequence, called a preamble. Per *access slot* there are 16 different preamble code sequences for each 4096 chips that mobile stations can use for collision-free access. This means that 16 parallel access channels are available per access slot.

The contention phase is not only used so that mobile stations can assert themselves against other mobile stations in the Slotted ALOHA technique. More importantly, a type of power control called *power ramping* is carried out in the contention phase. A mobile station without a connection to a base station also does not have a DPCCH over which power control information can be exchanged. Power ramping is able to implement power control with avoidance of near-far effects.

A mobile station that wants to access the PRACH selects an available access slot and then one of the 16 preamble sequences. The preamble is then transmitted at a lower transmitter power and an acknowledgement is awaited that is received over the *Acquisition Indication Channel* (AICH). If the mobile station does not receive an acknowledgement from the base station or if it receives a negative acknowledgement, it selects a new access slot and a new preamble and transmits both with a somewhat higher transmitter power (see Figure 8.12). This procedure is repeated for the maximum number of attempts until a positive acknowledgement is received.

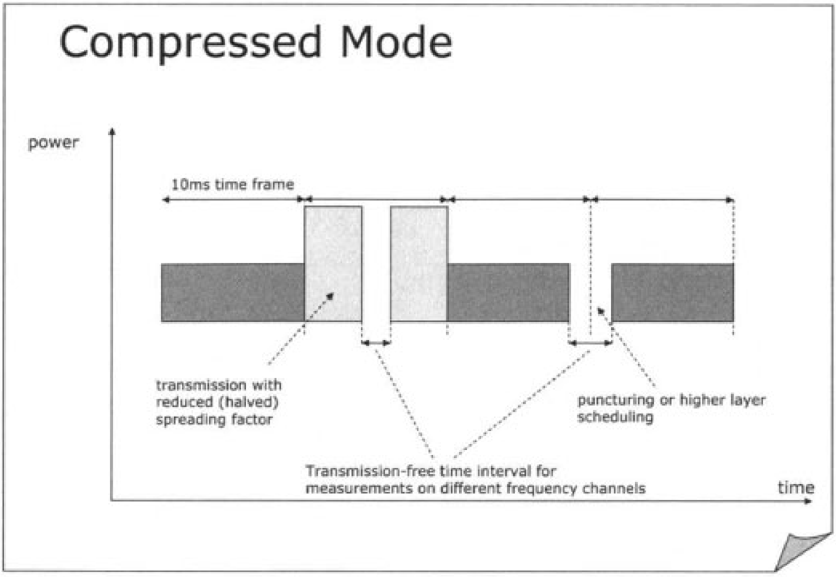


Figure 8.11: Using compressed mode

In the case of a successful contention phase, i.e., a positive acknowledgement, the mobile station transmits its random access message that may be 10 ms or 20 ms long with a delay of three or four time slots.

It is not necessary for all access slots available for random access to be available constantly in each cell. Instead the BCH notifies mobile stations of the appropriate cell configuration. This enables the allocation of different access slots to different classes of service so that a varying quality of service can be implemented for the services.

The 10 ms or 20 ms long random access message consists of bursts, illustrated in Figure 8.13. Similar to data transmission over the DPDCH, 15 bursts are transmitted every 10 ms. The minimum spreading factor is 32 so that a maximum of 80 bits can be transmitted per burst. With a message length of 20 ms a total maximum of 2400 bits (gross) is therefore transmitted.

The message bits are transmitted over the in-phase branch of the modulator. At the same time a spreading code with the spreading factor 256 is used to transmit pilot bits for channel assessment and TFCI information for the configuration of the physical layer of the receiver. As is the case with dedicated data transmission over the DPDCH/DPCCH, this control information is transmitted over the quadrature branch of the modulator.

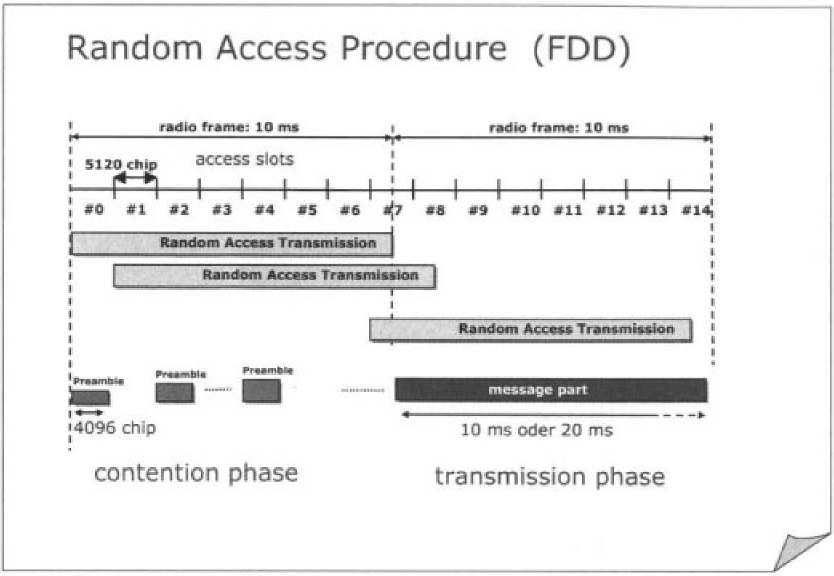


Figure 8.12: Time structure of physical random access channel

8.2.5 Cell search procedure in FDD

During the cell search procedure, the terminal searches for a cell and provides the downlink scrambling code of this cell after frame synchronisation.

The cell search procedure normally involves three steps: during the first step of the procedure the terminal uses the *Primary Synchronisation Code* (PSC) of the *Primary Synchronisation Channel* (P-SCH) to synchronise to the time slots of the cell. The PSC is the same in all cells and is exactly 256 chips long. A simple matched filter can be used to detect this chip sequence in the receive signal. Due to the autocorrelation properties of the PSC the matched filter output is a series of impulses corresponding to the time slot structure.

In the second step of the cell search procedure the terminal uses the *Secondary Synchronisation Channel* (SSC) of the secondary synchronisation channel to determine the code group of the cell identified in the first step after frame synchronisation.

The evaluation of three successive 256 chip blocks is sufficient to detect the position within the frame and to identify the code group used.

In the last step of the cell search procedure the terminal identifies the exact primary scrambling code being used by the cell. The terminal therefore compares all the possible code sequences of a group with the primary code

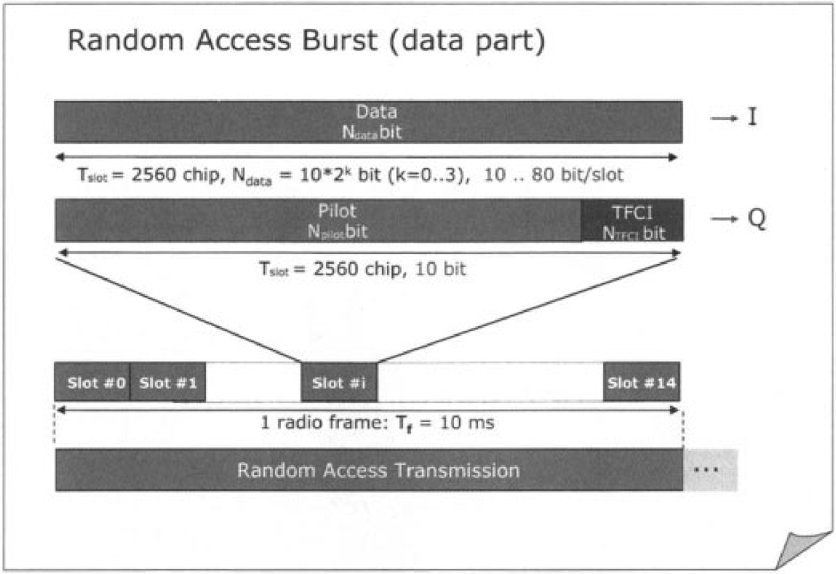


Figure 8.13: Time and burst structure with data transmission over a PRACH

used by the CPICH. This process takes place quickly because only 8 primary codes are defined in any code group. Once the primary scrambling code is established, the P-CCPCH in which the system and cell-specific information is transmitted can be read.

8.3 Physical channels in TDD mode

The physical channels in TDD mode are listed in Figure 8.14. As explained earlier, transport channels are mapped to physical channels in the physical layer. With few exceptions, the same transport channels that exist in FDD mode also exist in TDD mode; the protocol stacks of both modes differ particularly in the physical layer.

In TDD mode a physical channel is defined by the spreading code, the frequency channel and the time slot within a time frame. What is more, a physical channel can occupy a time slot in each time frame or only in a subset of all frames. In addition to a dedicated physical channel, six common physical channels also exist in TDD mode. These physical channels are:

- **Dedicated Physical Channel (DPCH):** The DPCH transmits the user and control data of a connection using dedicated physical channels. This channel exists both on the uplink and the downlink.

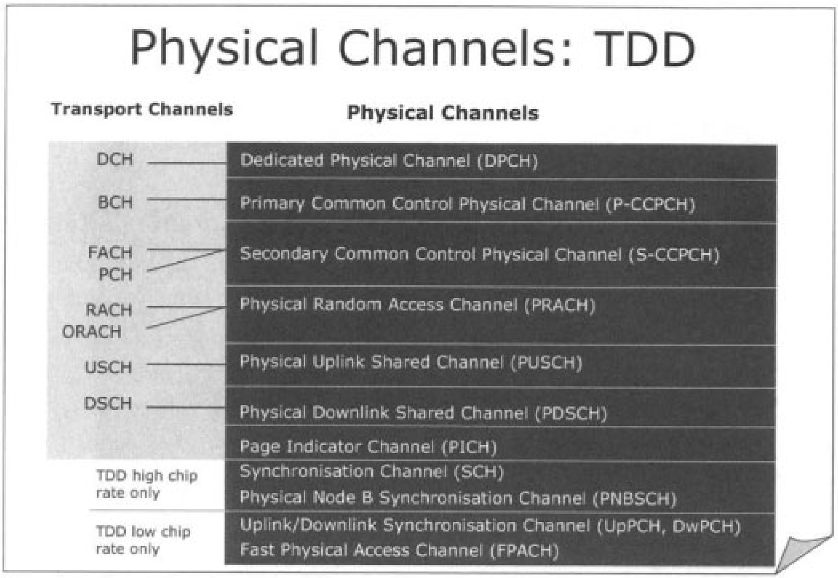


Figure 8.14: Overview of physical channels in TDD mode

- **Common Control Physical Channel (CCPCH):** Broadcast services in a cell are implemented on the downlink over a CCPCH. These services can be identified through an observation of the transport channels that are mapped to the CCPCH. The CCPCH is divided into a primary (P-CCPCH) and a secondary (S-CCPCH) subchannel. Transport blocks of the BCH are transmitted over the P-CCPCH; the P-CCPCH is used to broadcast system information within a cell. The S-CCPCH transmits the transport channels FACH and PCH and is thus used for power control and paging.
- **Physical Random Access Channel (PRACH):** Random access is carried out with the PRACH. The PRACH only exists on the uplink.
- **Physical Uplink Shared Channel (PUSCH):** The *Physical Uplink Shared Channel* (PUSCH) is a common channel, which means that it can be used by different mobile stations. User and control data is transmitted over it.
- **Physical Downlink Shared Channel (PDSCH):** The PDSCH is the counterpart to the PUSCH on the downlink.
- **Paging Indication Channel (PICH):** The *Paging Indication Channel* (PICH) carries out paging and can replace one or more subchannels for paging on the S-CCPCH.

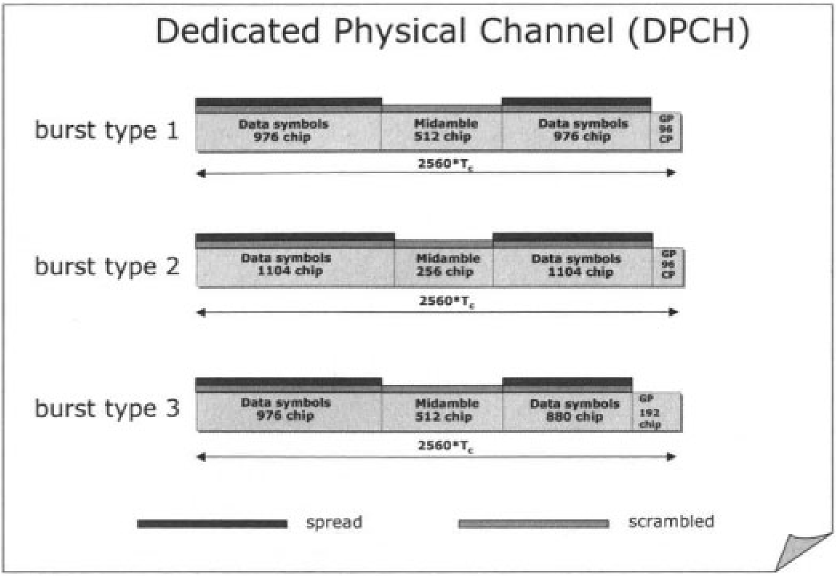


Figure 8.15: Burst structures in TDD mode

- **Physical Node B Synchronisation Channel (PNBSCH):** The *Physical Node B Synchronisation Channel* (PNBSCH) is used for mutual Node B synchronisation over the air.

Figure 8.15 shows the three different burst types in TDD mode. A burst is exactly the length of one time slot and consists of four fields: two data fields, which are separated by a so-called *midamble* sequence, and a guard time, called a *guard period* (GP), which compensates for the differences in the runtimes of the signals of different mobile stations. The three burst types are distinguished by the length of the midamble and the length of the guard period, and, consequently, in the size of the data fields.

The midamble is a training sequence and is particularly used for channel assessment and for the support of joint detection in the receiver. The longer the training sequence, the better the receiver can assess the channels of the different connections and the higher the number of the signals that can be synchronised in the receiver and received jointly. The guard period is needed to compensate for synchronisation errors and runtime differences. The shorter the guard period, the more precisely sender and receiver have to be synchronised.

Burst type 1 has a long midamble and a short guard period. It is used both on the uplink and the downlink. A maximum of 16 bursts of type 1 may be used simultaneously per slot and transmission direction, depending on the different spreading codes.

To the benefit of the length of the data fields, burst type 2 has a shorter midamble. Since all physical channels of a cell are transmitted with perfect synchronisation on the downlink, a shorter midamble on the downlink is usually sufficient. Burst type 2 may only be used on the uplink when a maximum of four bursts per slot are being transmitted simultaneously.

A short guard period is sufficient for compensating for runtime differences when an active connection exists between mobile and base station. A longer guard period is necessary for random access and for handover. This is carried out in burst type 3 at the expense of the length of the second data field.

All three burst types can be used for dedicated data transmission over the DPCH and the PUSCH or *Packet Data Channel* (PDCH), although type 3 is reserved for the uplink. The P-CCPCH always uses type 1, the S-CCPCH type 1 or type 2. For the reasons mentioned above, the PRACH only uses burst type 3.

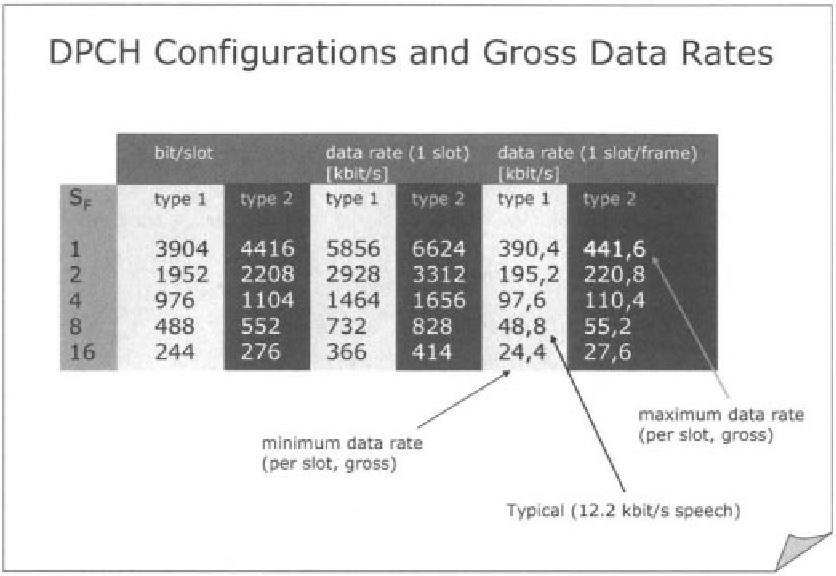
The PICH and the SCH use different time and burst structures that are not further explained here.

Figure 8.16 shows the transmission rates at the radio interface with the use of type 1 and type 2 bursts. The number of bits transmitted per time slot depends on the spreading factor as well as the size of the data fields in the respective bursts. A maximum bit rate of 6624 kbit/s can be achieved within a time slot; with one burst per time frame this produces an average gross bit rate of 441.6 kbit/s with a spreading factor of one. Higher transmission rates can only be realised through the use of multiple time slots in a time frame.

The standard describes mobile stations that can occupy a maximum of nine time slots on the uplink and a maximum of twelve on the downlink. The maximum achievable gross bit rate is therefore 3974.4 kbit/s on the uplink and 5299.2 kbit/s on the downlink. It is clear that a bearer service with a net bit rate of 2 Mbit/s can only be realised on the downlink and only with a code rate of more than 1/3. The same restrictions that apply for such services in FDD mode also apply in TDD mode. The radio propagation environment in particular must permit the use of a higher code rate.

The standard-compliant reference configuration for voice transmission with 12.2 kbit/s net provides for a spreading factor of eight; with burst type 1 this produces a gross bit rate of 48.8 kbit/s.

Variable transmission rates can be achieved in three ways in TDD mode: through selection of a spreading factor, through the parallel transmission of several physical channels in one time slot or through the parallel transmission of several physical channels in several time slots of a frame. Figure 8.17 shows how higher data rates can be achieved through multicode or multislot operations.



DPCH Configurations and Gross Data Rates

S_F	bit/slot type 1	bit/slot type 2	data rate (1 slot) [kbit/s] type 1	data rate (1 slot) [kbit/s] type 2	data rate (1 slot/frame) [kbit/s] type 1	data rate (1 slot/frame) [kbit/s] type 2
1	3904	4416	5856	6624	390,4	441,6
2	1952	2208	2928	3312	195,2	220,8
4	976	1104	1464	1656	97,6	110,4
8	488	552	732	828	48,8	55,2
16	244	276	366	414	24,4	27,6

Figure 8.16: Configurations and transmission rates of DPCH in TDD mode

Even in TDD mode information about the transport format or the current transport format combination can be exchanged between communicating physical layers over a corresponding indicator (TFCI). Whether and when a TFCI is transmitted is negotiated between the communicating partners. In addition, a power control command (TPC) is also transmitted on the uplink. This information is transmitted in the data fields of a burst. The same as in FDD mode, TFCI and TPC apply to all physical channels of a connection and therefore do not have to be transmitted in all physical channels.

In multicode transmission, TPC and TFCI are only transmitted in the data fields of a burst; in multslot transmission, they are only transmitted in the first burst that is transmitted within a time frame (see Figure 8.17). The TPC and TFCI information is always transmitted with a spreading factor of 16, irrespective of which spreading factor is being used within the data fields.

As shown in Figure 8.18, the different physical channels are multiplexed within a time slot before modulation. Every two consecutive bits of a physical channel are combined into a complex-valued symbol that is spread with the actual spreading code of the physical channel and then scrambled with a cell-specific scrambling code. Figure 8.18 (top) shows the multiplexing on the uplink of two DPCH belonging to the same CCTrCH. The weighting factor G is used to carry out power control and therefore has the same effect on both data streams. Afterwards each branch is subject to the weighting γ, which matches

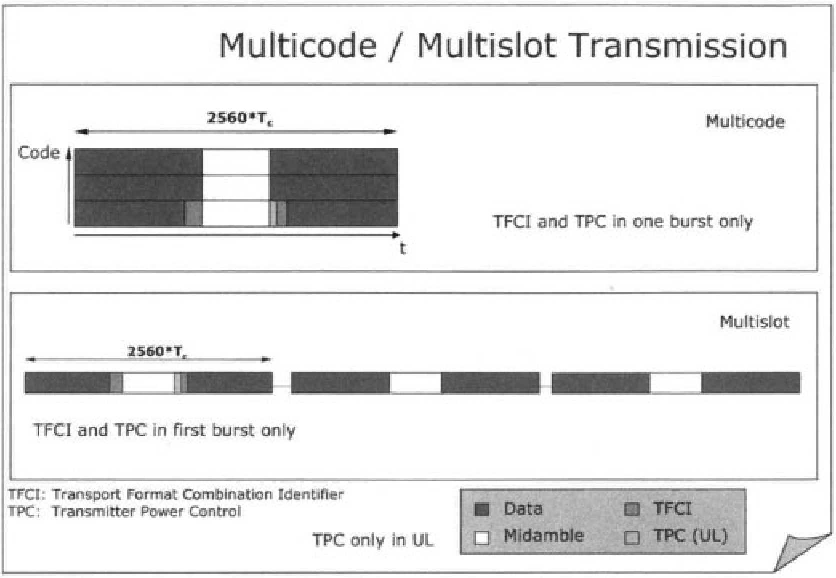


Figure 8.17: Multicode and multislot transmission in TDD mode

the power of the individual physical channels according to their spreading factors. The two complex-valued data streams are added and the sum is weighted with the factor β dependent on the actual transport format combination. The resulting signal is finally transmitted by means of QPSK modulation. If several CCTrCH exist on the uplink, then the same procedure is followed separately by the physical channels of the other CCTrCH.

Physical channels are multiplexed on the downlink in a similar way to on the uplink, with the difference that each physical channel is only weighted on its own. All weighted complex-valued chip streams are added and the sum transmitted through QPSK modulation.

8.4 Physical channels in TDD mode low chip rate option

All physical channels of the TDD mode but the SCH and the PNBSCH are existent in the low chip rate option as well although they are realised using a different timeslot, frame and burst structure. The following two additional physical channels are defined for the low chip rate option:

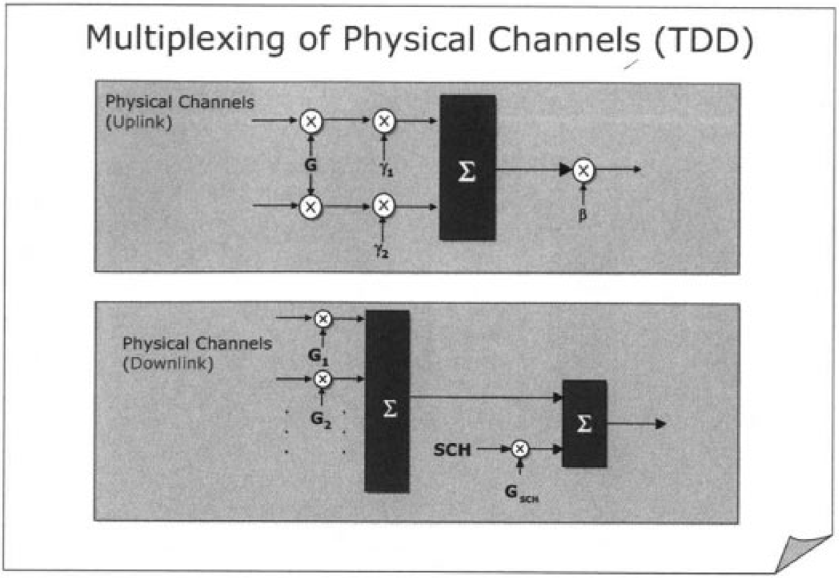


Figure 8.18: Multiplexing physical channels in TDD mode

- **Fast Physical Access Channel (FPACH):** The *Fast Physical Access Channel* (FPACH) is a downlink channel that is used to transmit the acknowledgement of a detected signature with timing and power level adjustment indication from the Node B to the UE. The FPACH uses spreading factor 16 with a single burst whose spreading code, training sequence and time slot position are configured by the network and signalled on the BCH.
- **Dedicated Physical Synchronisation Channels (DwPCH, UpPCH):** The *Downlink Physical Synchronisation Channel* (DwPCH) and the *Uplink Physical Synchronisation Channel* (UpPCH) are mapped to the DwPTS and the UpPTS, respectively, that exist in each sub-frame. They are used for downlink and uplink synchronisation. The synchronisation pattern sent in the DwPCH and UpPCH bursts are not spread. The DwPCH burst is transmitted at each sub-frame with an antenna which provides coverage for the whole cell.

Figure 8.16 shows the transmission rates at the radio interface for the low chip rate option. The number of bits transmitted per time slot depends on the spreading factor as well as the number of bits used for e.g. TFCI or TPC information included in the data part of the burst. Figure 8.16 shows the maximum bit rate that can be achieved without transmitting control information like TFCI or TPC. In the downlink a maximum gross data rate of

TDD low chip rate gross data rates

	Bits /Slot		data rate 1 slot/sub-frame [kbit/s]		data rate 6 slots/sub-frame [kbit/s]	
S_F	QPSK	8PSK	QPSK	8PSK	QPSK	8PSK
1	1408	2112	209.6	422.4	1257.6	2534.4
2	701	-	140.2	-	841.2	-
4	352	-	70.4	-	422.4	-
8	176	-	35.2	-	211.2	-
16	88	132	17.6	24.6	105.6	158.4

Note: in DL only S_F 1 or 16

Figure 8.19: Configurations and transmission rates of DPCH in low chip rate TDD mode

1257.6 kbit/s can be achieved with spreading factor 1 and the use of 6 time slots within each sub-frame. If, alternatively, the 8PSK Modulation is used, the maximum achievable transmission rate is 2534.4 kbit/s.

In the uplink direction the maximum data rate is 1257.6 kbit/s using 6 time slots per sub-frame with spreading factor 1 and QPSK modulation. With 8PSK modulation this maximum value increases to 253.4 kbit/s. It is clear that a bearer service with a net bit rate of 2 Mbit/s cannot be realised with one frequency channel at a time.

Following the standard-compliant reference configuration for voice transmission with 12.2 kbit/s net a two bursts with spreading factor 16 are used per sub-frame, both in the same time slot.

8.5 Mapping of transport channels to physical channels

This section takes another look at the mapping of transport channels to physical channels using the example of the transmission of voice data over dedicated channels in FDD and TDD mode. Figure 8.20 presents a schematic illustra-

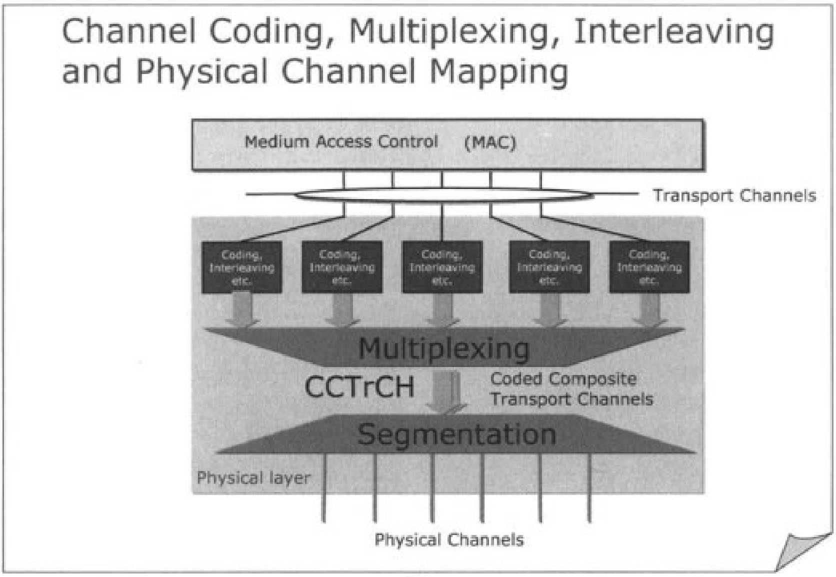


Figure 8.20: Structure of physical layer in UTRA protocol stack

tion of the structure of the physical layer in the UTRA protocol stack. As already explained, data blocks called transport blocks are transmitted over the transport channels. Each transport block passes through a chain of security and matching functions, such as channel coding, interleaving and puncturing. Data blocks that are subject to the same coding, interleaving and matching can be multiplexed to channels called *Coded Composite Transport Channel* (CCTrCH). A CCTrCH is then mapped through segmentation to one or more physical channels. A physical channel can therefore transmit the signalling and user data of different transport channels simultaneously.

Let us take another look at the example of the 12.2 kbit/s reference channel for voice data transmission, as shown in Figure 8.21. In the *Medium Access Control* (MAC) layer of the UTRA protocol stack the two logical channels *Dedicated Traffic Channel* (DTCH) and *Dedicated Control Channel* (DCCH) are mapped to two dedicated transport channels, called *Dedicated Channel* (DCH), for the purpose of transmitting user and signalling data. The mapping takes place in such a way that transport blocks each with 244 bits of user data are transmitted over the first transport channel every 20 ms and that transport blocks each the length of 100 bits are transmitted over the second transport channel every 40 ms. Channel coding and rate matching for both transport channels are the same. This means that they can be multiplexed to a CCTrCH.

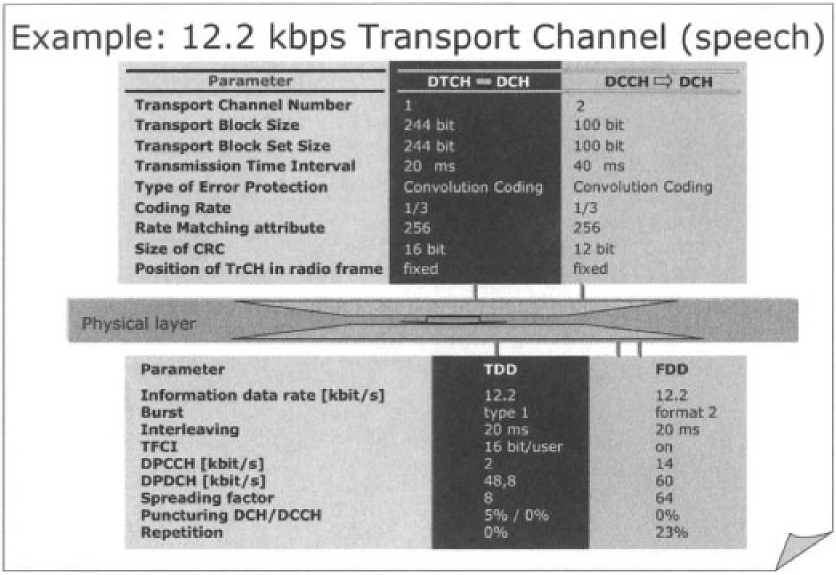


Figure 8.21: Mapping of 12.2 kbit/s reference channels to physical channels in FDD and TDD mode

No multicode or multislot transmission is required to transmit the data of the CCTrCH; one dedicated physical channel suffices. On the uplink in FDD mode this is a DPCH with a type 1 burst per time frame and a spreading factor 8, and in FDD mode this is a DPDCH with the spreading factor 64. Parallel to the DPDCH pilot bits, TFCI and power control information is transmitted over a DPCCH. In TDD mode this takes place within the data fields of the DPCH burst.

What is noticeable is that at 48.8 kbit/s the gross rate of the physical channel in TDD mode is, first, lower than that of the DPDCH in FDD mode and, second, also lower than required for the gross data after coding with rate 1/3. Therefore, 5% of the data ready for transmission has to be reduced through puncturing, thus through a systematic removal of bits. The gross bit rate of the physical channel in the FDD reference configuration is higher than the necessary bit rate. Therefore, through the repetition of bits around 23% more is transmitted than would be necessary on the CCTrCH.

Figure 8.22 shows how the transport blocks of both DCH transport channels are mapped to the dedicated physical DPDCH on the FDD uplink. The illustration shows a time frame of 40 ms. In this time frame two transport blocks each with 244 bits are transmitted over the first transport channel, one

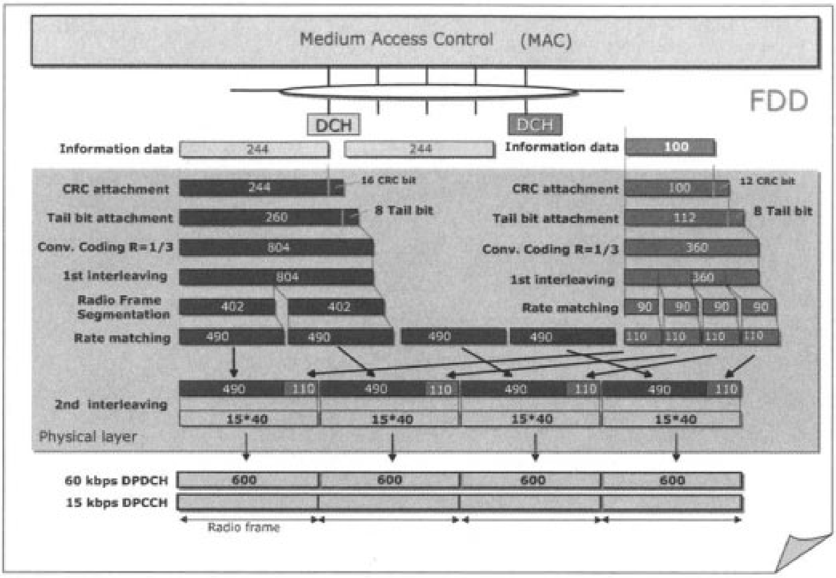


Figure 8.22: Mapping of two DCH to a DPDCH in FDD mode (uplink), 12.2 kbit/s reference channel

every 20 ms. The second DCH delivers a transport block the length of 100 bits every 40 ms.

A checksum is attached to each transport block for error detection. Since the channel coder requires data blocks of a specific length, the lengths of the data blocks are lengthened - if required - after the attachment of the checksum. The channel coding with the rate 1/3 adds redundancy for forward-error correction to the data stream, thereby increasing the data quantity by the factor 3. Afterwards the bits of the resulting data blocks are systematically interleaved within a block.

The coded data blocks are divided into equal-sized parts according to the transmission interval of the transport block. The 804 bits resulting from the 244 bit long transport blocks transmitted every 20 ms are divided into two blocks of 402 bits each. Of these blocks each one is transmitted within a 10 ms time frame. In a similar way the data block resulting from the 100 bits of the second transport block is divided up over four successive time frames.

As explained above, the transmission capacity of a DPDCH is larger than necessary according to the channel coding. The data quantity is therefore artificially increased through a repetition of bits before the data blocks are multiplexed serially to a CCTrCH. Over this CCTrCH 40 bits, or a total of

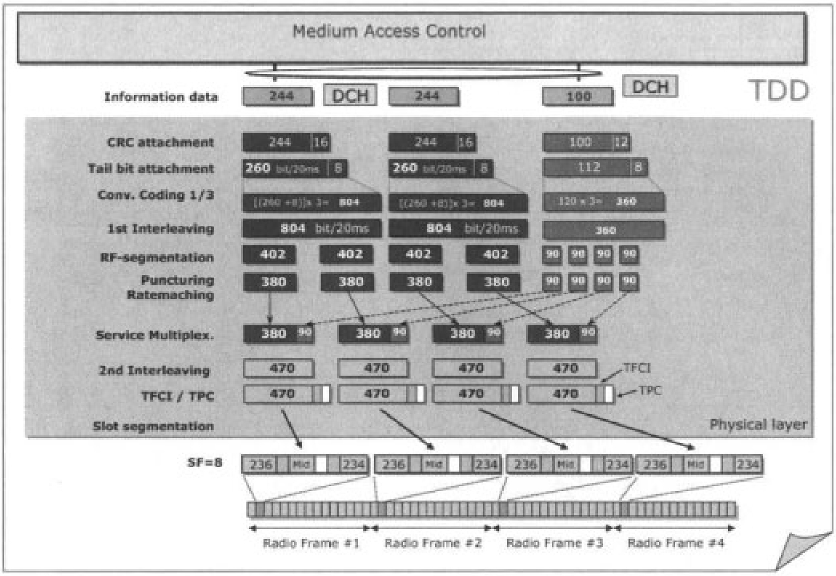


Figure 8.23: Mapping of two DCH to one DPCH in TDD mode (uplink), 12.2 kbit/s reference channel

600 bits per time frame, are transmitted in each time slot. The multiplexed data blocks are interleaved again before the spreading.

In TDD mode transport channels are mapped in the same way as they are in FDD mode, as shown in Figure 8.23. However, the characteristics of physical channels are not the same in FDD mode and in TDD mode. As explained earlier, the quantity of data for transmission within a time frame after channel coding and interleaving is greater than the physical DPCH permits. Therefore, after the first interleaving bits are systematically removed from each data block through puncturing. The punctured data blocks are serially multiplexed to a CCTrCH and after spreading they are transmitted within four time frames. At the same time a type 1 burst is sent per time frame.

In TDD low chip rate mode, channel coding, rate matching, frame size equalisation and interleaving are done in the same way as in the high chip rate mode. The main difference in the mapping of transport channels to physical channels in TDD low chip rate mode is the additional so-called sub-frame segmentation after the second interleaving. Since the radio frame in low chip rate TDD is subdivided into 2 sub-frames of 5 ms each, the basic mapping unit is a sub-frame. The rate matching procedure ensures that the data blocks of the CCTrCH have an even number of bits and can be segmented and mapped to two sub-frames.

Table 8.1: FDD uplink reference configuration

Net bit rate DCH for DTCH [kbit/s]	12.2	64	144	384
Net bit rate DCH for DCCH [kbit/s]	2.4	2.4	2.4	2.4
Gross bit rate DPDCH [kbit/s]	60	240	480	960
Gross bit rate DPCCH [kbit/s]	15	15	15	15
Spreading factor DPDCH	64	16	8	4
Rate matching DTCH [%]	+22	+19	+8	-18
Rate matching DCCH [%]	+22	+19	+9	-18
Interleaving depth	20	40	40	40
Number of DPDCHs	1	1	1	1

Table 8.1 presents the different reference channels for the uplink in FDD mode. Such reference configurations are used for the conformity tests of mobile and base stations and can be interpreted as examples of how different net transmission rates can be realised. Signalling data with a net rate of 2.4 kbit/s is always simultaneously transmitted with the user data over a transport channel. The transmission rate of a physical channel depends on the spreading factor. If a DPDCH is not sufficient, then several physical channels are used. Rate matching ensures that the data rate of the CCTrCH conforms with that of the physical channels.

8.6 Questions

8.1 What is a physical channel?

8.2 How is a physical channel characterised in FDD mode? In TDD mode?

8.3 What is the purpose of dedicated physical channels?

8.4 What are shared physical channels used for?

8.5 What is the difference between them?

8.6 How many bits are transmitted on the uplink (UL) per frame over the DPDCH with a spreading factor SF = 64?

8.7 How many bits are transmitted on the downlink (DL) per frame over the DPDCH with a spreading factor SF = 64?

8.8 Why is the bit rate of a dedicated channel on the FDD UL twice as high as on the FDD DL when the same spreading factor is the same?

8.9 What are the possibilities on the FDD uplink of achieving different transmission rates? Which conditions have to be maintained?

8.10 What are the possibilities on the FDD downlink of achieving different transmission rates? Which conditions have to be maintained?

8.11 Calculate the maximum transmission rate theoretically possible for a mobile station (MS) on the FDD uplink (DPDCH, gross).

8.12 Calculate the maximum transmission rate theoretically possible for a base station (BS) on the FDD downlink (DPDCH, gross).

8.13 What is the maximum amount of data that can be transmitted during random access? How high is the transmission rate?

8.14 How many users can share a time slot in FDD mode? How many in TDD mode?

8.15 Calculate the maximum transmission rate possible for a mobile station (MS) on the TDD uplink (DPDCH, gross).

8.16 Calculate the maximum transmission rate possible for a base station (BS) on the TDD downlink (DPDCH, gross).

9 Cellular CDMA Networks

9.1 Interference

The behaviour of cellular CDMA mobile radio networks differs considerably from that of cellular TDMA/FDMA networks. The reason for this is a basically different interference situation: In CDMA networks all users utilise the same frequency channel. Consequently, the resulting co-channel interference is comparatively high whereas the carrier-to-interference ratios are appropriately low. Added to this is that co-channel interference in TDMA/FDMA networks always originates in other cells; in CDMA networks all users of the same cell also cause interference to one another (see Chapter 6).

Figure 9.1 shows how co-channel interference originates on the uplink in CDMA networks. Interference power caused by users of the same cell is called intracell interference. The interference power resulting from connections in co-channel cells is called intercell interference. CDMA networks typically have the cluster size one, which means that all senders in the network are causing mutual interference. Due to the close proximity of intracell interferers, the intracell interference is usually greater than the intercell interference. Consequently, many of the effects noticeable in cellular CDMA networks are essentially caused by intracell interference.

Intracell co-channel interference does not occur in TDMA/FDMA networks, which is a key reason for the different behaviour between TDMA/FDMA and CDMA networks.

9.2 Cell breathing

A specified, service-specific C/I ratio must be maintained at the receiver to guarantee a sufficient quality of service. A growth of traffic in a network automatically leads to an increase in interference power – intracell as well as intercell interference. The coverage range, i.e., maximum achievable distance between base station and mobile station is determined by the maximum power of the transmitter. This transmitter power must be sufficient to guarantee the necessary C/I ratio at the receiver despite the signal attenuation that occurs with radio propagation. The higher the interference power is, the less the signal may be attenuated during radio propagation with the same transmitter power, i.e., an increase in interference power means a reduction

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff
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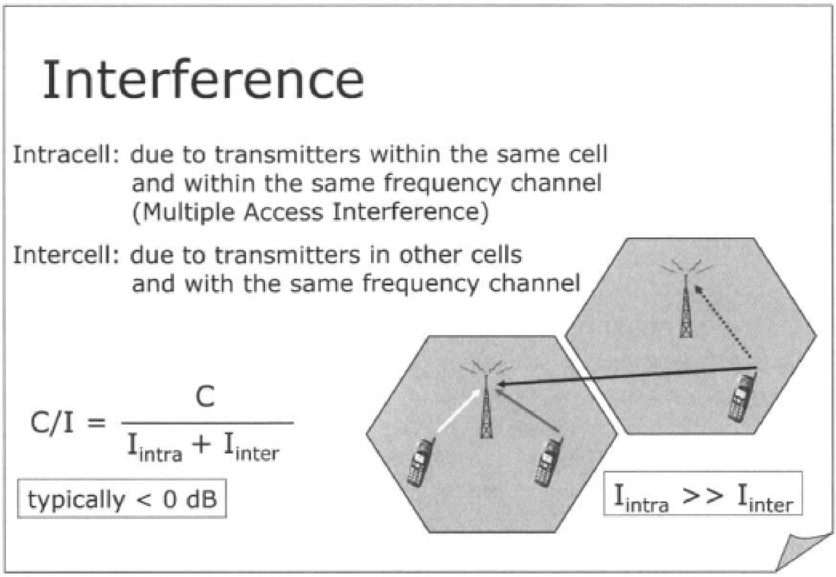

Figure 9.1: Intercell and intracell interference in cellular CDMA networks

in the maximum coverage range. The load-dependent change in range is also referred to as *cell breathing.*

Figure 9.2 shows the effect of cell breathing. In cellular CDMA networks intracell and intercell interference rises as traffic increases and the connection quality often degrades. Since this process – increase in traffic and interference – occurs slowly and not suddenly in real networks, one refers to a ***graceful degradation*** of quality of service for all users. If the traffic continues to increase, then the interference power does as well. What can occur if this process continues is that the available transmitter power of a transmitter is not sufficient to comply with the required C/I at the receiver due to the high radio field attenuation; communication is then no longer possible. Due to the high level of interference power, the only connections that can be maintained are those where the transmitters are able to guarantee the required C/I because of a closer proximity to the receiver and thus lower propagation attenuation. The useable area of a cell thus shrinks as traffic increases in a network.

Since the traffic load in a network typically changes during the course of day, the cells of a CDMA network grow and shrink depending on the time of day. This change of cell size is called ***breathing.*** This breathing is essentially determined by the intracell interference. The intracell interference is in turn determined by the orthogonality of the spreading codes and the quality of the receiver. In UMTS FDD mode the spreading codes on the uplink are only

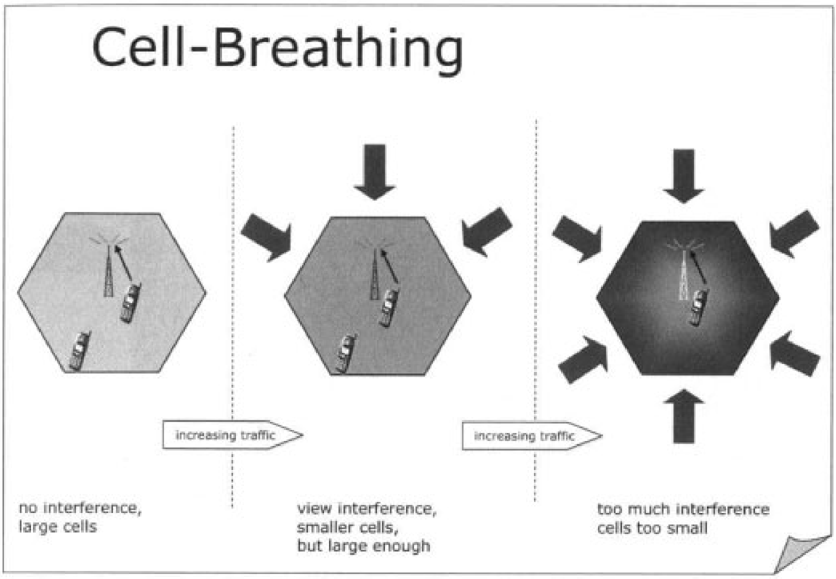

Figure 9.2: The effect of cell breathing

quasi-orthogonal; they are orthogonal on the downlink. In UMTS TDD mode orthogonal codes are used both on the uplink and on the downlink. The use of joint detection may be required with the orthogonal codes. Cell breathing is therefore particularly prevalent on the uplink of UMTS FDD mode.

What plays a role in cell size is the attenuation that a signal can experience in radio propagation without the C/I ratio at the receiver falling below the required value, even in the case of maximum transmitter power. This maximum allowable propagation attenuation is also called link budget. Since the propagation attenuation is proportional to the propagation distance, the link budget ultimately determines the coverage range and thus the size of a cell.

Figure 9.3 shows how the link budget is calculated for the 8 kbit/s voice service on the uplink in UMTS FDD mode [10]. With a maximum power of 21 dBm, the transmitter of the mobile station must be able to achieve the required C/I ratio of -20.9 dB at the receiver. The noise power is made up partly of thermal noise and partly of intracell and intercell interference. With a total interference of -97 dBm the user signal must therefore be received with at least the power of -117.9 dBm. Together with the antenna gain of mobile and base station of a total, for example, of 11 dB, the maximum allowable propagation attenuation is 149.9 dB.

It is clear from Figure 9.3 that the total interference power is directly entered as a summand in the calculation of the link budget (calculation in decibels).

Link Budget

(Example FDD UL, speech, 8 kbit/s)

UE EIRP [dBm]	21	
Receiver Noise Floor [dBm]	-103	
Interference Power [dBm]	-98.26	
Total (Noise + Interference) [dBm]	-97	=f(n)
Required C/I [dB]	-20.9	
Receiver Sensitivity [dBm]	-117.9	
Gains and Losses at the BS [dB]	11	
Maximum Pathloss [dB]	149.9	
Fading Margin [dB]	11.3	
Handover Gain [dB]	3	
Link Budget [dB]	140.7	

Figure 9.3: Sample link budget calculation

The effect of cell breathing is evident here: a higher interference power reduces the link budget, i.e., the coverage range becomes smaller.

The link budget for the downlink is calculated in a similar way; however, on the downlink all connections have to share the maximum transmitter power of the base station. On the downlink the transmitter power available per connection therefore depends on the interference power as well as on the number of connections per cell.

The breathing of radio cells therefore is different on the uplink and on the downlink. Figure 9.4 shows the link budget as a function of traffic per cell, i.e., the number of connections per cell [10]. It is noticeable that the link budget reduces as the traffic increases due to the increase in interference power.

When the traffic is low, the link budget is considerably lower on the uplink than on the downlink. The cell size is therefore determined by the uplink. When the traffic volume increases, then the link budget for the downlink falls more quickly than for the uplink since more and more connections have to share the transmitter power of the base station on the downlink and the interference power increases at the same time. As of a certain volume of traffic, the link budget, and consequently the cell size, is no longer determined by the uplink but instead by the downlink. This has to be taken into account in the planning of UMTS radio networks.

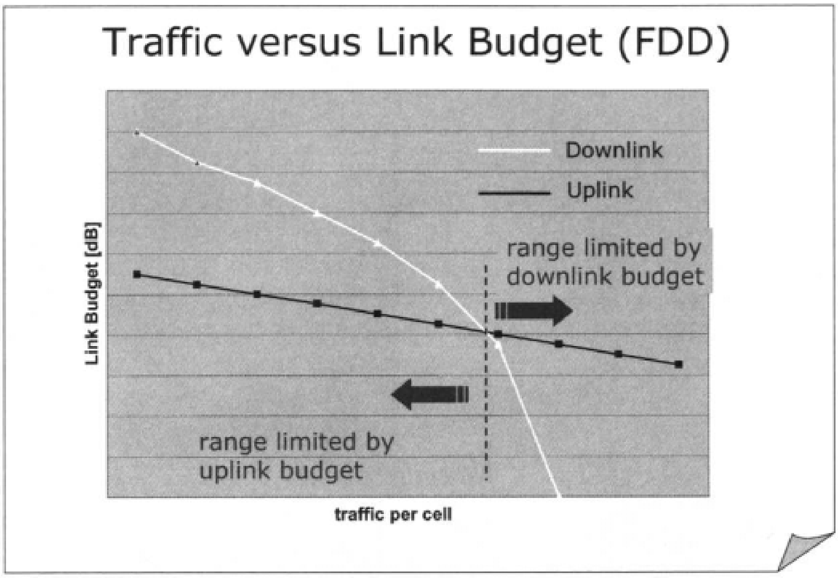

Figure 9.4: The effect of cell breathing on the link budget

9.3 Traffic capacity in cellular CDMA networks

Since the size of a cell depends on the traffic in the network, the location planning for a CDMA network is always only optimal for a certain volume of traffic. When traffic volumes are low, large areas of overlapping occur; with very low traffic volumes the cells can shrink to such an extent that gaps occur in the radio coverage. Furthermore, the cells *breathe* differently for services that require different C/I ratios. The planning of a CDMA network must therefore not only take into account a certain traffic volume but also a certain service or service mix and the respective quality of service (see Figure 9.5).

As explained in Chapter 6, the gross transmission rate is reverse proportional to the spreading factor. The required carrier-to-interference ratio at the receiver therefore increases the higher the transmission rate of the respective service. A higher required C/I ratio also means a smaller maximum range of radio coverage. Radio coverage for services with a high spreading factor is therefore higher than for services with a low spreading factor. Depending on the network planning, such high-rate services can only be used in the proximity of the base station, whereas services with a lower transmission rate, such as voice services, are available in the entire cell.

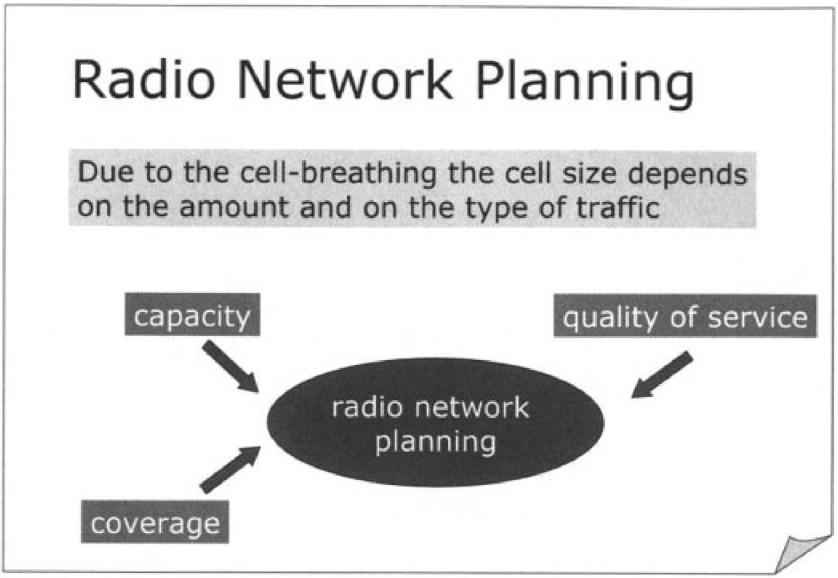

Figure 9.5: Influences on radio network planning

The capacity of a cell in a CDMA network essentially depends on the orthogonality and number of spreading codes used. Perfectly orthogonal codes guarantee that the different physical channels do not cause mutual interference. In this case, the traffic capacity of a CDMA cell is only determined by the number of mutually orthogonal codes. In normal cases where only a scrambling code is used on the downlink in UMTS FDD mode, the theoretical capacity is determined by the number of available OVSF spreading codes. There are exactly four spreading codes with the spreading factor four that are orthogonal to one another.

If the spreading codes are only quasi-orthogonal, as is the case on the uplink of UMTS FDD mode, then cell capacity is determined by the interference this creates. Figure 9.6 clarifies the principle of calculating the capacity of a CDMA cell for the uplink. The transmitters of the mobile stations are controlled by the power control in such a way that in ideal circumstances the signals received at the base station all have the same power, thus preventing near-far effects. Of n transmitters in a cell, one of them thus always supplies the wanted signal, the remaining $n-1$ contribute to the interference power. As Figure 9.6 shows, the maximum user capacity per cell therefore now only depends on the minimum required C/I ratio.

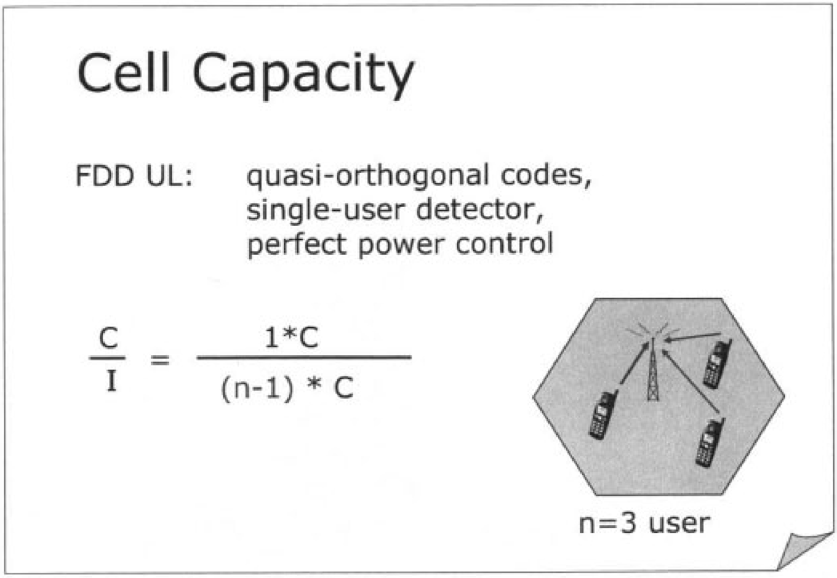

Figure 9.6: Simplified calculation of traffic capacity without intercell interference

In CDMA networks the interference power is a factor that determines the network capacity. CDMA networks are therefore also referred to as being *interference-limited* (see Chapter 2).

The connection between user numbers and C/I is illustrated in Figure 9.7. The C/I at the receiver is spread over the number n of active senders in a cell. The more senders that are active in a cell, the higher is the interference power at the receiver and the smaller the C/I ratio.

The C/I ratio required depends on various factors. What is particularly important for determining the required C/I ratio are the spreading factor, the channel coding and modulation as well as the tolerable residual bit error ratio. It is obvious from Figure 9.7 that with a required C/I of -20 dB around 100 users can be active in an individual cell or that with 100 users only a C/I ratio of a maximum of -20 dB can be achieved.

The calculation of cell capacity in a multicell mobile radio network has to take into account the intracell interference as well as the intercell interference and thermal noise (see Figure 9.8). In UMTS the spreading codes of different cells are only quasi-orthogonal to one another. All senders in co-channel cells thus produce intercell interference. The interference shown on the uplink in Figure 9.8 originates in the transmitters of the mobile stations. In a multi-cell situation the equation given can no longer simply be resolved based on the number n of active participants, because the intercell interference also

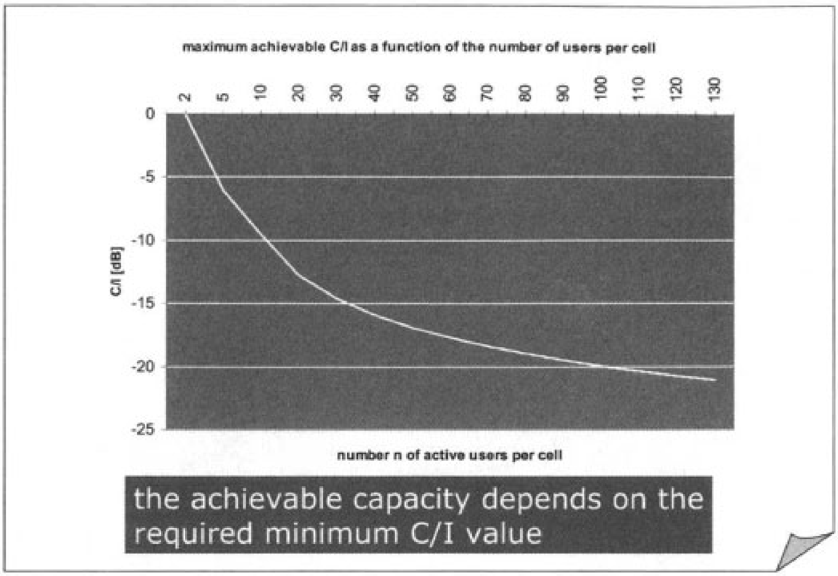

Figure 9.7: Carrier-to-interference ratio based on number of active users per cell

depends on the number of participants in the co-channel cells. However, the cell capacity can also be calculated for a multicell situation. It is easy to see that the cell capacity for a single-cell network is always larger than the cell capacity for a multicell one. Due to thermal noise and intercell interference, the cell capacity in multicell networks is only about half that of single-cell networks that have no intercell interference or thermal noise.

9.4 Soft handover

A handover, the forwarding of a connection from a cell to a neighbouring cell, requires a change of physical channel over which the communication is implemented. Handovers in UMTS can be implemented as hard handovers, soft handovers or softer handovers. Soft handovers and softer handovers are only used in UMTS FDD mode.

A mobile station that approaches a cell boundary can pass into a soft handover. This means that it is communicating simultaneously with up to three base stations. Each base station transmits the same information on its own physical channel. The mobile station must therefore be able to decode up to three physical channels of different base stations and merge the data streams. Each base station participating in the soft handover decodes the physical up-

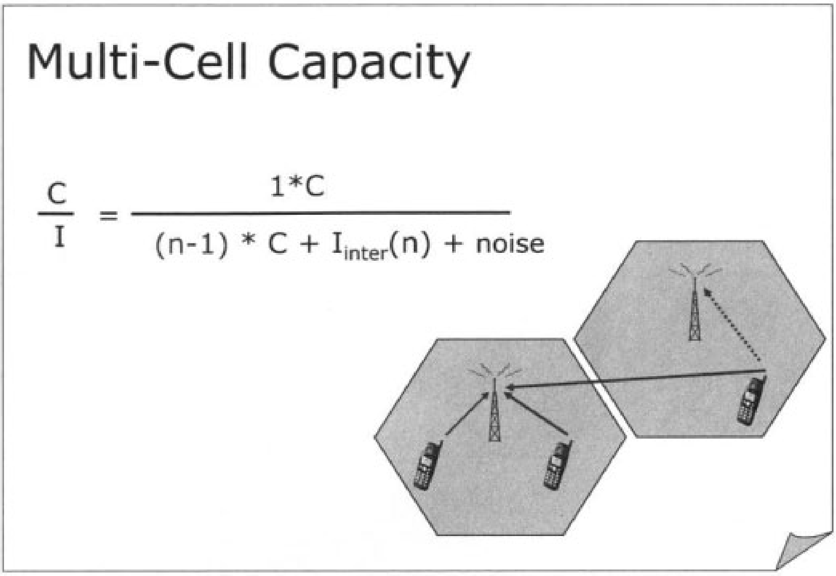

Figure 9.8: Calculation of traffic capacity in a multicell CDMA network

link channel of the mobile station on the uplink. Through the soft handover a mobile station maintains connections with several base stations during the handover phase; there is no sudden changeover of physical channel as is familiar from GSM. The handover is therefore referred to as a *soft* handover. If the mobile station does not change cells but only the sector of a cell during handover, the handover is referred to as a softer handover. In principle a softer handover functions the same as a soft handover (see Figure 4.13).

The hard handover is the one used in UMTS TDD mode. The physical channels of a connection become hard, i.e., changed from one time frame to another.

The simultaneous reception of different physical channels in a soft handover results in additional computation for a mobile station in a CDMA network. If the physical channels also had a different frequency channel, the calculation effort, especially in the hardware, would be incomparably higher. Therefore, TDMA/FDMA systems only use the hard handover. Inter-frequency handovers, which require a change of frequency channel, are also implemented by a hard handover in UMTS FDD mode, just like the UMTS handover according to GSM.

An active mobile station constantly measures different physical parameters, such as the received signal strength of the pilot channels of neighbouring cells. The results of such measurements are transmitted to the RNC and used there

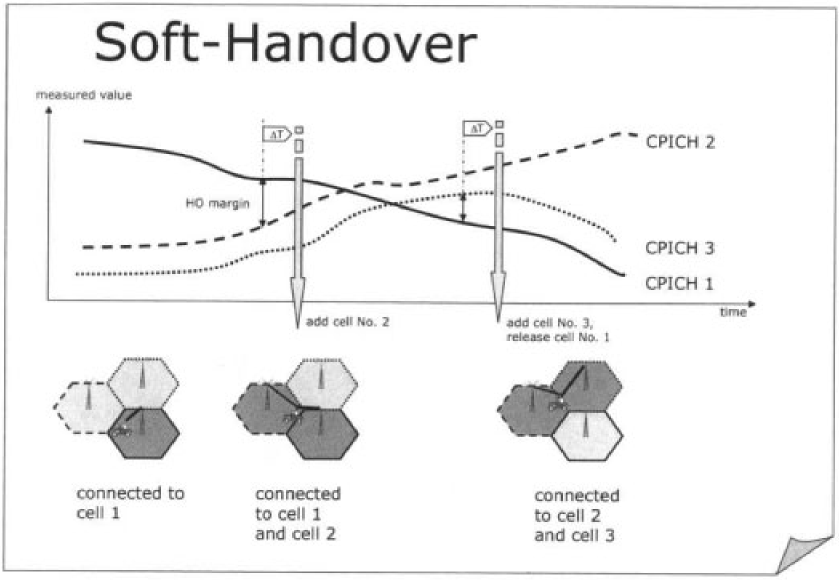

Figure 9.9: Functioning of a soft handover

to control the soft handover. Figure 9.9 shows an example of a soft handover based on the characteristics of the received signal strength measured by a mobile station for the pilot signals of three cells.

The mobile station first only communicates with base station 1. If the mobile station starts to move in the direction of base station 2, then the received signal strength for the CPICH of base station 2 increases. If the received signal strengths for CPICH 1 and CPICH 2 differ by a maximum of an amount called the ***handover margin*** during the period ΔT, a connection is also established to the second base station. The mobile station thus finds itself in a soft handover. The connection to the first base station is then cleared when the received signal strength of CPICH 1 is smaller by a certain amount than that of CPICH 2 or if reception by another base station is considerably better. In the latter case, the connection to base station 1 is cleared and one established to base station 3 (see Figure 9.9).

Due to the effects of shadowing, a mobile station on a cell boundary will receive different base stations with varying signal strength at short time intervals. The margins shown in Figure 9.9 that cause a type of hysteresis effect during cell changing are used to prevent so-called ***pingpong*** handovers in which a frequent switching back and forth occurs between different cells.

Although soft handovers are associated with additional complexity in terms of computing and organisation for the mobile station and in the radio access

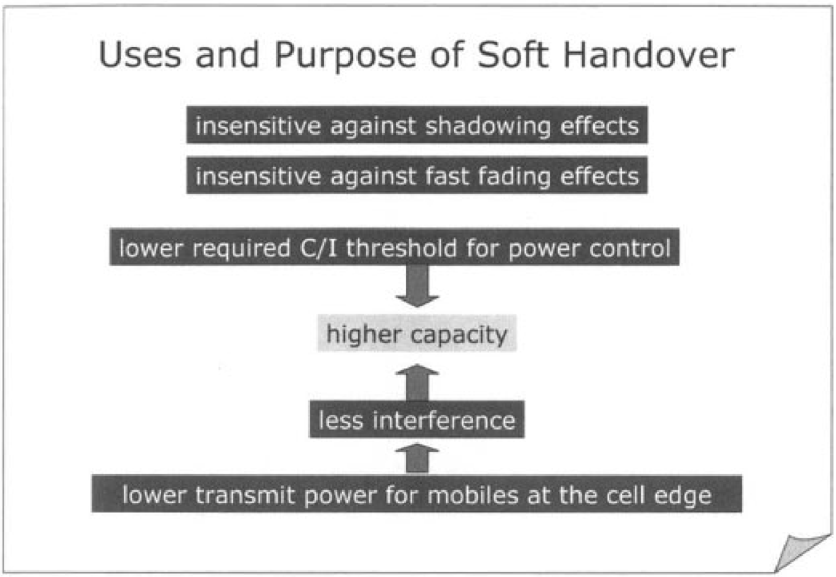

Figure 9.10: Uses and purpose of soft handovers

network, they represent an important mechanism for the capacity of cellular CDMA networks (see Figure 9.10). When a mobile station is provided with radio coverage by several base stations, there is macro diversity that reduces the effects of slow fading as well as fast fading.

Since shadowing is generally only correlated weakly to different base stations, macro diversity increases the probability that a sufficiently strong connection can be maintained to at least one base station. The fast fading on the connections to different base stations is also basically uncorrelated. It is this lower sensitivity compared to fast fading that can be used to reduce the required carrier-to-interference ratio that is maintained through power control. The reason is that a reduction is possible in the protection margin contained in the link budget calculation for fast fading.

During a soft handover the mobile station receives power control commands from all base stations involved in the handover. For communication it suffices if only one base station receives the signals of the mobile station with sufficient strength. The mobile station therefore adjusts its transmitter power downward so long until it receives a command to reduce its transmitter power from at least one base station. If the transmitter power were only controlled by one base station, there would be the danger of a mobile station finding itself in the radio coverage area of a neighbouring cell, but not being controlled by it. As explained earlier, precise power control is necessary to counter near-far effects. This means that the base station to which the lowest possible propagation attenuation exists should be the one to control the mobile station.

To summarise what was explained earlier: a low required C/I ratio and a reduction in interference power through precise power control equate to increased capacity in a radio network.

9.5 Questions

The values presented in Figure 9.3 for calculating a sample link budget for the 8 kbit/s voice service on the FDD uplink should be used as the basis for your responses to the following questions. Other examples of coverage range calculations can be found in [10].

9.1 Why is strong co-channel interference characteristic of CDMA systems?

9.2 Why is efficient communication possible despite the strong co-channel interference?

9.3 What is the maximum range of an FDD cell on the uplink? The path loss (PL) is calculated on the basis of $\mathrm{PL} = 135 + 35.2 \cdot \log_{10}(R/[\mathrm{km}])$.

9.4 An interference power of -98 dBm (without noise) exists for the maximum load established by the network operator. How large can the network operator plan its cells?

9.5 What size does a cell shrink to if the interference increases by 3 dB?

9.6 A GSM network operator wants to equip its GSM locations additionally with UTRA FDD base stations. How large are the GSM/UTRA cells? The link budget for the GSM cells is 154 dB.

9.7 How high can the interference power be on the FDD UL so that the same cell size can be achieved as with a GSM system?

9.8 Explain why on the FDD UL it is the transmission direction that determines cell size when the usage is low. Why does the downlink determine the coverage range when usage increases?

9.9 What is the difference between a soft handover and a softer handover?

9.10 Which handover is typically an inter-controller or an intra-controller handover?

9.11 Why are soft handovers necessary in CDMA networks? What role does the near-far effect play in this connection with a soft handover?

9.12 How is a soft handover typically implemented on the uplink? On the downlink? Briefly explain how it functions.

9.13 What determines the number of base stations involved in a soft handover?

9.14 Allocate soft and hard handovers on the basis of implementability:

(a) UTRA FDD - UTRA - FDD, *intra-cell, intra-frequency*

(b) UTRA FDD - UTRA - FDD, *inter-cell, intra-frequency*

(c) UTRA FDD - UTRA - FDD, *inter-cell, inter-frequency*

(d) UTRA FDD - GSM

(e) UTRA FDD - UTRA TDD

(f) UTRA TDD - GSM

9.15 What are the problems that occur when a hard handover is implemented in UTRA-FDD?

10 Service Architectures and Services in UMTS

10.1 Virtual Home Environment (VHE)

The introduction of UMTS will make many new services possible. User acceptance of these services is vital; therefore, the services should be available at all times and work everywhere. Furthermore, they should be easy to use. The following scenario could be an example.

A customer may be using different terminals for different situations: a notebook with a built-in UMTS terminal during the day and an integrated system in the car. The customer should be able to access all services personalised by him with each of his terminals. The representation of the services has to be adapted to the possibilities of the respective terminal, but their general use and operation must be independent of the terminal. However, current surveys indicate that customers are only using a small proportion of the options offered by their terminals because they are unable to deal with the complexity of the technology involved (i.e. MMS composition). Each change of terminal is currently associated with a new learning phase, which reduces the level of acceptance and thereby service usage.

With UMTS, many network operators will be introducing their own new services that may not even be standardised. When the user changes to another *serving network* (see Section 4.2), he should be able to continue using the services he is familiar with from his home network. This could include personalised information or information about certain other users.

A new concept, called *Virtual Home Environment* (VHE), is being used to meet these requirements. VHE refers to the portability of personalised services across network borders and between different terminals. It enables users to "know" their services and continue to use the knowledge they have acquired about how to operate "their" services. This spawns a higher customer acceptance and an increase in the use of services. A user should always have the feeling that he or she is working in a familiar environment [26, 28].

The logical concept behind this approach is called *Personal Service Environment* (PSE). At the centre is the user who makes a personal selection of the services available. These services are then always available to the user - with all adaptations that were made (see Figure 10.2).

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff
 ISBN: 0-470-84557-0

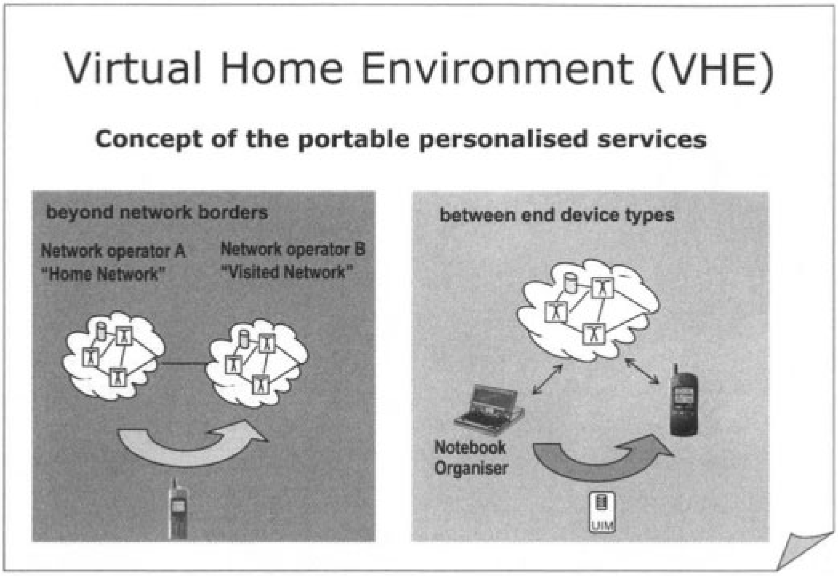

Figure 10.1: Concept of the Virtual Home Environment (VHE)

The selection of services is filed in the user profile of the particular user. In addition to general data, the user profile can also include personal interests and similar data. For example, every day at 17:00 hours on normal working days the user wants an overview of the traffic situation in the city in order to plan his trip home from work. The subscribed services could be managed on a situation-dependent basis in the user profile.

When executed, the services access the user profile and adapt individually to the customer's requirements. The services are stored and executed in a so-called *home environment*, an infrastructure for the provision of these services. This logical structure does not have to be limited to the fixed network infrastructure and can also include other parts, such as the UE or the USIM.

Although a network operator can set up a 3G network, but does not necessarily have all information requested by his users at his disposal, he can enter into agreements with third parties, called *Value-Added Service Providers*, and offer them interfaces to his execution environments. These providers can then make their own content available to users over these interfaces. A bank that provides its customers with access to their accounts or deposits over a *Personal Service Environment* (PSE) would be a provider of value-added services in this case. Since access to the personal service environment is integrated, a customer can carry out financial transactions with his or her bank from any location and with any terminal.

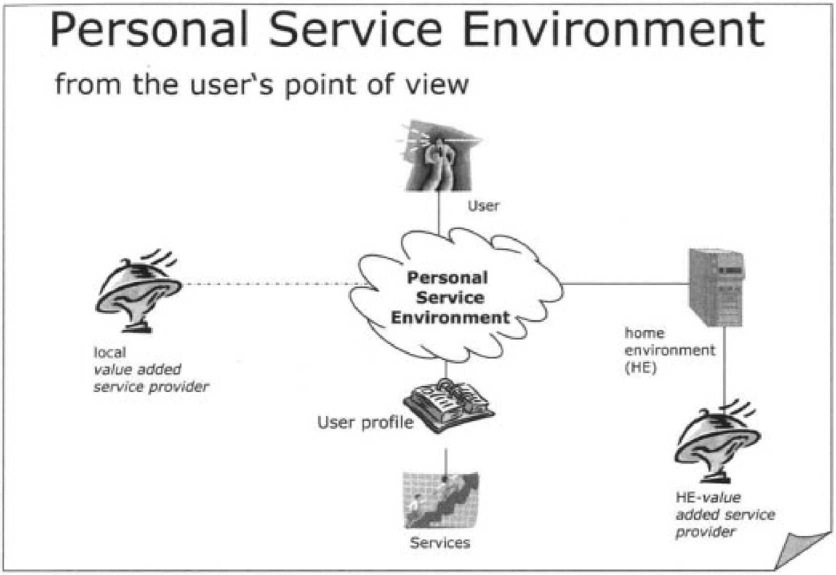

Figure 10.2: Personal service environment

The interaction of these personalised services and information about the requirements of a user is called the *Personal Service Environment* (PSE). Furthermore, users can use information from independent service providers at any time. The network operator has no influence on these services since the provider of the services is not in a contractual agreement with the network operator. As a result, these services are normally not personalised by the PSE.

The basic architecture shown in Figure 10.3 is used in the provision of personalised services:

- User-specific information is stored in a data storage unit DAT. This can be an account number or the destination call number for situation-dependent call forwarding.
- The code for the program providing the service is filed in service programme storage PRG.
- At runtime the program is loaded together with the data into the execution environment EXE. This execution environment has well-defined interfaces for communication with the outside world. Special commands are usually available for reloading other data from data and program storage or for communicating with the user via the *Man-Machine-*

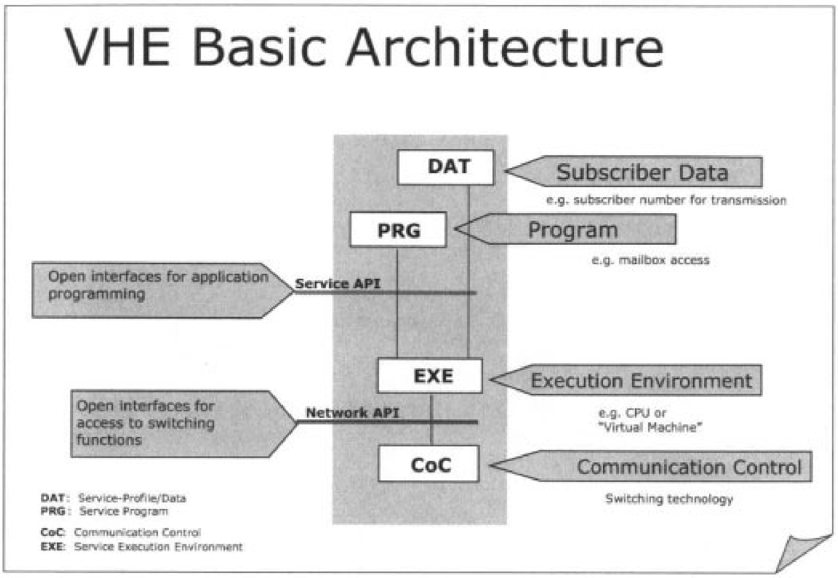

Figure 10.3: Virtual Home Environment: basic architecture

Interface (MMI). These commands are combined in the *Service API* that programs can access during runtime.

- The actual communication routines are contained in the communication control block (CoC). This refers to the control of the communications functions of the node on which the program is running. The program can control the communication at runtime over an open interface, called the *Network API*. For example, the user selects the service "Selective availability". The relevant program is loaded and started in the execution environment. Depending on the time of day, it forwards incoming calls to the user's mobile number, office number or private telephone number. These numbers are filed in the data storage unit DAT and are queried via the Service API. The Network API, which provides the call forwarding function, handles the programming of the call forwarding.

Due to the open interfaces and the defined execution environment, the programmer of the application can develop the service independent of the actual hardware and software running on the node.

This basic architecture is applied to all components involved in the service execution (see Figure 10.4). It is found in the home network, in the visited network, in the terminal and in the USIM. The RAN is mainly transparent

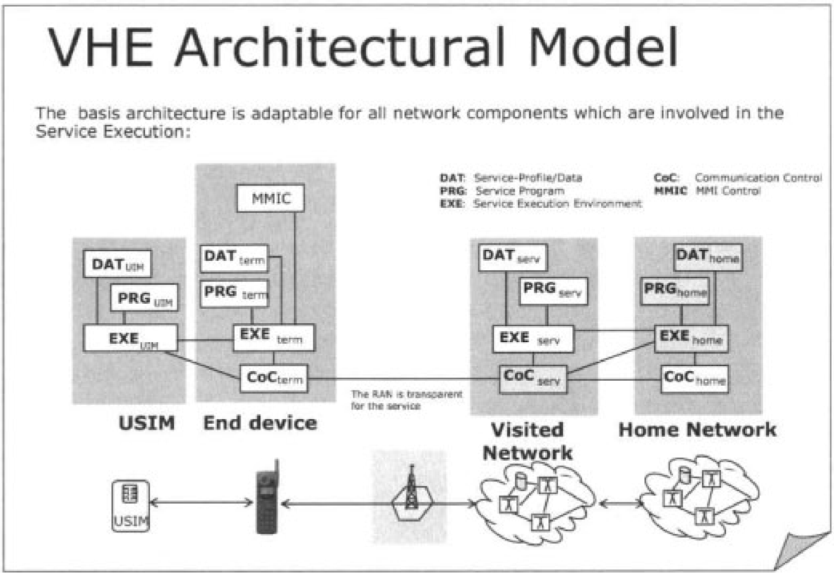

Figure 10.4: Virtual Home Environment: architectural model

for the provision of these higher services (commands exist at the Iu-Interface to determine the position of a mobile station for location-based services).

Network nodes in the home network and in the visited network can contain data and program storage as well as an execution environment for programs. The architecture for program execution in a home network is described below under the keyword OSA. Services subscribed to in the home network can be loaded into network nodes in the visited network using the *Customised Application of Mobile Enhanced Logic* (CAMEL) concept. This enables the user to have access to his or her personalised *Intelligent Network* (IN)-services irrespective of the network currently being used [31, 32]. CAMEL is already used in current GSM implementations, i.e. for the provision of roaming to prepaid users.

A terminal can also be used to provide services. In addition to an execution environment and storage for programs and data, it offers communication control functions and a user interface for data input and output. Services can even be provided over the USIM as well. New SIM cards are able to dynamically download Java programs from the network *over-the-air* and execute them within their CPU. These programs can also be stored on the SIM card by the network operator before the card is issued to the customer. For data input and output and for communication control, the USIM uses a defined interface to access the functions of the terminal (*SIM Application Toolkit* (SAT)).

Some card vendors are already offering full-value JAVA cards with 32 or even 64 KBytes of usable memory.

Current GSM networks are already partially supporting program execution on SIM cards and in terminals (see Figure 10.5). With WAP, programs can be transferred from the network to a terminal, then become executed within the WAP browser in the terminal and enable access to personalised services.

Although WAP has not been a real commercial success so far, the underlying architecture is open and flexible. The main reason why WAP has not been successful is attributed to the fact that it was promoted as providing full mobile access to the Internet and raised high expectations that the technology has not yet been able to fulfil. Moreover, the circuit-switched WAP access offered when WAP was introduced makes little sense because of the time-based billing. Finally, because the conent representation is incompatible with HTML, less content is available than with the Internet.

Some new services like *Multimedia Message Service* (MMS), which is seen by many as the worthy successor to SMS, are based on the WAP protocols *Wireless Transaction Protocol* (WTP) and *Wireless Session Protocol* (WSP). The MMS services currently offered are based on GPRS, but as UMTS becomes available, the new bearer services can be used as well.

10.2 Mobile Station Application Execution Environment (MExE)

MExE (see Figure 10.6) offers a new, well-defined execution environment for programs within the terminals. The programs are downloaded from servers in the network and then run in the terminal. Communication with the network is not required. MExE provides a number of security functions to ensure that applications are certified and secure.

Through MExE a terminal is transformed from a pure telephone to an intelligent device that can provide the user with all sorts of programs. The terminal thus evolves from a pure communication device and gains flexibility. In addition to telecommunications, the terminal will then also support games, home banking, PDA functions and music. For MExE a terminal must provide sufficient computing capabilities and memory. Some vendors of terminals have already announced that the next generation of GSM telephones can use up to 64 MB of memory, and some GSM telephones are already supporting storage modules based on flash memory. Through MExE these devices will be compatible with one another and it will be possible for applications to be run in every MExE-compliant terminal.

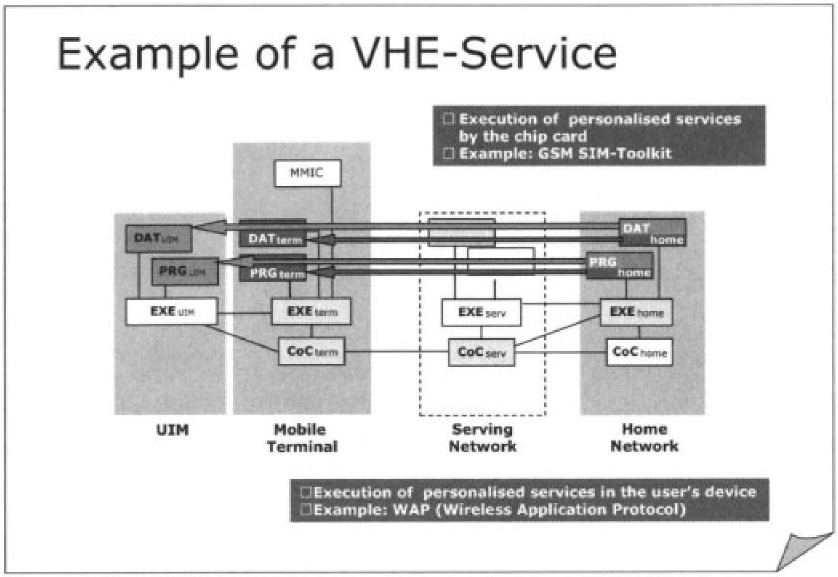

Figure 10.5: Example of a VHE service

MExE can currently support two types of programmes: Java and WAP applications. This list may be expanded later to include other languages. MExe programs belong to one of currently three classmarks:

- MExE classmark 1 - WAP Environment: Applications based on this MExE classmark use mandatory features of the WAP standard and can run on nearly any phone. Since WAP phones usually have limited display capabilities, these applications often have a limited user interface.
- MExE classmark 2 - Personal Java: This environment is based on the Java 2 Standard Edition and provides the applications with a full-scale Java environment. The runtime environment is extended by a software API to support telephony control, messaging, address book modifications, etc. Classmark 2 seems to provide the most powerful runtime environment.
- MExe classmark 3 - Java 2 Micro Edition: The third classmark uses the special Java version J2ME developed for small and portable devices. Again, special APIs to access telephony functions have been added.

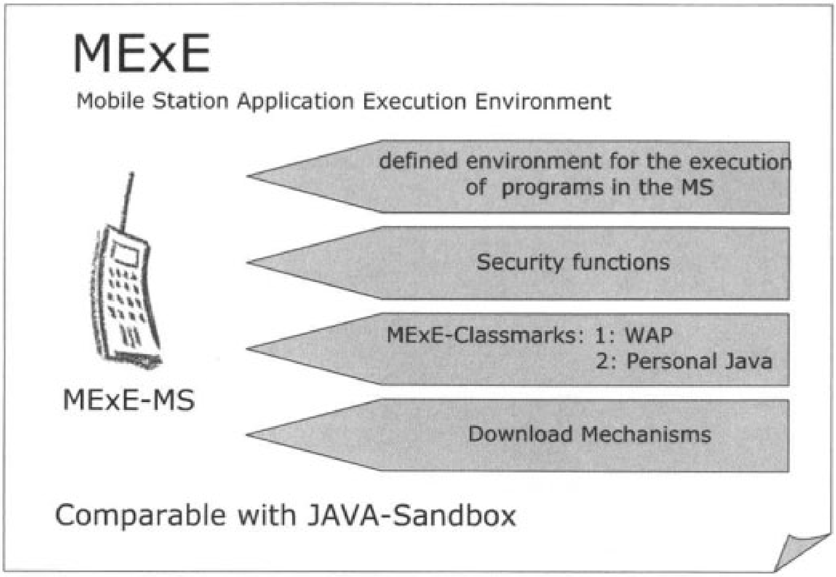

Figure 10.6: Mobile Station Application Execution Environment (MExE)

10.3 SIM Application Toolkit (SAT)

Plans for UMTS also include extending the *SIM Application Toolkit* (SAT) familiar from GSM to an *UMTS SIM Application Toolkit* (USAT) that will help to provide services similar to those offered by MExE. The advantage of USAT is the contractual relationship between network operator and customer (see Figure 10.7). The network operator retains ownership of the SIM card and therefore has more control over the execution environment than is the case with MExE. Thus the network operator only authorises applications it has requested on a card and which he can use to earn money.

Since a SIM card does not have its own user interfaces, there is a set of commands it can use to output information on the display of a terminal and that informs it of keyboard entries. However, the options compared to MExE are limited because a SIM card has to function with all terminals.

A direct comparison between USAT and MExE reveals pros and cons to both approaches. On the positive side, USAT offers a secure environment and stronger control from the view of the network operator. The advantages of MExE is that the processor in the execution environment is usually more powerful and applications can be larger. Both approaches allow new applications to be implemented transparently for the user.

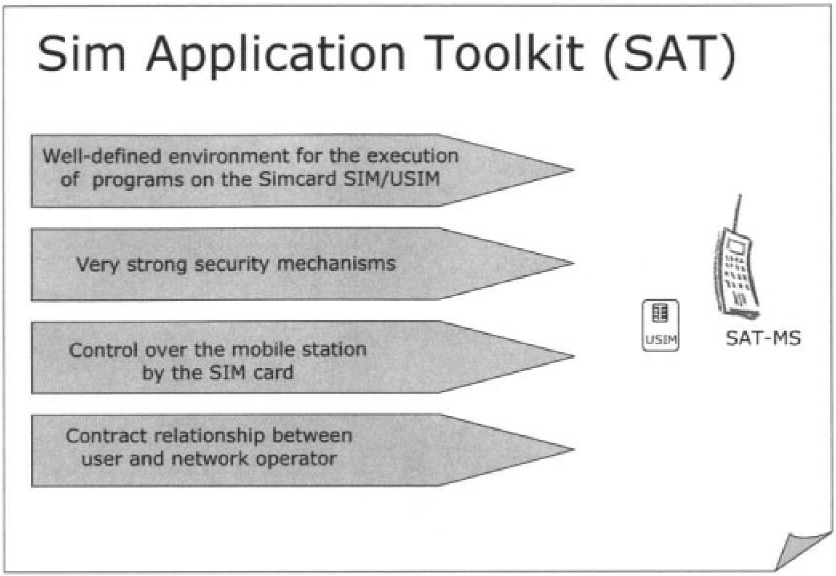

Figure 10.7: SIM Application Toolkit (SAT)

10.4 Open Service Architecture (OSA)

A very flexible, open architecture is necessary so that new applications with yet unknown characteristics can be run in 3G networks in the future. *Open Service Architecture* (OSA) is laying the foundation for the later development of these new services in UMTS (see Figure 10.8).

OSA consists of a defined interface, the OSA API, that enables external service providers to offer services to mobile radio network customers. These so-called *Home Environment Value-Added Service Providers* process data in their own infrastructures and transfer it over the OSA API to the mobile radio network. The aim is for the OSA API to be secure and open, i.e., vendor-independent. Programmers of an application can use functions of the UMTS network via the API without knowledge of its specific implementation within the network [31, 32].

A gateway in the UMTS network accepts the data and forwards it to internal servers of the network. The service may be controlled in the terminal by SAT or MExE applications. The major advantage of this solution for the operator is that he also makes a profit on the service and his business is not only limited to the pure data transmission. Users are assured that they are only using filtered and tested applications on their terminals.

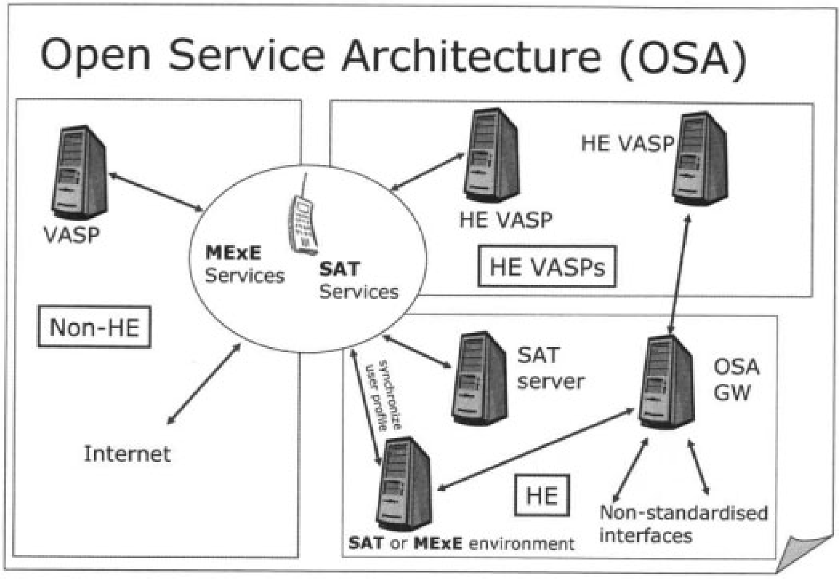

Figure 10.8: Open Service Architecture (OSA)

Of course, customers can download applications onto their terminals directly from the Internet or from other service providers. However, there is no guarantee then that an application will function as it should and it could even cause damage. Some experts are already talking boldly about "mobile phone viruses".

Decoupling applications development from network operations over an open interface is a very important factor for commercial success, because it makes it easier for external companies to develop new services for UMTS. This approach resembles the success of the PC architecture that has also become very popular due to its openness and has made many alternative proprietary systems redundant.

However, the approach is not uncontroversial. The *Session Initiation Protocol* (SIP) presented in Section 4.10.1 also enables the open development of new services, in this case even beyond the Internet and UMTS. Whereas many telecommunications firms support OSA, SIP finds support in the IT camp. Because of this competition many companies are still undecided as to which system to implement.

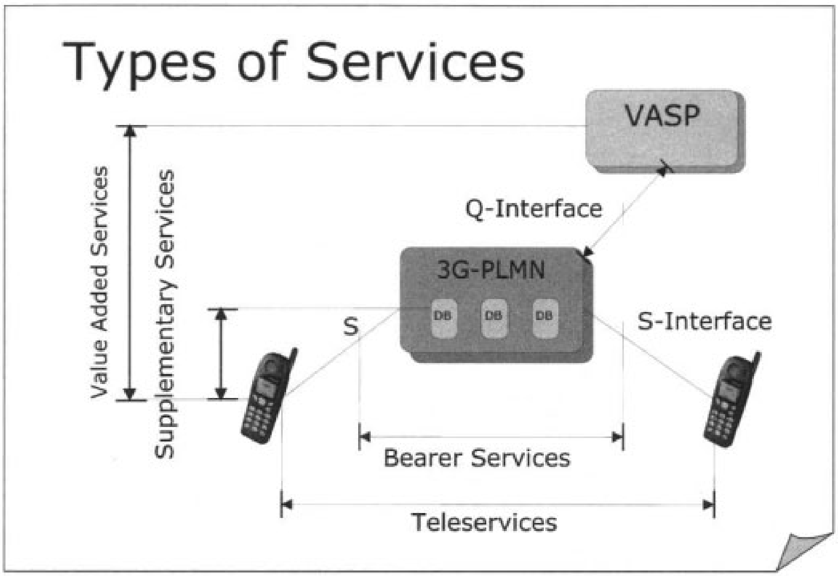

Figure 10.9: Classification of services

10.5 Services and mobile applications

The different services can be categorised into different classes according to the recommendations of the ITU: bearer services, teleservices, supplementary services and value-added services (see Figure 10.9).

Bearer services represent the pure transmission capacity of a network. They are defined by the data rate, the permitted delay, the tolerated bit error ratio, and so forth [29, 25]. Each telecommunications service is based on bearer services. Examples of bearer services include an ISDN B-channel with 64 kbit/s or a GSM data channel with 9.6 kbit/s.

The *teleservices* are based on these bearer services. Compared to bearer services, teleservices are also characterised by the parameter of the end-to-end connection between participating terminals. Voice telephony and the fax service are examples of a teleservice. A teleservice can be provided over different bearer services: for example, a voice telephone call can be conducted either over an ISDN B-channel or over a GSM traffic channel.

Services called *supplementary services* can be negotiated between a terminal and the network. Supplementary services supplement the teleservices with additional service features, such as call diversion and call number display.

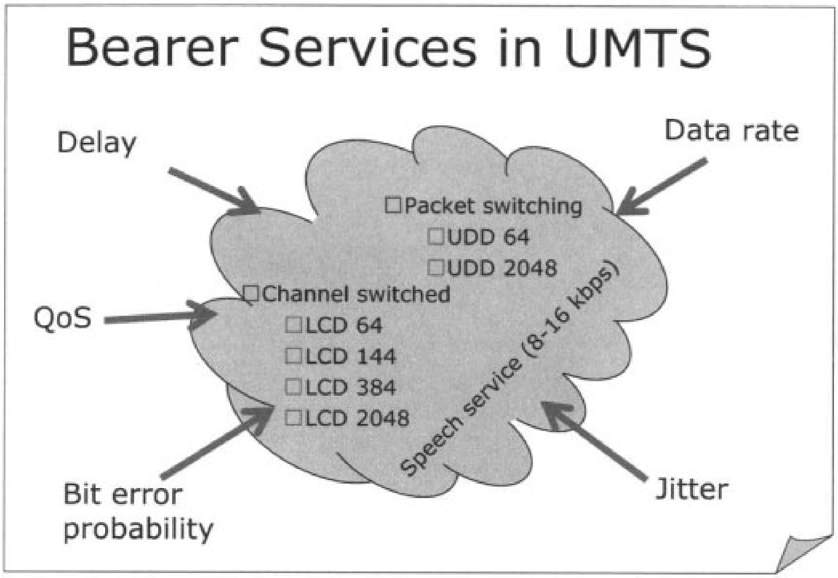

Figure 10.10: Some bearer services in UMTS

Supplementary services were introduced in fixed networks with the migration to the ISDN and are already available in GSM.

Visionaries see ***value-added services*** as a possibility for covering the immense cost of UMTS: Value-added services are services that are often taken over by external value-added service providers and that increase the added value created by teleservices. Value-added services could include the automatic transmission of traffic jam warnings by terminals in telematic services. In the future terminals will take over more and more of the functions currently being handled by other devices. For example, network operators are considering offering payment functions with UMTS. All these new services are refining simple telecommunications services into a value-added service.

No exact bearer services are defined in the UMTS standard. Although the standard includes a number of reference configurations, the radio interface can be configured very flexibly. The possibilities extend from very low bit rates to data rates exceeding 1 Mbit/s with varying quality of service parameters.

Figure 10.10 shows some of the quality of service parameters that define a bearer service. The label *Long Constraint Delay* (LCD) stands for *long constrained delay data (circuit-switched)* and refers to continuously transmitted data, such as video streams and audio data. Transmission using *Unconstrained Delay Data* (UDD) (*unconstrained delay data - packet switched*) achieves a

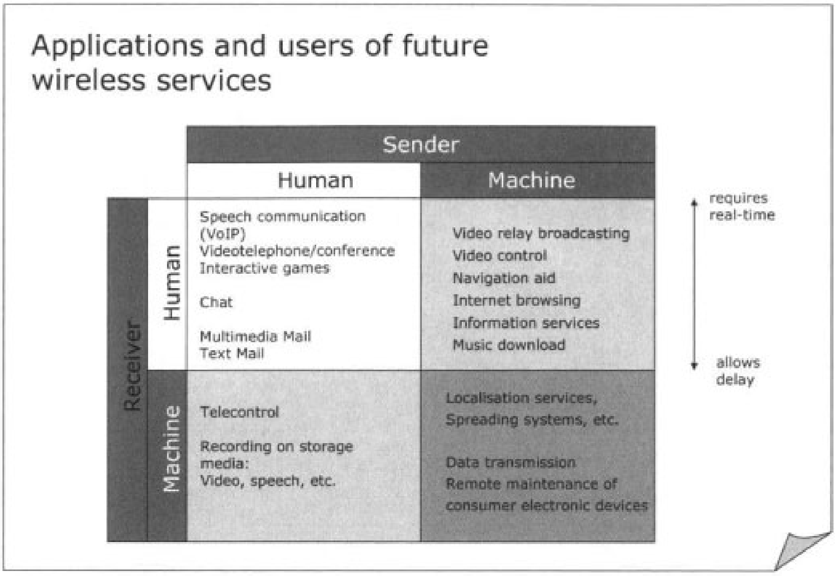

Figure 10.11: Applications and users of future wireless services

high level of data integrity because defective data packets are detected and requested again over an *Automatic Repeat Request* (ARQ) technique.

The applications and users of future wireless services are shown in Figure 10.11. Humans and machines will become senders as well as recepients of large data volumes when they start using real-time and other applications. Wireless communication between machines is expected to produce more message traffic than the communication initiated by people or directed towards people.

The restrictions that exist today in terms of quality of service, characterised by low throughput, high transmission delays and bit error probability, will largely be eliminated. This will be possible due to wireless access with an average (2 Mbit/s) to high (> 25 Mbit/s) transmission rate, comparable to the fixed network. Support will be provided for multimedia (MM) services and applications in the office, in the home and in high-traffic public service areas; support for these high data rates will even be available in a later stage for moving terminals (in vehicles) [11].

The different telecommunications networks that form the basis for such services as transport platforms appear at the bottom of Figure 10.12. These services include ISDN, broadband ISDN, Metropolitan Area Networks (city networks), Local Area Networks (inside buildings), broadband distributed networks (cable television distributed networks that in the future will transmit

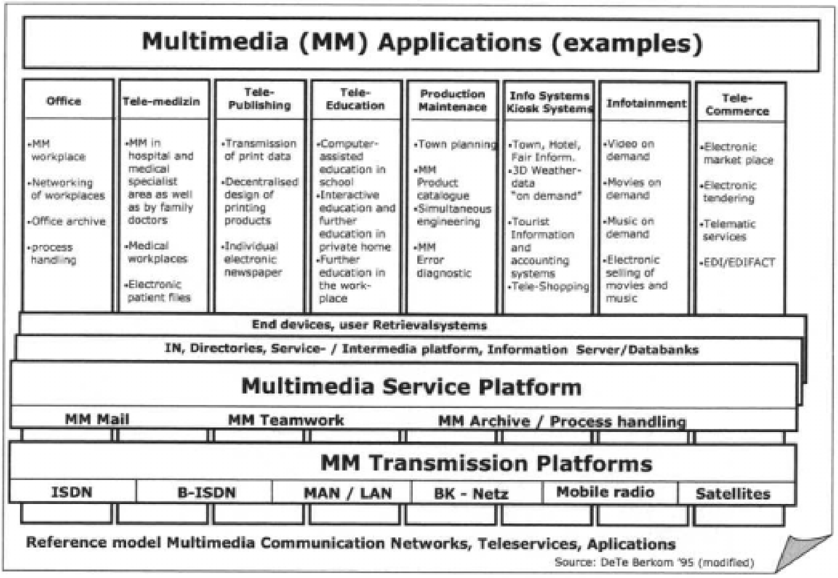

Figure 10.12: Examples of multimedia applications

bidirectionally) and cellular mobile and satellite radio networks. In the upper part of the figure is a list of examples of representative MM applications that are run on workplace computers (called service platforms).

Recent research shows that the support of *Multimedia* (MM) services for moving and mobile users can be met for many applications through a cost-effective solution based on the principle of cellular networks. The mobile radio network functions as a back-up in case a broadband network is not available but when there is an immediate need for important communication and cost plays a only secondary role. The communication would involve voice and important data connection with low bit rates. This system is described in Section 11.9.

The content of MM communication between people and/or machines can be categorised and divided into four sectors:

- Information
- Production data
- Communication
- Entertainment

The users of the services listed in Figure 10.13 can also be roughly broken down into private individuals, self-employed people (*small office*, *home office*), small to medium-sized organisations and large organisations. Some of these services or applications will most likely generate the most traffic over the coming years, namely through network-supported trade between organisations and customers (*electronic commerce*, *business-to-customer*), network-supported

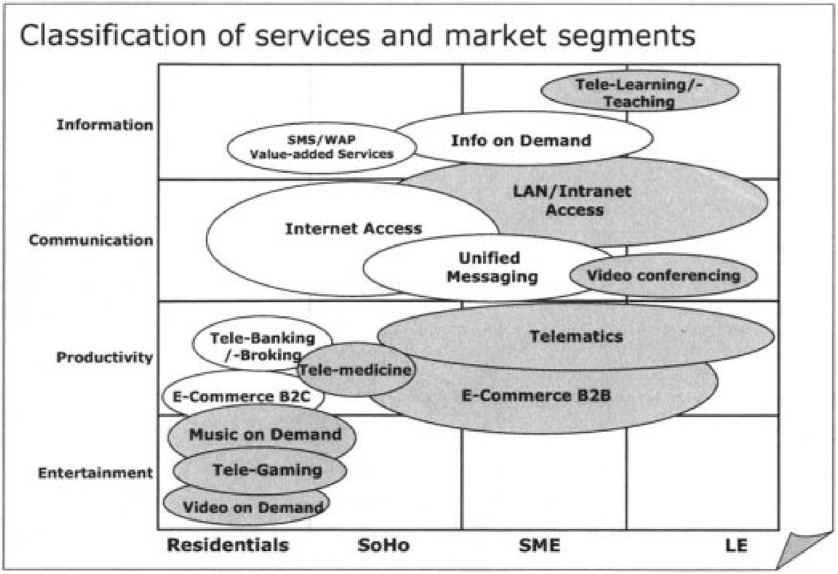

Figure 10.13: Categorisation of services and market segments

banking and market transactions (*telebanking, telebrokering*), Internet access, added-value services on the basis of *short-message services* in GSM) and browser programs suitable for mobile terminals, e.g., the Wireless Application Protocol of the GSM/GPRS service or UMTS, as well as controlled access to digital content. More importance will also be given to the uniform support of different communications services, such as the telephone, fax, data transmission, paging, email and mobile radio. This will be provided by platforms that support uniform communications services (*unified messaging*).

The users of these services will for the most part be from the groups shown in Figure 10.13. The size of the ovals represents the scope of the data quantities involved. In addition to the services and applications already mentioned, the other ones shown in Figure 10.12 are also of interest. These sectors are expected to make an important contribution to the use of wireless or mobile radio networks in the future.

Even a small GSM mobile terminal is capable of supporting a wealth of services and applications, as shown in Figure 10.14. Internet-related telecommunications services contribute 30% of the revenue; access services contribute a further 13%. Location-based services equate to 15%, the same level as the network-supported banking business. The remaining anticipated revenue is distributed over smaller segments such as simple information services, mobile

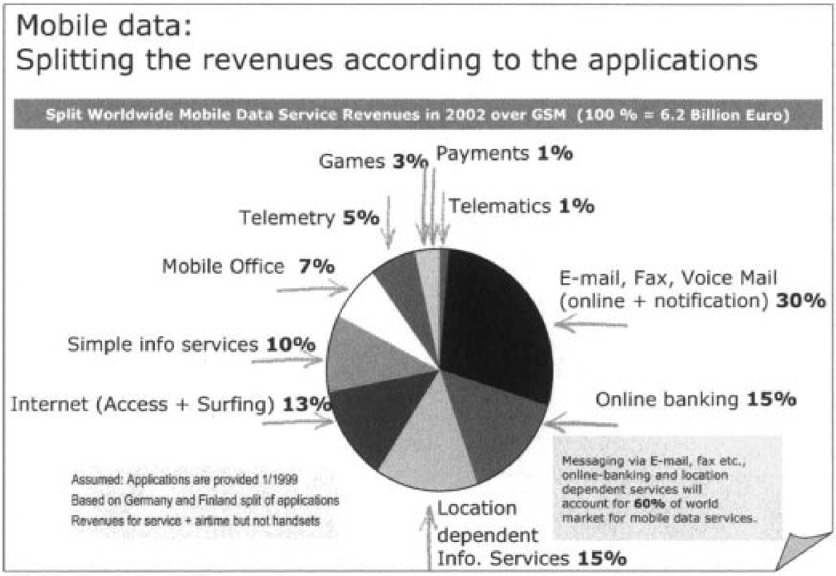

Figure 10.14: Breakdown of income from mobile data applications

offices, building and home automation via telemetry, traffic telematics, games and online payment systems.

10.6 The voice service in UMTS

UMTS will use a new speech codec with a data rate that can be changed dynamically to match current conditions. The *Adaptive Multi-Rate* (AMR) codec can transmit voice data with data rates between 12.2 kbit/s and 4.75 kbit/s [30]. The data rate can be changed per command every 20 ms. Depending on the quality of the radio interface, these commands are transmitted from the decoder of the voice frame to the coder. No layer 3 signalling is involved in this change. With a low bit-error link, voice can be transmitted clearly with a low data rate. When the bit-error ratio increases, the data rate of the speech coder then increases so that high-quality voice can still be transmitted. Such an adaptive speech codec is particularly suitable for use with UMTS because the transmission rate of the physical channels can vary very quickly and be matched to the required transmission capacity. This ultimately maximises the traffic capacity of the UMTS network.

With a data rate of 12.2 kbit/s, the codec corresponds to a GSM Enhanced-Full-Rate Codec. Each implementation of an AMR code must contain all 8 coding levels. The voice data is delivered to the codec as PCM-coded speech with 8000 samples/s. On the basis of *Voice Activity Detection* (VAD), the codec recognises whether a user currently talks. If there is no voice data to send, *Silence Descriptor* (SID) frames with a data rate of 1.8 kbit/s are transmitted. The decoder generates synthetic *comfort noise* from the data received to prevent the receiver from having the impression that the connection has been cut off. The general expectation is that the new codec will provide a considerable improvement to voice quality, even under difficult conditions.

10.7 Questions

10.1 Name two advantages the *Virtual Home Environment* (VHE) offers users.

10.2 Describe the four elements of the VHE basic architecture and explain how they interact. Why are defined and open interfaces important?

10.3 Explain the basic difference between SAT and MexE.

10.4 Give two examples each of a bearer service, a teleservice, a supplementary service and a value-added service.

10.5 Name three parameters that describe a bearer service.

10.6 Explain why value-added services can possibly make the high investment in UMTS profitable.

10.7 What is the maximum data rate needed by an AMR speech codec? What is the minimum data rate needed?

10.8 With which scanning rate is PCM-coded voice information supplied to the AMR codec?

11 The Next Generation of Mobile Radio Systems

Wireless access to fixed networks at mean and high bit rates will increase in importance: wired communication, for example, via cable or fibre optics, severely limits the use of network-dependent services. The Internet is a good example of this. Wireless Internet access necessitates an almost omnipresent level of availability. It requires high-bit data transmission and a considerably higher capacity of radio system spectrum than is offered by current cellular systems. This is the only way that multiple users can communicate simultaneously at the same location without causing interference to one another.

This chapter starts by characterising the difference between cordless, wireless and mobile radio systems. Only the latter allows communication at high speeds. Cordless systems were originally developed for telephony inside buildings and replace the cable between telephone terminal and handset with a radio path. The concept of wireless systems comes from the computer technology area and describes radio-supported data transmission systems. Radio transmission is not possible unless a licence exists for the frequency spectrum. The achievable message traffic capacity mainly depends on the bandwidth (measured in Hertz) provided for a radio system.

The advertising promised amazing things for third-generation systems. But even here the wheel has not been reinvented and the evolutionary steps taken *vis-à-vis* 2G systems have been impressive but not sensational. The anticipated 2 Mbit/s transmission rate for mobile users is not realisable and was also not specified as such. In fact, many factors have interacted together to produce a usable bit rate per user that only approximates that of an ISDN basic rate channel.

Wireless local networks (W-LAN) are excellent for high bit rate Internet access. Systems of this type have been available since 2000 and starting in 2002 they will be able to fulfil almost everyone's requirements. However, multimedia communication will not be possible at a favourable cost everywhere without a combination of cellular mobile radio and *Wireless Local Area Network* (W-LAN). Integrated systems that can support two (or more) radio standards will be introduced as from 2004.

A first version of W-LANs will be transmitting at 2.4 GHz, directly above the UMTS spectrum, and a second version, expected in 2002, will be transmitting in frequency ranges that tend to have quasi-optical propagation characteristics (5–6 GHz). Since sufficient spectrum is available in that range, one can expect

UMTS: The Fundamentals. B. Walke, R. Seidenberg and M. P. Althoff

that for a communication relationship multihop transmissions will be used via radio in order to connect two terminals not located in the radio range. Multihop systems are being mentioned, meaning the use of several sequential radio relay nodes providing links for a communication connection.

It is not always possible to plan the route of message flow and the participating terminals. Ad-hoc networks allow the operation of self-organising networks that configure themselves when activated and then permit communication between participating terminals (plug-and-play). Ad-hoc network capabilities can, for example, be useful in the organisation of multihop networks.

11.1 Cordless, wireless and mobile radio systems

A distinction is made between *Personal Area Networks* (PANs) that only have significance in the immediate vicinity of a user: cordless and wireless radio networks, which typically enable access to public or private fixed networks for telecommunications services, and mobile radio networks. The latter is divided into generations.

In Figure 11.1 GSM and its further development GPRS, which is used to transmit packet-oriented data, are classified as second generation systems; other representations refer to GPRS as a 2.5th generation. The combination of high bit rate W-LANs and 3G mobile radio systems is categorised as the 3.5th generation, with W-LANs only expected in hot spots with high message traffic. These are sometimes also referred to as evolved 3G systems. Integrated W-LAN and mobile radio systems that allow an almost omnipresent availability of the network for supplying multimedia services are categorised as next generation (NG) systems.

It should be noted that high bit rate W-LAN systems transmit in a frequency range that is embedded much higher in the spectrum (5-6 GHz) than mobile radio systems. This will make considerable demands on the terminals that must be able to operate two radio standards in very widely separated frequency ranges.

There is a considerable difference between the speeds of movement supported by the different radio standards; the multiplex transmission rate available to all terminals at the radio interface to a base station is also very different (see Figure 11.2). The 2G GSM system and its further developments GPRS and EDGE (*Enhanced Data Rates for GSM Evolution*) cover all speeds of movement but only offer (depending on the modulation and coding scheme) aggregate data rates between 80 kbit/s (GSM) or 128 kbit/s (GPRS) or 384 kbit/s (EDGE) at the radio interface.

The development of cellular mobile radio networks from 2G to 3G and then to 4G systems shows a constant growth in the number of radio standards. With

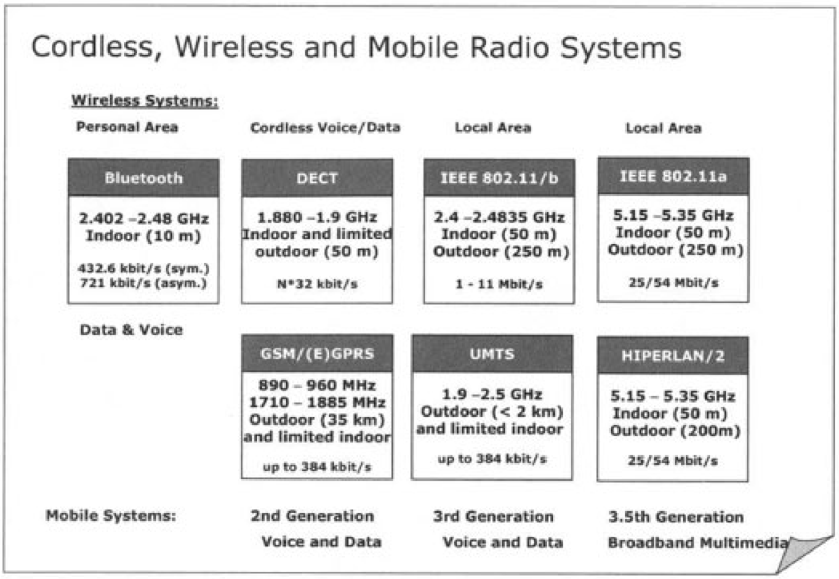

Figure 11.1: Cordless, wireless and mobile radio systems

2G there were only four standards worldwide, namely the two very closely related systems TDMA (the USA) and PDC (Japan), GSM (Europe but introduced worldwide) and the cdma2000 system using code-spreading techniques (the USA). The last two systems were actually further developments that evolved into the 2.5th generation (evolved 2G). With 3G systems, which are identified as *International Mobile Telecommunications at 2000 MHz* (IMT-2000) systems by the *International Telecommunications Union* (ITU), there are six mobile radio standards (of which only three main ones are represented in Figure 11.3) and two classes of W-LANs.

cdma2000 1XEV is a further development of the corresponding 2G system with higher transmission rates. The designation *Wideband Code Division Multiple Access* (WCDMA) conceals the two UMTS radio standards as well as TD-SCDMA, which is the 3G standard favoured by China and has a close similarity to the *Time Division - Code Division Multiple Access* (TD-CDMA) UMTS standard. EDGE is the more important of the two IMT-2000 TDMA-based standards; the other one is called DECT (both were developed in Europe).

It is obvious that 3G systems will also be undergoing further development. The integration of W-LANs in mobile radio systems will enable transmission rates of up to 25 Mbit/s at the air interface. 4G is represented as a research area here, which will comprise, among other things, the Mobile Broad-

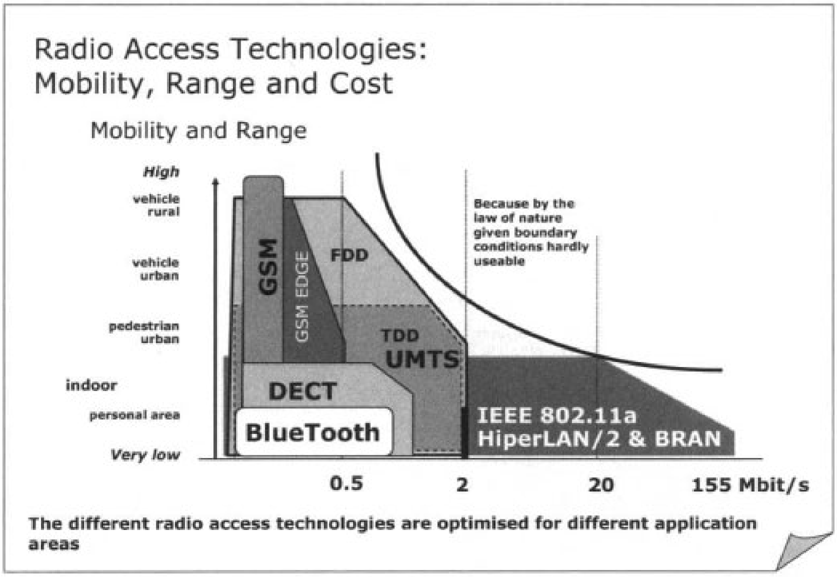

Figure 11.2: Mobility and data rates of different radio technologies

band System (4 /40 60 GHz), so-called ad-hoc networks and probably a new family of mobile radio standards that will probably be transmitting using coded multi-carrier modulation (coded *Orthogonal Frequency Division Multiplex* (OFDM)). The aim of the further development is to improve the quality of service for wireless and mobile communication. Instead of ultimate resolutions being provided for all problems relating to existing systems, 5G systems will be appearing on the horizon in the years beyond 2010.

A look at Figure 11.3 also highlights the fact that more radio standards have been generated than ever before to implement the same applications and that this trend will probably continue. The reason is that the competitive market is looking for the best solutions seen from applications. Some of the facts highlight this:

- The original intention of the IMT-2000 initiative to produce a uniform radio standard worldwide was not achieved and also will not be achieved in the future.
- New applications and the frequency bands allocated to them will require new radio standards that are adapted to the respective purpose. This will result in an increase in the number of radio standards.
- The frequency range in the spectrum to be covered by 4G systems is very large and it is not likely that a single radio terminal will be able

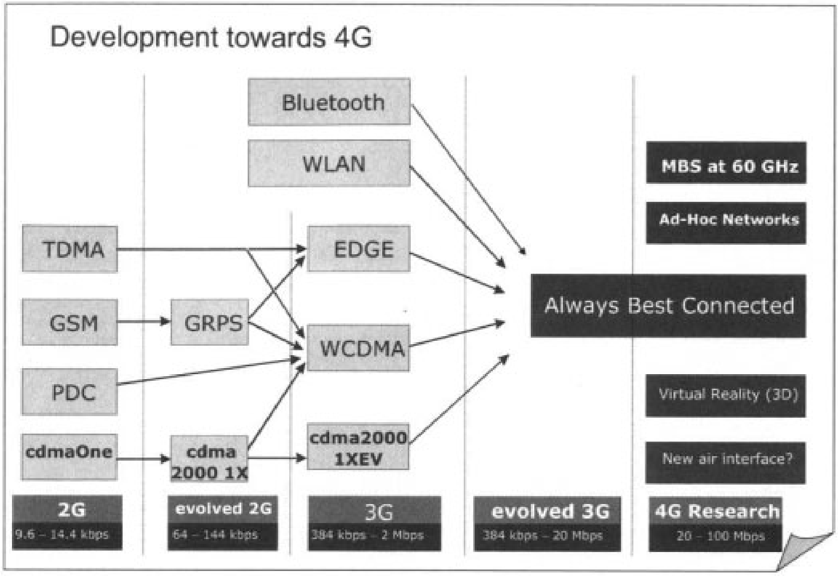

Figure 11.3: 4G development direction

to cover it in the foreseeable future. The size of terminal this would require and its power usage would not be in line with specific demands (as small as possible with a maximum battery life).

- The 3G radio standards are a temporary solution that will be replaced by improved systems.

- One of the greatest challenges in the development of mobile radio standards is and will continue to be the provision of higher transmission rates per user. This goal is trongly dependent on the technology progress towards higher processing power and lower power consumption. Of course the prerequisite for meeting this challenge is the availability of the necessary spectrum so that such systems can actually be implemented.

The table in Figure 11.4 indicates the time required to transmit a sampling of media objects of different dimensions – such as a Web page, the content of a book and a video – over different mobile radio systems. It is clear from the values given that mobile radio systems such as (E)GPRS and UMTS are unable to support continuous media with moving terminals, even with the assumption of the unrealistically high transmission rates given here.

Figure 11.5 schematically shows the anticipated development of terminals, radio standards, standards for fixed networks of mobile radio networks (so-called core networks) and applications for mobile use over the period of time. Ter-

Time needed to transmit the various media contents shown by using different radio network technologies

Application		Page of text	e-mail	Picture on laptop	WDR web home page	30-pp simple colour presentation	e-mail + excel file	Book (300 pptext)	30-pp simple colour presentation images	Professional quality photograph	15 minute video	CD-ROM
Size in bytes		3,125	3,125	50,000	100,000	100,000	750,000	937,500	2,000,000	2,400,000	300,000,000	650,000,000
Book equiv's (300pp)		1 page(s)	1 page(s)	16 page(s)	32 page(s)	32 page(s)	240 page(s)	1 book(s)	2.1 book(s)	2.6 book(s)	320 book(s)	693.3 book(s)
Size in bits		25,000	25,000	400,000	800,000	800,000	6,000,000	7,500,000	16,000,000	19,200,000	2,400,000,000	5,200,000,000
Size in mega/kbits		25 kbits	25 kbits	400 kbits	800 kbits	800 kbits	6 mbits	7.5 mbits	16 mbits	19.2 mbits	24 gbits	52 gbits
Technology	kbit/s	Time to transmit (secs)										
GSM today	9.6	2.6 secs	2.6 secs	41.7 secs	1.4 mins	1.4 mins	10.4 mins	13 mins	27.8 mins	33.3 mins	2.9 days	6.3 days
HSCSD	28.8	0.87 secs	0.87 secs	13.9 secs	27.8 secs	27.8 secs	3.5 mins	4.3 mins	9.3 mins	11.1 mins	23.1 hours	2.1 days
GPRS	115	0.22 secs	0.22 secs	3.5 secs	7 secs	7 secs	52.2 secs	1.1 mins	2.3 mins	2.8 mins	5.8 hours	12.6 hours
EDGE	384	0.07 secs	0.07 secs	1 secs	2.1 secs	2.1 secs	15.6 secs	19.5 secs	41.7 secs	50 secs	1.7 hours	3.8 hours
UMTS on the move	384	0.07 secs	0.07 secs	1 secs	2.1 secs	2.1 secs	15.6 secs	19.5 secs	41.7 secs	50 secs	1.7 hours	3.8 hours
Stationary UMTS	2000	0.01 secs	0.01 secs	0.2 secs	0.4 secs	0.4 secs	3 secs	3.8 secs	8 secs	9.6 secs	20 mins	43.3 mins

Telecoms speeds are measured in bits per second but most items that are transmitted are measured in bytes (there being 8 bits to a byte)
Bytes are measured in powers of 2 - 2^{10} is 1,024, a kilobyte, whereas a kilobyte is 1,000
The indicative speeds are theoretical measures of raw bandwidth; effective bandwidth is inevitably slower given the performance of modems, switches and other network elements

Source: UBS Warburg

Figure 11.4: Time requirement for the transmission of media objects over different mobile radio systems

minals have developed from single-band to multi-band devices (e.g., initially a GSM terminal only operated at 900 MHz, later it also had to be able to operate in the 1800 /1900 MHz band). Their development will have to continue in the direction of multi-standard devices to adapt to the different radio interfaces of competing operators. This is the expectation of 3G and 4G systems.

Third-generation radio interfaces will soon be transformed into (evolved) 3G+ interfaces that integrate cellular and broadcasting services or become modified 3G systems that can provide optimal support to services with asymmetrical message traffic.

The mobile radio core network will be developed so that it can support packet-oriented high bit rate data and multimedia applications, with more advanced Internet protocols eventually being used in combination with *Asynchronous Transfer Mode* (ATM) technology. Multimedia applications that transmit high bit rates will be supported by a combination of the highest capacity radio interfaces. All services familiar from the various wireless and mobile radio systems will be carried over the different radio interfaces in order to achieve unrestricted connection between user terminals and partner terminals over radio.

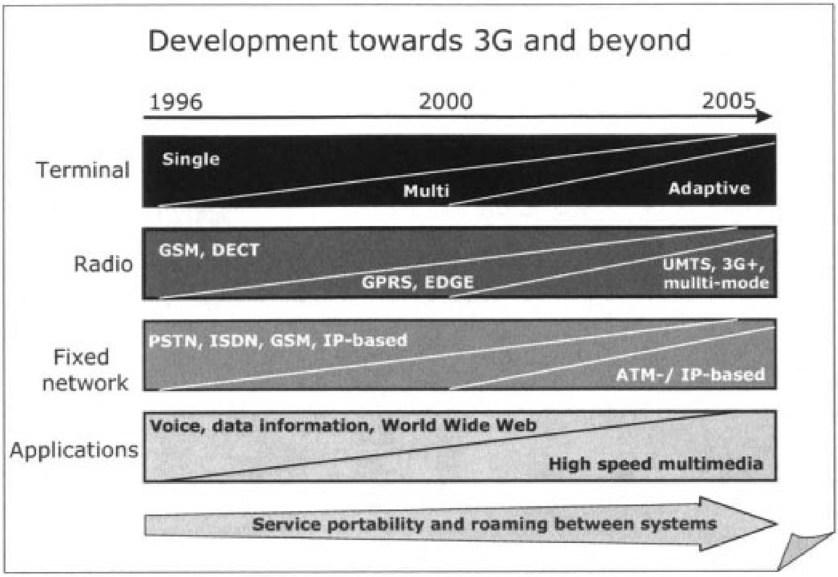

Figure 11.5: Anticipated development of terminals, radio standards and fixed network standards of mobile radio networks

11.2 Asymmetric traffic in mobile radio systems

The projections published by the UMTS-Forum and the ITU on future requirements for frequency bandwidth for the different services are presented in Figure 11.6. A significant asymmetry is expected in spectrum requirements particularly for average to high bit rate multimedia traffic. This leads to the conclusion that the downlink of IMT-2000 systems will ultimately be the transmission direction that determines the capacity. Accordingly, IMT-2000 systems will be developed so that considerably higher transmission rates can be achieved on the downlink than on the uplink. The options of variable duplex spacing and simultaneous use of several *Frequency Division Multiplex* (FDM) channels on the downlink are already being discussed in conjunction with UMTS standardisation

An allocation of extension bands for IMT-2000 systems should take into account the changing degree of asymmetry in traffic volumes time-wise and physically. The ideal approach would be a dynamic use of spectrum adapted to the characteristics of the respective traffic volume. An interesting way to establish such a use of spectrum is through the integration of high bit rate digital broadcasting services (downlink) into cellular mobile radio systems.

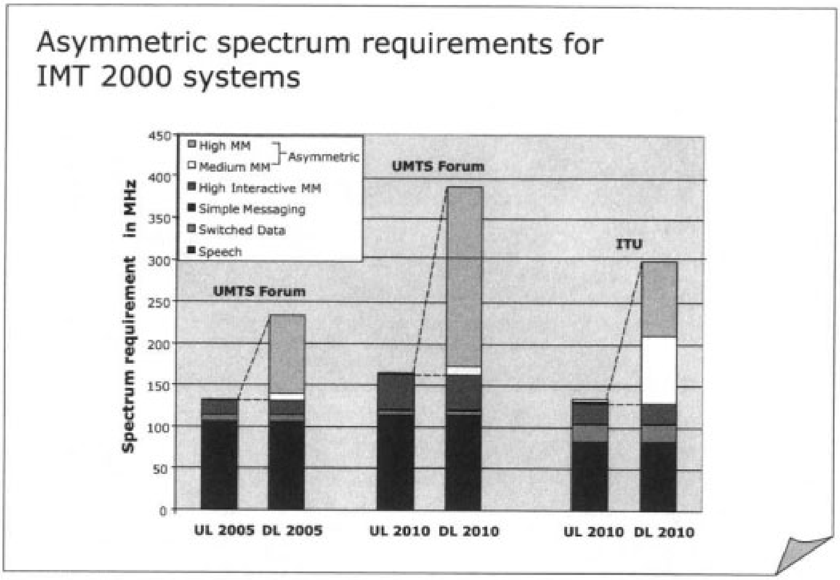

Figure 11.6: Advance calculation of future requirements for frequency bandwidth (sources: UMTS-Forum, Report No. 6 and ITU-R)

A service called ***High Speed Downlink Packet Access*** (HSDPA) has recently been standardised as part of UMTS release 5. It allows the application of adaptive modulation techniques to go beyond QPSK modulation, thereby increasing the number of bits per symbol transmitted from two to three, four and even six under good, better and excellent receive conditions, respectively [21].

11.3 Spectrum issues

In 2000 various mobile radio operators in Great Britain and in Germany paid licensing fees of approximately 5 (UK) or 8 billion (Germany) Euros each at auction in order to obtain 10 MHz of paired spectrum for 3G systems for an operating period of 20 years. Most of the operators also bid on and received 5 MHz of unpaired spectrum for the sum of 0.2 million Euros. These licences require that the new 3G service must reach, e.g. in Germany, at least 25% of the population by 2003 and at least 50% by 2005. The building of the infrastructure required as well as the subsidy costs for the new terminals will cost each operator an additional 5-10 billion Euros. In addition to 3G/UMTS, the new terminals also have to support GSM/GPRS in two frequency bands since this is the only standard available throughout the countries and worldwide.

As a result of the high licensing fees, infrastructure costs and subsidies in Europe, the expectation is that services requiring a high transmission rate will be too expensive for the normal user. Consequently, the focus will be more on services that manage with low amounts of data and data transmission rates per use occurrence but can be charged at a high tariff similar to or even beyond that of a voice minute. Examples of this include the short-message service of GSM and the I-mode service of NTT DoCoMo. Especially, *Wireless Application Protocol* (WAP)-based services are a big hope to pay back the investments made by the operators.

The current allocation of spectrum by the *World Administrative Radio Conference* (WARC)-92 (1992 session) for 3G systems is not sufficient to provide the spectrum capacity required to implement services with mean or even higher required data rates for a significant number of simultaneous users.

Around 200 MHz to 400 MHz of additional spectrum, used in pico-cellular cell structures, is required to achieve the required capacity. In all probability such a broad frequency band is only available in the spectrum above 2.69 GHz, therefore above the IMT-2000 extension bands as defined by WRC-2000 (see Figure 3.17).

With the knowledge that spectrum for mobile radio services with mean to high transmission rates is scarce, WRC-92 also provided for spectrum between 5 GHz and 6 GHz for wireless broadband systems (W-LANs) (see Figure 11.7). It is proposed that this spectrum be usable without a licence. The size of this spectrum is 455 MHz in Europe, 300 MHz in the USA and 100 MHz in Japan.

This allocation of spectrum will greatly accelerate the breakthrough needed for wireless Internet access. The first standards and associated products for wireless access already exist, e.g., *Institute for Electrical and Electronics Engineers* (IEEE) 802.11(b), a system that transmits in the *Industrial, Scientific and Medical* (ISM) band at 2.4 GHz (79 MHz bandwidth), as well as IEEE 802.11a and *High Performance Radio Local Area Network Type 2* (HiperLAN/2) that are standardised for the 5 GHz range and will be available as products starting in 2002. The W-LAN spectrum was partly planned for indoor radio coverage with a transmitter power of 200 mW *Equivalent Isotropic Radiated Power* (EIRP) and partly for the supply of outdoor coverage with 1 W EIRP.

Mobile radio is characterised by its need to provide radio coverage to large geographical areas for use by highly mobile users (in vehicles, trains). The costs for this coverage should be within a limit that will result in an acceptance of the services provided.

All spectrum allocations for mobile radio systems above 1 GHz (= 1000 MHz) are considered unfavourable for mobile radio use because the attenuation of

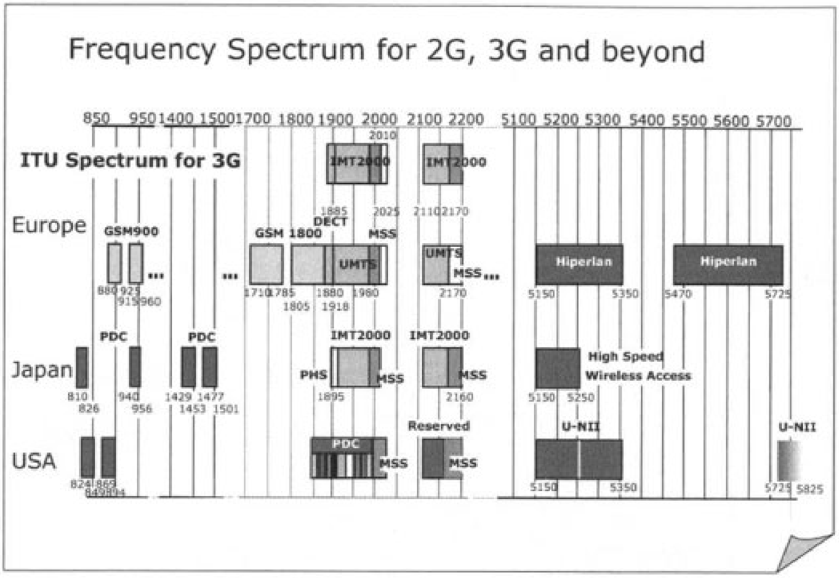

Figure 11.7: Worldwide frequency allocation for 3G and 4G systems

the radio waves when propagating through obstacles increases with frequency, and diffraction as a prerequisite for unbroken illumination, also of shadowed areas, is clearly smaller. (The degradations of propagation characteristics are naturally not abrupt but the radio propagation becomes more and more unfavourable as the frequency increases.) Allocations above 2 GHz are only acceptable as a compromise when no other ranges in the spectrum are available or can be freed up. Allocations above 2.5 GHz (as planned for the IMT-2000 extension band) are unreasonable for rural areas and can only be used elsewhere if a very high investment is made in infrastructure.

Another problem with the licensing for IMT-2000 systems in some countries is that the licensing fees themselves are much too high (and there are many reasons for this). As a result, the competition that was intended to provide reasonably priced mobile radio services has effectively been eliminated, contrary to what the political players claimed. At the same time 3G technology in Europe has suffered a setback that can never be made good again: its sluggish introduction in Europe is providing manufacturers of 3G systems other than UMTS with an advantage, giving them a head start in unit production and thus enabling them to establish themselves successfully on the world market. Due to the greed of the governments of Great Britain and Germany (two of the most important markets in Europe), UMTS most likely will have difficulties in repeating the global success of GSM. Yet down the road it will be the engineers who are unfairly forced to take the blame for not having the capabilities to compete in the marketplace.

11.4 Mobile radio and television frequencies

A comparison of the allocated frequency bands for mobile radio and terrestrial television broadcasting in Germany produces the following conclusions:

- All mobile radio operators combined have licences for 365 MHz.
- In total television (TV) occupies a spectrum of 431 MHz in the following ranges:

  Band I: 47–68 MHz (21 MHz)

  Band III: 174–230 MHz (of which 216–230 MHz is used for DAB; 42 MHz)

  Band IV/V: 470–862 MHz (of which 838–862 MHz is used by the military; 368 MHz)

  Calculated in the value of UMTS licensing costs, this equates to a value of 144 billion Euros.
- The situation is quite similar in other countries in Europe.
- The economic benefit of the allocation of TV spectrum is very low. It bears no relationship to its market value (even if one were to rate it much lower today).
- At one time 100% of all TV users received coverage from terrestrial TV; today, in most countries this figure is very low now, e.g. 8% in Germany, most users now receive their TV programmes by cable or satellite.
- Since a very high percentage of TV users in many countries have decided against using the terrestrial TV service, TV operators cannot continue to retain their excessive spectrum allocation. This spectrum would be optimally suitable for mobile radio. Since there are large areas in all countries where only some programmes can be received terrestrially, large parts of the occupied TV spectrum are wasted, even considering the conversion to digital television.

The regional public and private broadcasting companies claim within the framework of their role of performing a public service to have the right to the continued free use of the terrestrial spectrum or the corresponding TV channels used in the distribution network as part of the basis of their existence. The broadcasting companies and the respective regional politicians have a mutual interest in one another. The system appears to be relatively stable, even if the reason for its existence has largely become invalid. The realities of the market seem to be ignored here.

As part of its role to ensure the shaping of free public opinion, it was decided about 50 years ago that all citizens must be served by radio broadcasting. The concept of radio broadcasting also includes the television service. The federal government meets this demand by providing radio spectrum free of charge to the appropriate institutions.

After a period of transition (agreed between the federal states) during which analogue and digital TV would coexist, the digitalisation of television would render around 35–50% of the currently allocated spectrum unnecessary. The other possibility is to introduce more terrestrial broadcast programmes and fill up the spectrum freed by digitalisation of the service.

The introduction of broadcast services with digital content accessible over the Internet planned by the broadcasting companies would be considerable competition for mobile radio operators. Many of the data applications expected contain content that provides subscribers with information about current events, e.g., the stock market and ongoing developments, sports results and sports tables and lottery numbers. Such content can easily be implemented through broadcast services communication and the demand for this kind of information can be served by mobile radio operators against fees per call.

Figure 11.8 shows the current use of television bands in the spectrum between 470 MHz and 862 MHz that is of particular interest for mobile radio, this is the range provided by the ITU for these services. A part of the television band is used for DVB-T testing. In some countries the television band actually ends at below 862 MHz. Generally speaking, organisations responsible for public services (this does not include the broadcasting companies) have more than 30% of the entire spectrum under their control along with the benefit of free licensing.

The three alternative scenarios 1 to 3 shown were developed by two research projects [4, 5] that are researching possibilities for the future use of the television band for mobile radio. The currently popular view is to integrate both services into a joint system with television bands planned to implement a high capacity downlink for asymmetric mobile radio services. There are different ways in which this is possible:

- Use of digital television (*Digital Video Broadcasting Terrestrial* (DVB-T)) on the downlink to supplement UMTS services there. This requires a common coordination control in an integrated mobile radio and distribution communication system, which is just now being studied.
- Time-shared operation of mobile radio and broadcast services on the same TV channel. In scenario 1 channels of the band used by the military 814–862 MHz operate this way for mobile radio.

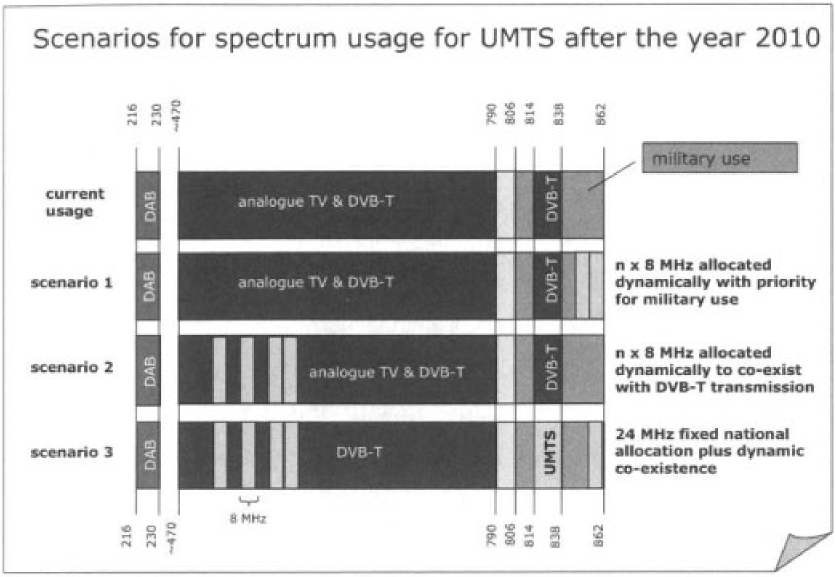

Figure 11.8: Scenarios for spectrum use in Germany after 2010

- In scenario 2 a public mobile radio network is operated on certain television channels that can vary from region to region. This is referred to as dynamic channel allocation, which will be possible depending on the time of day. A television channel is 8 MHz wide. Based on the results of the projects mentioned, interference-free operation of UMTS, which has a channel grid of 5 MHz and a channel bandwidth of 3.8 MHz, will be possible.
- Scenario 3 assumes that a certain number of television channels are permanently allocated to mobile radio.

All scenarios will have practical significance in the future, because it is hardly likely that terrestrial television will be able to retain the frequencies currently allocated to it in the long run. The hybrid system resulting from the integration of both system types would of course require multi-mode terminals that support UMTS as well as DVB-T. Due to the high power usage in DVB-T receivers, such operations would only appear feasible for permanently installed terminals in vehicles. Experiments in 2001 in both projects showed that all ideas are feasible in practice. However, the MPEG-2 container specified for video transmission makes the DVB-T standard less suitable for data transport. Even at high speeds a distributed communication data rate of 12 Mbit/s could be achieved per DVB-T (8 MHz) channel. Cell broadcast, a service in

UMTS, in particular could be implemented cost-effectively on this basis and the capacity of UMTS increased.

A homogeneous system with one radio standard (instead of two), such as the one assumed in scenarios 1 to 3, of course appears more attractive. The asymmetric downlink could then be operated by all mobile radio base stations and not only from the small number of television towers. This would increase capacity substantially and provide a more location-specific broadcast service than is possible with DVB-T.

The freed up TV spectrum could be used for the following:

- Constant allocation of individual channels or channels from TV channel groups to mobile radio.
- Operation of 3G downlinks in the TV band (for asymmetric traffic).
- Combination of 3G and DVB-T systems, e.g. based on the concepts of [4] and [5].
- Use of spectrum by an operating company consisting of mobile radio and television companies, each contributing its own spectrum.
- Time-shared use of TV channels for mobile radio and TV.

Digital broadcast communication over television networks based on the DVB-T standard would benefit from the availability of a reverse channel for (location-specific) activation of the desired content. The reverse channel can easily be implemented over the fixed network. Mobile radio networks would make the reverse channel available from all locations and even for moving users. Through user-specific control over a reverse channel the broadcast communication network would have the capacity to distribute personal content (for a charge). This would make it a competitor of the mobile radio networks.

Mobile radio networks will also offer content that can be provided most cost-effectively by broadcast communication. The 3G standards provide for this possibility. The integration of TV broadcast communication systems with mobile radio networks enables a utilisation of the strengths of the respective networks to increase the capacity of 3G systems for broadcast services.

Broadcast communication requires a high-capacity downlink from the base station. From the view of mobile radio operators, under some circumstances (e.g., no alternative available) support of a second radio (DVB-T) in the mobile terminal is sensible. It would be more advantageous and economic if broadcast communication used the same radio interface (3G) as personal mobile radio.

Since many TV channels are not used during the day, especially outside of cities and beyond metropolitan areas, but mobile telephone users are frequently on the move on roads between cities, a dynamically controlled use of

TV channels would make sense. Certain channels would then be used for mobile radio during the day and then for television towards the evening. Mobile terminals would have to be able to receive the different carrier frequencies and synthesise them. New studies show there are certain situations in which DVB-T and UMTS could even be operated simultaneously without much interference.

11.5 Electromagnetic compatibility

One topic that is causing heated discussion concerns the possible harmful effects of the electromagnetic radiation emitted by mobile telephones. Along with the warming effects in biological organisms, the time-fluctuating pulse-type power caused by the time-multiplex technique used by the GSM system is presented as being harmful. The radiated power pulsating with a frequency of 271 Hz is mainly found in the handset. The operation in the base station of simultaneous time-multiplexing channels in the same frequency band results in a quasi-continuous power flow. All concerns regarding biological harm due to pulse-type signal flow are mostly related to the handset, the use of which is a free choice. There is no proof of biological harm based on the official boundary values and most of what one hears is supposition and the small possibility of a residual risk.

GSM-based mobile radio system has been available on an area-wide basis in many countries in Europe since about 1995. The operation of GSM could be viewed as a big field study to test the harmful effects of the electromagnetic waves produced by mobile radio. So far the results have not provided any proof of an adverse impact on people's health.

The radio and television broadcast network has been in existence on an area-wide basis for a good 50 years. The radiated power it uses is clearly 100 times greater than that of mobile radio. It is true that many transmitter antennas are located far from inhabited areas; but this is not always the case. Many television towers are sited in the middle of large cities. Measurements show that local radio coverage is very good on an area-wide basis (much better than with mobile radio) and that the radiation power is often much higher than with mobile radio [37]. In a country like Germany the total radiated power per mobile radio operator on a country-wide basis is 400 kW per radio channel, corresponding to 40 fully switched on electric kitchen cookers (distributed over all of the country). With television broadcast the radiated power is approximately 100 times more per channel and there are over 20 terrestrial television channels in operation. As mentioned, this "warming of the atmosphere" caused by television transmitters could be totally eliminated if there was an the intension to do so. Broadband cable-based distribution networks and satellites are now available and could be used more extensively.

In UMTS-based mobile radio systems the power is only affected by the power control and is radiated evenly (see Chapters 7 and 8), unlike GSM where it pulsates. The power of transmitted signals in UMTS is distributed over a broad frequency spectrum, whereas in GSM the radiated power is always only concentrated on around one-ninth of the overall frequency bandwidth assuming a cluster size nine.

If on a medium-term basis the UMTS was subject to a higher spectral efficiency than GSM, it would use less energy to transmit the same quantity of data than GSM would in the same period. Since it is projected that there will be a considerable increase in the amount of data transmitted over mobile radio in the future, the radiation produced through the introduction of UMTS will be much less compared to GSM. Concerned citizens should therefore support the introduction of UMTS. Instead they are using their energy to fight against this new technology because they do not want to see any more antenna sites. It is unfortunate that the political forces are feeding the concerns of the people and stirring their fears without any scientifically based reason.

Almost everything a person uses or does has benefits and side effects. Generally the benefits are weighed against the disadvantages associated with the possible side effects, e.g., driving a car (5000 dead, 30,000 seriously injured per year in Germany alone), using medication, smoking, drinking alcohol and consuming food. If the risks appear too high, people cut down on the usage or totally do without. In the case of mobile radio, the highest source of radiation by far is the mobile telephone handset used close to the head – and not the radio masts. Everyone can therefore easily control the amount of radiation being received.

11.6 UMTS message traffic and capacity

According to the specifications for the development of the UMTS standard, the target volume of data throughput is 600–700 kbit/s per 5 MHz carrier frequency. Figure 11.9 shows the throughput overload resulting from the traffic of WWW users on an emulated UMTS system (see [35]). With UMTS the user of a service is allocated a physical channel with a data transmission rate that can be selected from the system. Figure 11.9 shows the results gained for channels with data rates of 64, 128 and 256 kbit/s. What is noticeable is that maximum system throughput occurs the highest with 64 and 128 kbit/s channels, reaching approximately 570 kbit/s. The 2 Mbit/s data rate for users constantly mentioned in public debates is based on advertising-motivated simplifications. It is true that a 2 Mbit/s data rate is possible over a 5 MHz carrier, but only if only one user is communicating and there is little interference and time dispersion of the signal on the radio path. Unfortunately, this is practically never the case. The total data rate available per carrier is divided up among the users (see Chapter 8).

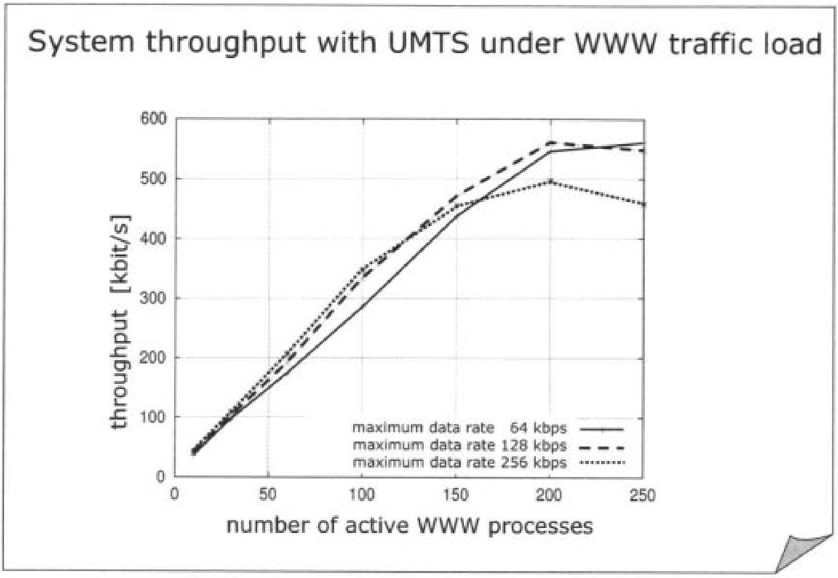

Figure 11.9: System throughput of UMTS with WWW traffic

System throughput is not of interest to the users of mobile radio networks: They are more concerned about the mean transmission rate available for their use. Figure 11.10 presents the results from an emulated *Frequency Division - Code Division Multiple Access* (FD-CDMA) system (UMTS) for users of the *World Wide Web* (WWW) service. The mean throughput available to a user when the number of mobile stations indicated are simultaneously running WWW processes is represented in kbit/s at the ordinate. The three curves apply to different transmission rates for the physical channels allocated to the users.

Each curve shows a kink that occurs when the capacity limits of the radio interface determined by the maximum allowable interference are reached. The interference occurs through the simultaneous transmission of data by different connections within the cell and by interference caused by transmission in neighbouring cells. If the respective kink point of the curves is exceeded, the transmission of some data has to be repeated due to the increasing interference-associated bit error ratio, and this reduces the traffic speed [38]. Besides the mean throughput, the variance of throughput is alos of concern. It has been found from the simulated UMTS system that the variance of the bit rate provided to a WWW process is extremely high and might result in below 10 [bit/s] and also above 100 kbit/s, dependent on the location of the terminal in the radio cell.

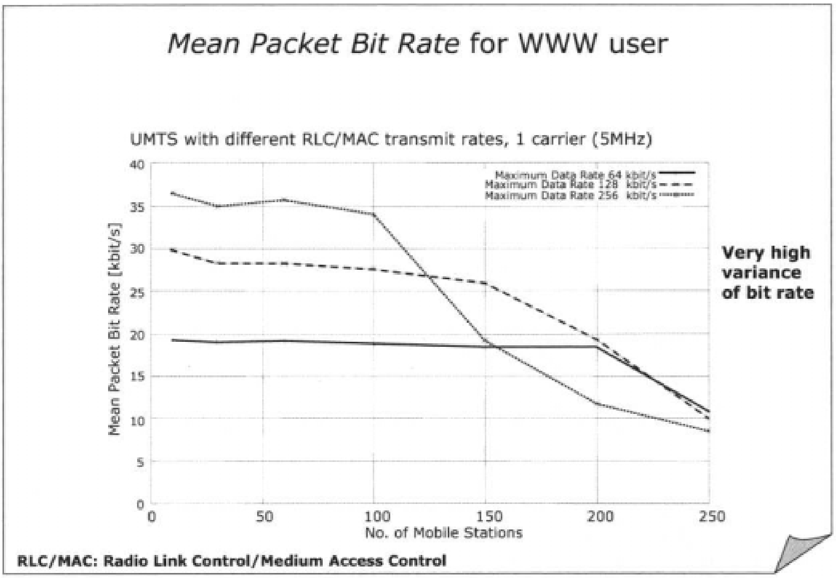

Figure 11.10: Mean packet data rates for users of WWW service

The limitations of 3G systems are low capacity and high licensing costs in some of the high potential regins. The reasons are as follows:

- Multimedia cannot be realised with up to 2 Mbit/s, contray to what is needed and was expected. Instead, 384 kbit/s can be seen as an exceptional high data rate available only to very few concurrent users in a radio cell.
- Owing to the high costs of the system few users will be able to afford the bills for a frequent use of high bit rate services.
- The entire capacity of a 10 MHz FDD band for both the downlink and the uplink in UMTS is much lower than the capacity of 2×17.5 or 2×22.5 MHz available with most GSM operators.
- A low number of users employing a high bit rate is less favourable in UMTS than a large number of users each using a low bit rate: UMTS is especially well suited for speech and lower data rate services such as multimedia short messages.
- The systems will not be able to offer the high bit rate, asymmetric services foreseen by the UMTS Forum and the ITU at a favourable cost.

- In the countries where the abnormal high licence fees have been paid, the cost to provide the 12.2 kbit/s voice service in UMTS is approximately six times higher than in GSM (but cannot be charged of course).
- The only option available to UMTS to make a profit is to offer low bit rate data services to a mass market. The respective applications are to still to be developed.

3G systems urgently require clearly more spectrum in the TV band (below 1 GHz) in order to be successful as a country-wide service. The current spectrum allocation makes it very difficult and costly to cover suburbs, motor ways and rural areas. In addition, W-LANs have to be integrated with UMTS to support multimedia traffic in densely populated areas.

11.7 Developments with W-LANs

Figure 11.11 shows the bands in Europe, the United States and Japan assigned to second generation Wireless LANs. The expected release will result in coexistence problems for the various systems due to the different W-LAN standards permitted.

Discussions are currently underway in Europe about which systems should be allowed in the bands and under what conditions. Approval for the systems are expected according to the allocation criteria developed in the UK in 2001: IEEE 802.11a and HiperLAN/2 will both be allowed if they apply *Dynamic Frequency Selection* (DFS) and *Transmitter Power Control* (TPC).

Exclusive licences are being assigned for part of the band (5570 MHz to 5725 MHz). Above 5725 MHz is a 100 MHz wide ISM band (industrial, scientific and medical usage, licence-free). *High Performance Radio Local Area Network Type 1* (HiperLAN/1) will not receive any exclusive bands. The large bandwidth planned for W-LANs in Europe will provide very large message traffic capacity due to the small radio range (cell size) possible with the specified transmitter power under the propagation conditions known to apply at the 5 GHz band.

The ETSI standard for W-LAN HiperLAN/2 is essentially based on the results of the project ATMmobil [15] that was based on the results of the European research project ACTS-MBS, see Section 3.3. The project objective was the development and prototype implementation of a high bit rate radio interface to extend the services of ATM (Asynchronous Transfer Mode) networks on a wireless basis to moving terminals, while achieving about the same quality of service as in the fixed network. Figure 11.12 shows the four applications scenarios being considered.

The cellular mobile broadband system shown in the upper left corner of Figure 11.12 works like a mobile radio system with mobility management and

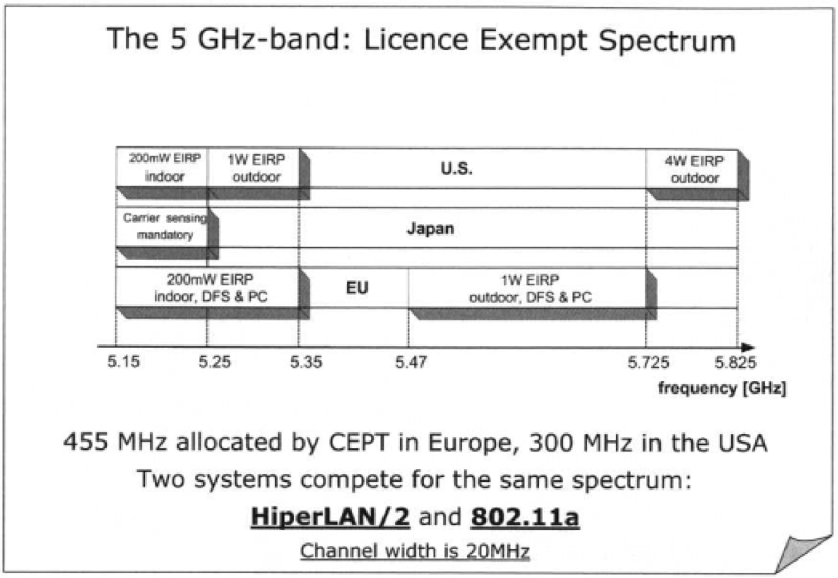

Figure 11.11: Spectrum allocation for W-LANs according to WRC 2000

handovers. It provides almost complete coverage to the area being supplied and enables continuous broadband communication at a 25 Mbit/s multiplex data rate for slow mobile terminals on a 20 MHz radio channel.

The point-to-multipoint radio relay system for bridging the last mile (radio local loop) enables buildings to be connected to the ATM fixed network and provides a multiplex data rate of 155 Mbit/s for this purpose, see lower left corner of Figure 11.12. The corresponding results are currently being incorporated into the standardisation of the ETSI/HiperACCESS system. Systems like this are also called *Wireless Metropolitan Area Networks* (W-MANs) or *Local Multipoint Distribution Systems* (LMDSs) and are aimed to operate at frequency bands like 5, 10, 17, 26, 30, 38 or 60 GHz.

A self-organising (plug-and-play) system was developed for the wireless connection of multimedia devices in the home. This system uses the radio interface of the cellular broadband system but has additional functions such as direct mode for direct transmission between terminals. The corresponding preliminary work has been adopted into the Home Extension of the HiperLAN/2 standard. The W-LAN scenario was developed for offices and factory automation. This too is supported by the same radio interface developed in cellular broadband systems. The Chair for Communication Networks (ComNets) at RWTH Aachen proposed the ATMmobil project, contributed to the concept for the radio interface, participated in all prototype implementations

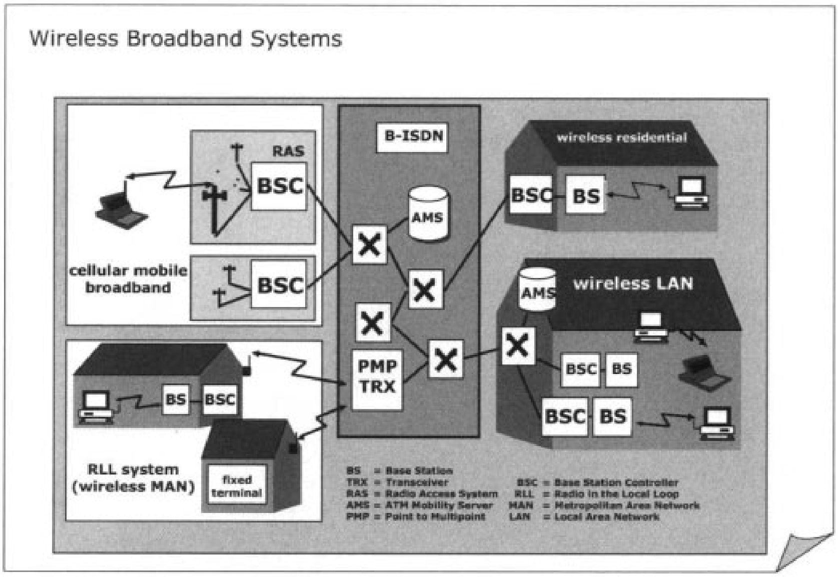

Figure 11.12: ATMmobil applications scenarios

of the four designs shown in Figure 11.12 and assumed responsibility for the technical project management.

The HiperLAN/1 standard was completed by ETSI one year before the HiperLAN/2 standard. No products are yet available for HiperLAN/2. At the same time the IEEE in the USA completed the W-LAN IEEE 802.11 standard, which provides a plug-and-play W-LAN for operation in the ISM band at 2.4 GHz. Due to the limited capacity of the ISM band because of interference by other systems operating in the same band (microwave ovens, Bluetooth), IEEE implemented a further development of Version 802.11a for operation in the 5 GHz band and completed the standard for it around the same time that ETSI provided the HiperLAN/2 standard, i.e. in the year 2000. HiperLAN/2 and IEEE 802.11a use the same transmission technique, a 52-carrier OFDM with adaptive modulation and coding in a 20 MHz channel. Each system has a different media access technique (see Figure 11.13), details are explained in [35].

HiperLAN/2 uses a centrally controlled reservation-based technique in which a terminal, usually the access point to the wired network, takes over the task of the central control entity and signals the MAC frame structure depending on the capacity requirements of the terminals to be served. The IEEE 802.11a standard describes a non-reservation-based *Carrier Sense Multiple Access with Collision Avoidance* (CSMA/CA) technique in which all terminals listen in

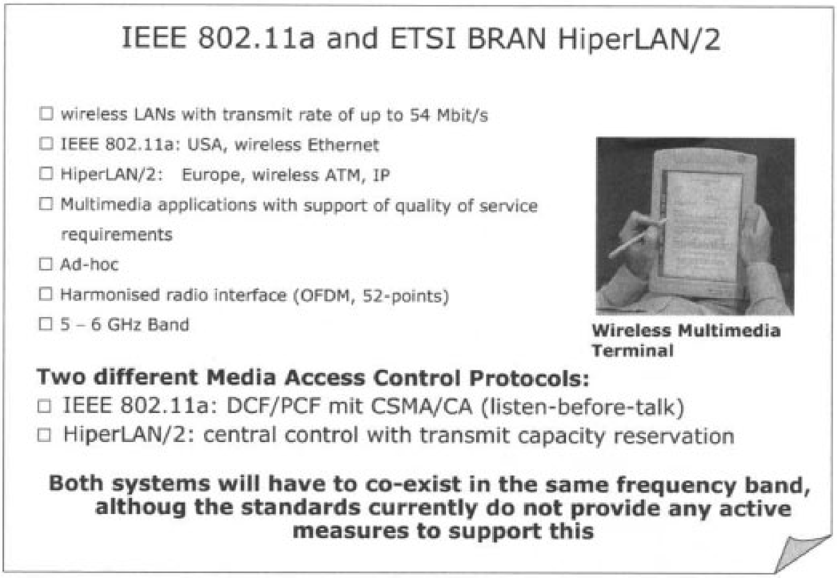

Figure 11.13: IEEE 802.11a and HiperLAN/2

on the medium before sending MAC frames, thus user data packets. The basic operation of the MAC protocol relies on the ***Distributed Coordination Function*** (DCF). This technique is very suitable for ad-hoc networks having the ability to self-organise their temporal or permanent functioning. An option of the IEEE 802.11a MAC protocol is the ***Point Coordination Function*** (PCF) where any station may take the role of a local coordinator of the packet traffic flow across the radio space. Thereby, real-time support can be given to services based on transmit capacity reservation to stations that have requested the PCF supporting station for that.

For the future it is necessary that W-LANs also make the quality of service supported in the fixed network (Internet) available on a wireless basis. HiperLAN/2 is fully equipped for this requirement; in the 802.11e group the IEEE developed an improvement to the existing standard orientated to the function principles of HiperLAN/2.

The HiperLAN/2 protocol reserves the radio medium for a 2 ms-long future time interval (the MAC frame) and through a central *Access Point* (AP) station allocates time slots (for the transmission of one or more ATM cells) for the uplink or the downlink to the terminals in its environment that are ready to transmit (see Figure 11.14). Each frame starts with an announcement of its structure by the *Period Control Protocol Data Unit* (PCtr-PDU). Using the request channels at the end of a MAC frame, the terminals can signal

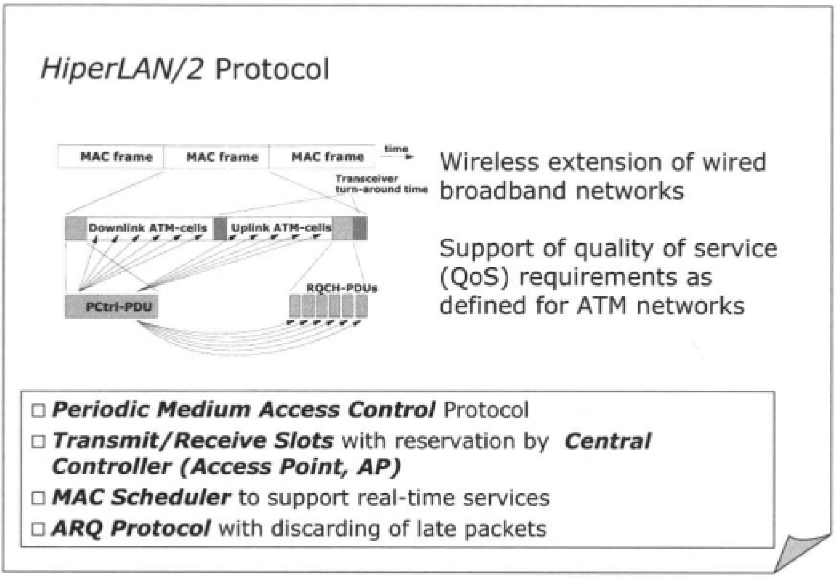

Figure 11.14: HiperLAN/2 MAC protocol

the AP that they wish to transmit in the next frame. Access to the *Request Channel* (RQCH) in the MAC frame is random-based. The capacity of the downlink and the uplink can be adapted as necessary to enable asymmetric traffic flows. An error-protection *Automatic Repeat Request* (ARQ) protocol is used above the MAC layer to detect transmission errors on the basis of a checksum and to request the retransmission of defective blocks.

The *Quality of Service* (QoS) is described by a parameter set containing values for performance characteristics such as throughput, delay, bit error rate, blocking probability and abnormal termination.

When an actual connection is established, numerical values are specified for the relevant performance characteristics; these are the values that are to be maintained for the duration of the connection to enable an appropriate use of the service. A distinction is made between two different classes of service (see the top three examples in Figure 11.15), each with certain characteristic requirements. The more restrictive the requirements of a service are (e.g., high throughput, minimal delays), the more complex it is for the network operator to provide the service and the higher the costs to the user.

Along with transmitter power control, *Link Adaptation* (LA) and *Dynamic Frequency Selection* (DFS) are suitable techniques for achieving an optimal coordination in the sense of quality of service, throughput and spectral ef-

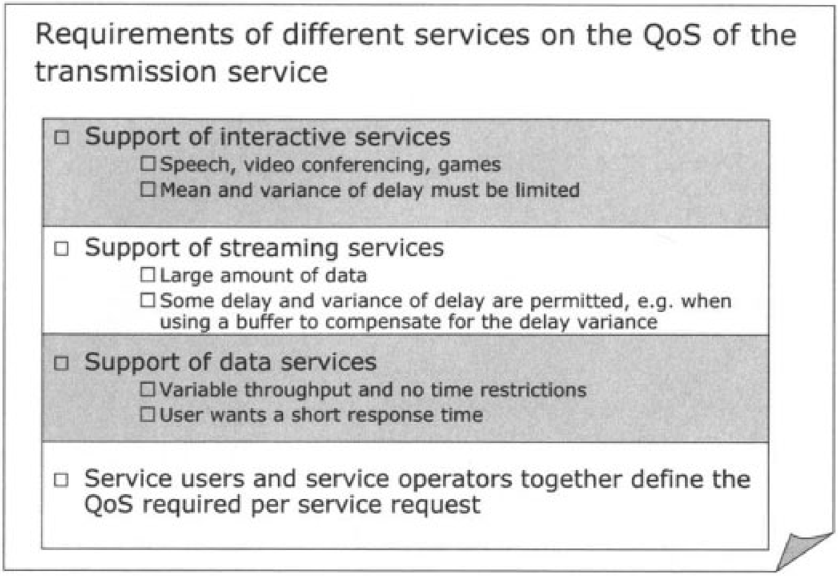

Figure 11.15: Requirements of different services for the quality of the transmission service

ficiency. It should also be noted that in addition to the two system types considered (IEEE 802.11a and HiperLAN/2), other radio systems whose operation should not be affected by unnecessary high emissions also operate in the 5 GHz frequency band, e.g. satellites.

Dynamic Frequency Selection and *Transmitter Power Control* will be able to contribute substantially to reduce the radio signal power to the level needed for a given communications relationship, thereby reducing the interference produced in other systems operating spatially distant on the same radio channel. For example, Figure 11.16 clearly shows that with a carrier-to-interference ratio of 15 dB a transmission mode that permits a bit rate of 9 Mbit/s has to be selected to achieve a *Packet Error Ratio* (PER) of 0.01.

Figure 11.17 (right) shows which range can be achieved with which transmitter power with *line of sight* or with shadowing (*obstructions*) using the most robust transmission mode BPSK plus code rate 1/2 resulting in a bit rate of 6 Mbit/s. What is noticeable is that very large cell radii are possible with line of sight connections but the ones with shadowing are very small, independent of the shown (maximum allowable) transmitter power. The graphics show the probability that with HiperLAN/2 a communications relationship is possible with one of the transmission modes (physical mode) in a given distance from the AP. For example, at a distance of 40 m data rates of 54 Mbit/s are possible

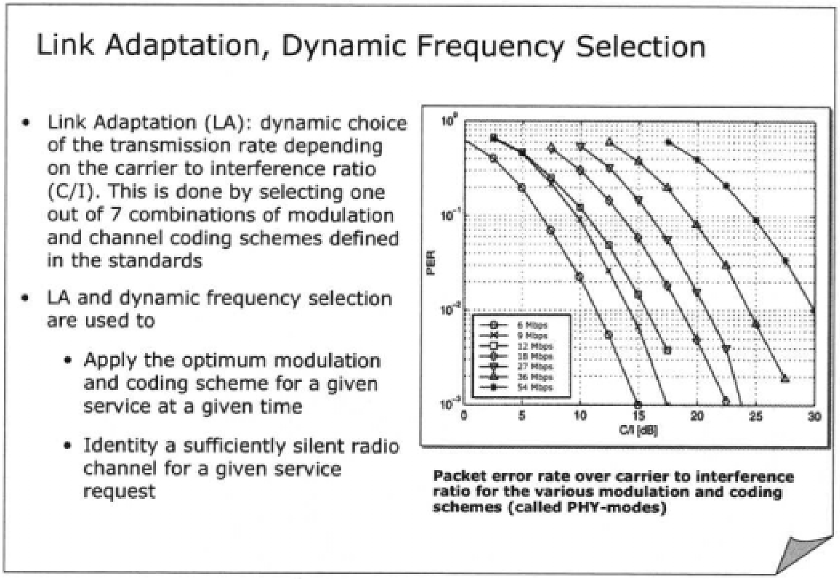

Figure 11.16: Link adaptation and dynamic frequency selection

in 40% of the cases, only 36 Mbit/s is possible in 40% of the cases and only 27 or 12 Mbit/s in the rest.

11.8 W-LANs in integrated radio networks

W-LANs will supplement cellular networks and allow high-speed multimedia communication. The services requiring high transmission rates predicted by the UMTS Forum have asymmetric traffic volumes and require a great amount of spectrum. These services or applications transport text, music and picture data that is stored and can be accessed over the fixed network. The corresponding mass of data usually does not have the same value per transmitted bit as speech data and in many cases a request for access does not require an immediate provision of the data as is the case in mobile radio networks. Instead of a user waiting a long time online for transmission over the mobile radio network, the accepted procedure is to wait off-line for a signal that the data is available in the terminal.

The conclusion is that W-LANs are a very suitable mechanism for providing these services at a particularly reasonable cost. At the same time there is less of a load on the mobile radio network. Figure 11.18 shows the scenario with the integration of 3G and W-LAN systems in a joint network introduced in

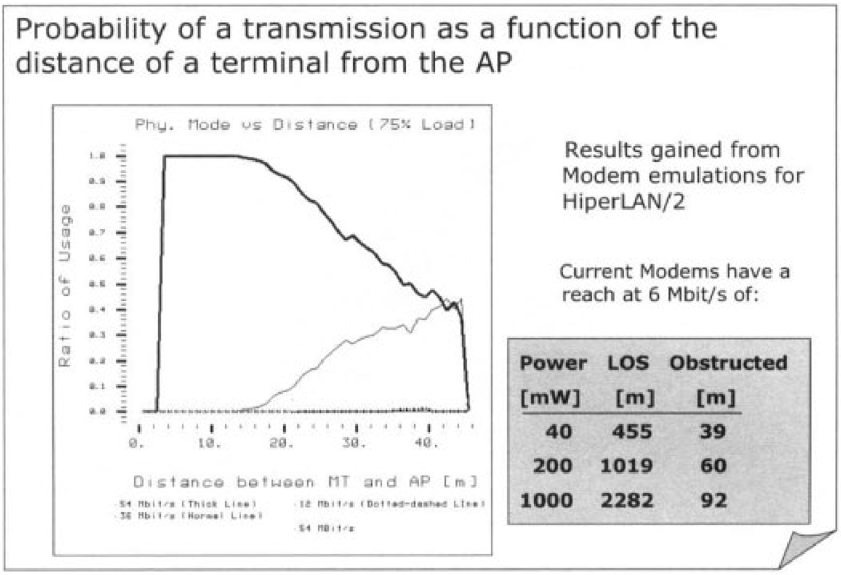

Figure 11.17: Range of HiperLAN/2 at 6 Mbit/s

the *Information Society Technology* (IST)/*Broadband Radio Access to IP Network* (BRAIN) project and reviewed in the IST/*Mobile Internet Networking Demonstrator* (MIND) project. A terminal with two different radio interfaces is required and is managed by the central mobility management function of the mobile radio fixed network. The most suitable radio access system is always used on a location-specific basis so that subscribers are offered the maximum quality of service.

The proposals of the BRAIN project provide for the availability of broadband services at locations with high user numbers, called "hot spots". A vertical handover to the 3G system takes place when a user leaves a W-LAN-supplied area. The current assumption is that W-LAN technology will be based on the ETSI/*Broadband Radio Access Network* (BRAN) HiperLAN/2 standard. The W-LAN standard IEEE 802.11 b/a is also under discussion. The integration of systems such as DVB-T, alongside *Personal Area Networks* (PANs) such as Bluetooth, in a joint network architecture would constitute another step forward (see Figure 11.19).

Figure 11.19 shows a layered architecture of the various heterogeneous radio networks combined here, including the following technologies [16]:

- Digital radio broadcasting
- 2G cellular networks, e.g., GSM und GPRS

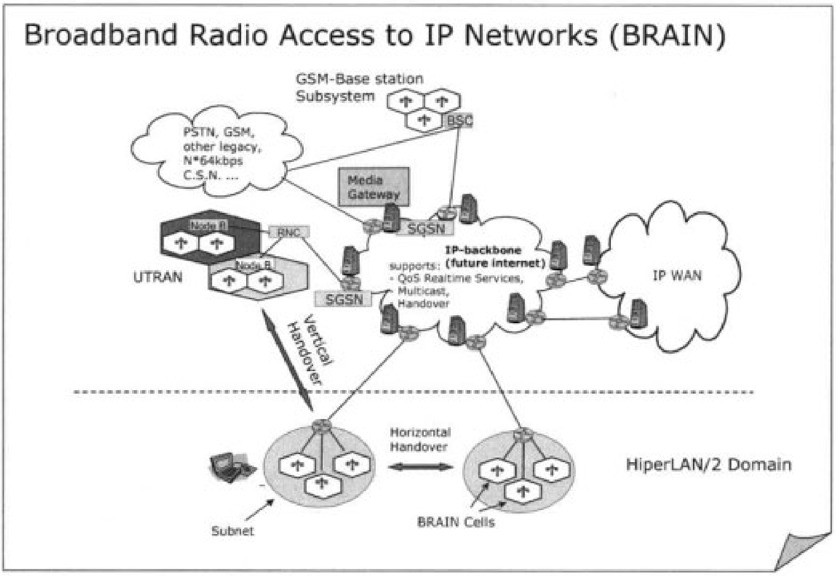

Figure 11.18: BRAIN architecture

- 3G cellular networks, e.g., UMTS
- W-LANs for use at locations with high public traffic (hot spots)
- Personal radio networks like Bluetooth and others.

The architecture shown is already realisable today because the individual components are available and only have to be combined in an appropriate format. This requires a considerable effort but vendors and network operators will be receptive to this idea as soon as they are convinced of the usefulness of the architecture. The radio interfaces that were specially developed for the different movement of speeds and services will then appear as access networks to a combined universal mobile radio network and be supported by a shared mobile radio-specific fixed network. This fixed network, in cooperation with the public telecommunications networks, will then provide subscribers with universal mobile access to all services familiar from the fixed networks. In addition, the strengths of mobile networks will appear in the form of the availability of all services that are only meaningful for mobile users, namely narrowband real-time and low-speed data, while high-speed data is provided via the non-mobile network part based on W-LANs.

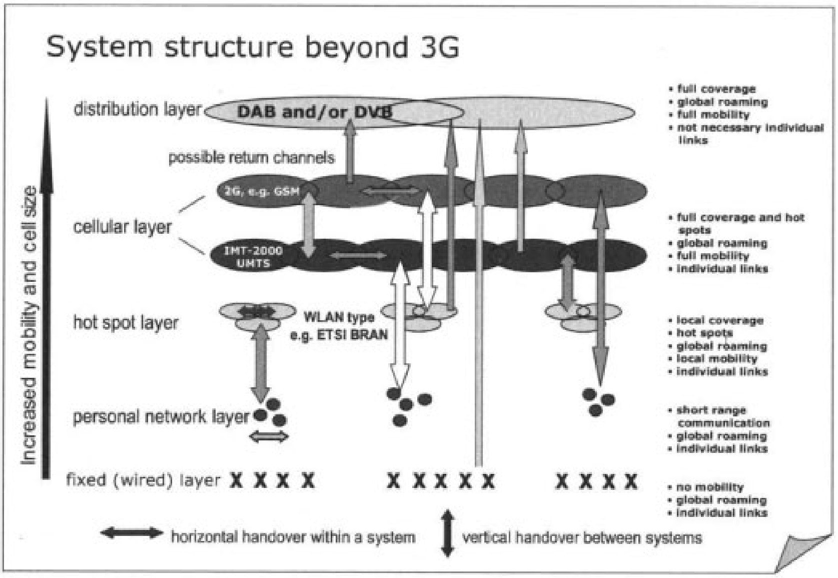

Figure 11.19: System structure beyond 3G

11.9 The wireless media system

Based on what has been explained so far, it is not hard to imagine that access to telecommunications networks and the Internet over *Wireless Base Stations* (WBSs), e.g. W-LANs will be restricted to locations with high levels of public traffic. Plus, this access will be implemented through a sufficiently dense network of distributed WBS access points (each of which forms an isolated cell). These WBS cells do not produce contiguous radio coverage but are activated or controlled in such a way that they enable virtually complete local radio coverage for access to the corresponding services of high bit rate fixed networks by wireless terminals. This access allows the terminals to use and transmit multimedia content. WBS might even evolve to become a Mobile Broadband System that is currently not possible to implement since HiperLAN/2 and IEEE 802.11a do not really allow for medium or high speed of movement.

This so-called *Wireless Media System* (WMS) [34] is similar to the proposed concept called Infostation System [12]. The difference between the WMS and Infostation is that the WMS is an integrated part of the mobile radio networks and relies on its mobility management support, including accounting and localization. This integration of WMS and mobile radio would provide many interesting advantages in comparison with the concepts of WBS access to IP

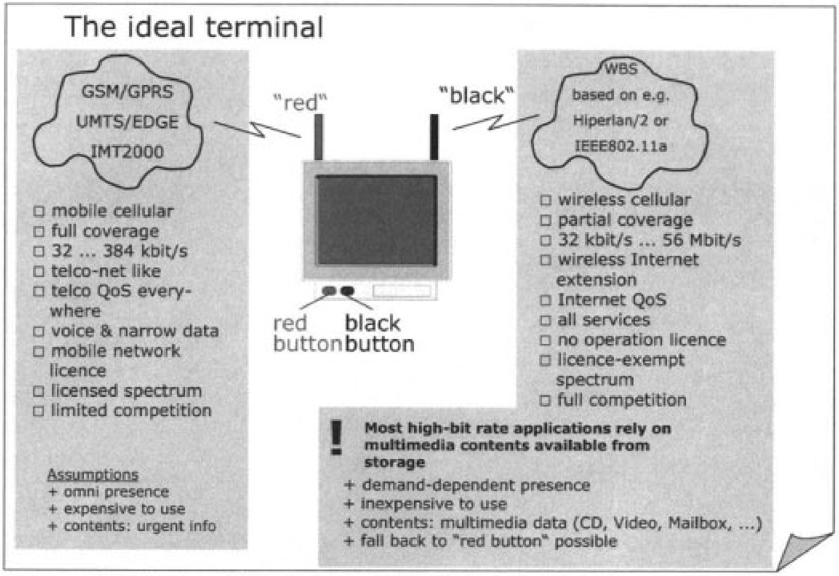

Figure 11.20: The ideal terminal

networks that are already known or are in the process of being introduced, as shown in Figure 11.20.

Figure 11.20 shows a terminal with two operating buttons "red" and "black" (representing the internal control for the use of a service and the respective network). These operating buttons control the use of two radio interfaces based on the current needs of the user and the requirements of the respective service, along with the usage costs. Cellular networks and WBS-based networks are incorporated as the transport or service platforms.

The networks are characterised according to their respective features. Cellular mobile radio networks offer low bit rate services with real-time features and are associated with relatively high usage costs.

W-LANs offer high bit rate services to the Internet with the appropriate quality of service and low usage costs.

Considering the high licensing fees for second and third generation mobile radio networks in Europe, along with the basically very high set-up and operation costs of these networks, users appear to find it attractive to be able to take advantage of the strengths of both technologies on a situation-specific basis. WBSs are expected to transmit in the licence-exempt spectrum, see Figure 11.11.

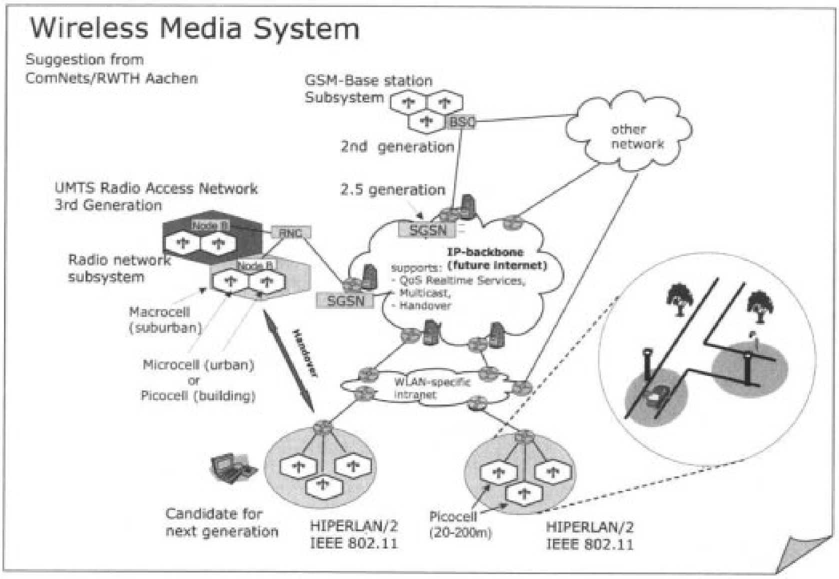

Figure 11.21: Architecture of MediaPoint systems

Yet users will probably as often as possible access WMS-based services that appear like high bit rate network access, available either locally or elsewhere. This is similar to users going to a newspaper stand, a petrol station or a telephone call box but instead of entering a concrete infrastructure, they only get close enough to it (up to a few hundred feet outdoors) so they can have wireless communication. In fact, the density of WBS cells in populated areas might be high enough that no specific decision on terminal movement is needed to have more or less permanent WMS access.

Figure 11.21 illustrates the integration of cellular mobile radio systems with their accompanying *Internet Protocol* (IP)-supported fixed networks (*core network*). It also shows how the proposed WMS, which is supported by its own Intranet [35, 36], can be integrated into a common core network. The assumption is that the terminals support at least one mobile radio and one WBS interface. The integration can be a close one in which the Intranet of the WMS is part of the core network of the mobile radio network. Alternatively, the integration can be loose in which case the two fixed network parts are coupled together for the exchange of control information. It is also conceivable that the two classes of systems could be operated independently of one another. In this context a mobile terminal could accept two *Subscriber Identity Modules* (SIMs) that contain contractual data for the mobile radio network or the WMS.

The vision of the BRAIN project regarding the integration of mobile radio networks and WBSs provides for vertical handovers. The suggestion is that handovers should be dispensed with and connections allowed to be cut off and instead quickly re-established in the new system if the systems concerned do not have shared mobility management. This is likely to be the case when the two networks are operated separately. With multimedia applications a handover from WMS to the mobile radio network means a drastic reduction in quality of service so that a user will be unhappy with or without a handover. When a handover takes place in the other direction, the current application should decide when the handover from the mobile radio network to the WMS takes place. It can then prevent an active connection from being cut off since the mobile radio network has area-wide radio coverage including locations that have WMS radio coverage.

Transmission in the proposed WMS is bi-directional between its base stations called *Media Access Points* (M-APs) connected to the fixed network and the *Mobile Terminals* (MTs). The MTs have a very large data storage facility for media data to cache the high-speed incoming data for later medium to low-speed consumption. Interactive services in real-time, such as voice and video communication, can obviously be supported over a WMS as long as the MT is operating in the range of the radio coverage of an M-AP. Although a WMS can be operated autonomously and independently of a mobile radio network, some applications benefit considerably if the system is in loose or even in close cooperation with such a network; however, for some applications this can be a disadvantage.

The following WMS features are noteworthy:

- Operation in a licence-exempt frequency band and use of a standardised radio interface, e.g., HiperLAN/2 or IEEE 802.11 x.
- Wireless transmission of media data with high bit rates at the air interface with the M-AP linked over a radio relay system, cable or optical fibre to an Intranet that connects to the WMS-specific multimedia server available.
- The M-AP generates certain local radio coverage.
- The number of M-APs selected in the service area is sufficient to cope with the anticipated density of mobile terminals and their message traffic.

Overall, the individual M-APs together do not achieve a continuous area-wide radio supply to a coverage area; the radio supply is instead scattered. A coverage area can be very large and include, for example, major motorways or a city or a densely populated metropolitan area. Smaller coverage areas are sports facilities, airports, train stations, and city centres.

Each M-AP has a very large storage capacity (as intermediate storage, cache memory) to enable communication with the MT to take place with the full transmission rate of the radio interface and not with the possibly limited transmission rate of the Intranet.

A special service control function in the MT and in the Intranet ensures that medial content loaded over the network is continuously available to a subscriber even if the radio supply is interrupted from time to time. This service control simulates a continuous connectivity to the WMS for the transmission of media data by buffering the medial content in the M-APs and MTs.

Spontaneous access to media data is typically executed with a situation-specific delay, since the respective MT must wait until it has reached a radio supply area with access to the WMS.

If the MT reaches an M-AP, it then refers to the session already established earlier with the WMS operator, receives the medial data it had requested earlier at a very high data rate from the cache memory of the respective M-AP and stores it in local mass storage for later use. The quantity of data transmitted is so large that a long enough time horizon is covered, e.g., half an hour, to accommodate the expected duration of the local processing (e.g., mailbox contents) or local usage. The MT transmits all data waiting for transmission to the M-AP as soon as it reaches its coverage area.

MTs can use all services known from cellular networks and the Internet, i.e., voice, data transmission, reception of broadcast transmissions and can operate interactive multimedia connections.

The subscriber uses his or her MT as usual. As part of the application being used, the MT sends commands to the WMS specifying which media data should be transmitted next. The M-AP most suitable for the transmission to the MT is determined either by the MT or by the WMS. The WMS refers to geographical information or movement patterns of the MT that it recognises as being typical of the respective MT, e.g., because they originate from previous observations. Localisation of the MT can also be supported from the cellular mobile radio system.

The user processes or consumes the data available in local storage in the MT without a radio connection to the WMS. When the next M-AP is reached, the processing results are transmitted by wireless over the WMS to the destination address.

One interesting feature of the proposed system is that cellular mobile radio systems can be used (over their radio interface) at any time to send urgent data immediately if necessary or to be available for urgent incoming communication requests. Likewise data can be requested over a mobile radio network for intermediate storage with the next M-AP a user will reach in the near future. Alternatively, the data can be stored with several M-APs operating in the vicinity of the MT for later loading in the MT when it reaches an M-AP. Data cached at M-APs will be cleared anyway after a timer has expired.

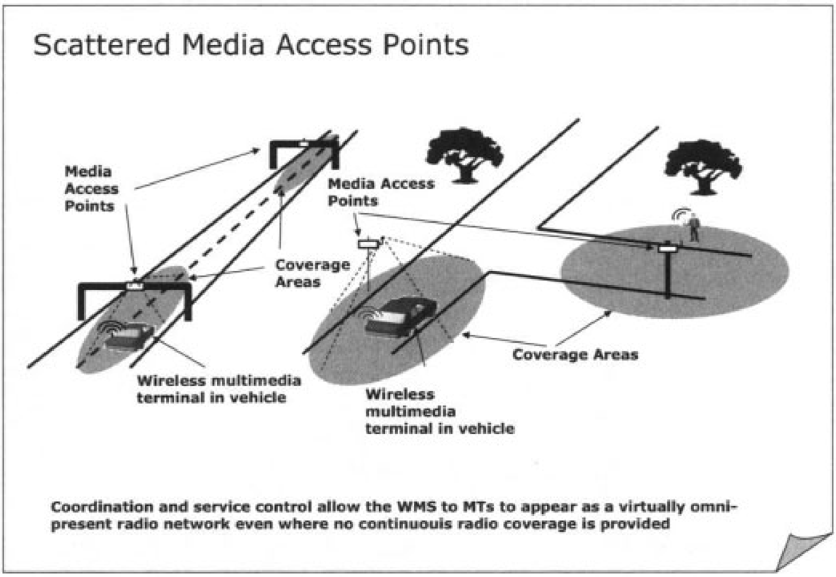

Figure 11.22: Media Access Points of the WMS for user coverage

If the M-AP uses directional antennas instead of omni-directional ones, the radio supply can be limited to areas in which MTs normally operate or move, e.g., roads that are used by vehicles with MTs or city areas for portable MTs. The coverage range of the radio waves is then increased and less interference power generated that could possibly cause interference to other radio cells of the WMS.

An example of the use of an WMS is shown in the scenario on the left in Figure 11.22. Two M-APs are mounted with their transmitting and receiving equipment at two traffic signal gantries of a motorway. Each M-AP has a control unit that is linked to the Intranet of the WMS or the local multimedia services under the control of an internal network service control.

So long as a car (or other vehicle) finds itself in the coverage area of an M-AP, the MT inside the car can set up a connection to request applications data from the WMS, load it wirelessly or send data such as emails or video recordings it has produced in the meantime. Once the car leaves the coverage zone, the connection is broken and cannot be resumed until the next M-AP is reached. The currently existing air interfaces of WBSs like HiperLAN/2 and IEEE 802.11x do not support a high-speed of movement. 50 km/h appears to be the upper speed limit today limiting the applicability of this scenario to urban usage.

A second scenario on the right in Figure11.22 shows the transmitting and receiving equipment (antennas) of M-APs mounted to the masts of street lamps in a residential area. Users in cars and those walking can set up a radio connection in the coverage area of the M-AP so long as they remain in that area. The connection is resumed when the next coverage zone is entered.

The 15-minute video (300.000.000 bytes) in Figure 11.4 gives an idea of the type of content that can be transmitted in a short amount of time over the radio interface of an M-AP.

With the transmission rate of 25 Mbit/s available at an M-AP, the 20 minutes that UMTS needs to transmit the video would be reduced to 1.6 minutes; this is the amount of time a car waits at a signal-controlled intersection. The video referred to could be part of a film that is then played and consumed offline in the MT. The next section of the film (another 15 minutes of video scenes) can then be loaded down from the network at the next M-AP. If the downloading process is successfully synchronised at successive M-APs, then the subscriber can view the entire film without any breaks when he or she leaves her car.

The M-AP of a WMS can, for example, transmit on the basis of the HiperLAN/2 (H/2) (or IEEE 802.11 x) standard and be sited in the middle of an intersection in a city. As Figure 11.23 shows, the radio coverage of the M-AP reaches the area shown in the light shading, namely a complete illumination of the intersection where it is located and a very far-reaching (up to 1 km) illumination of the streets off of it as long as no shadowing obstacles exist. It also reaches all vehicles parked or travelling there as well as people on foot. Due to the quasi-optical radio wave propagation, the side streets off the next intersection are not supplied and are therefore shown with dark shading.

Many M-AP sites each with access to the fixed telecommunications network are needed if H/2 or comparable WBS systems are to be used to supply radio coverage to a city. The number of sites corresponds to the number of base stations required by a 3G mobile radio system. The system will typically aim for base station sites with wave propagation below the roof tops so it can deal with large traffic capacity in a city. This means that the number of sites required for 3G mobile radio systems and the WMS are about the same if the WMS uses the technology shown for supplying coverage (this can change as is explained later).

The large number of radio standards added as a result of the introduction of 3G systems raises thoughts about what users really need in order to use the services available to them. Users obviously have to be equipped so they can make telephone calls anywhere and transmit data at a preferably high rate. As matters stand, GSM with its supplementary services GPRS and EGPRS is the only network able to provide comprehensive geographical coverage in Europe and beyond. The GSM interface is therefore a must.

Terminals that support more than one radio interface are called multi-mode mobile terminals. Not only can they be operated in different frequency bands

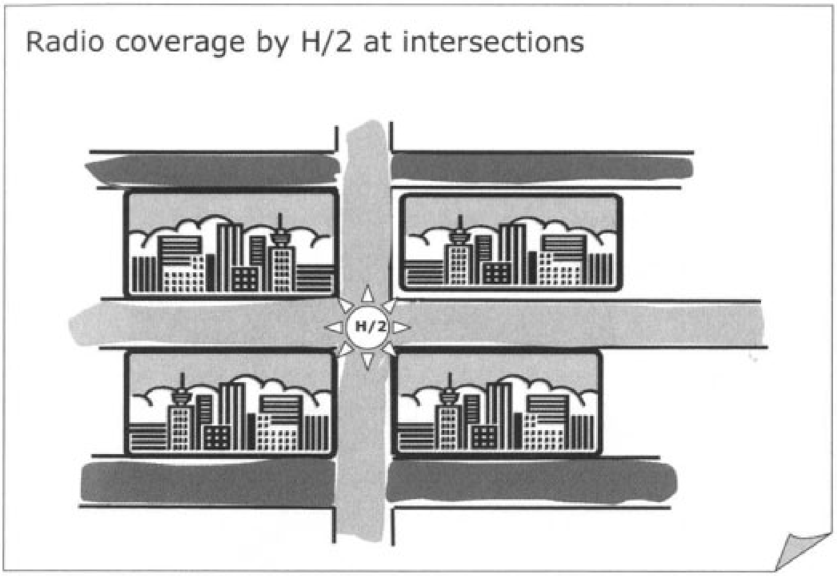

Figure 11.23: WBS radio coverage at intersections

such as 900, 1800 and 1900 MHz that are customary with GSM, but they also support different radio standards. Each additional radio standard supported in a mobile terminal increases the complexity of the terminal and the power consumption. Consequently, radio operators and equipment manufacturers will limit themselves to the absolutely necessary radio standards, at least for the terminals produced in large quantities for the mass market.

If a mobile terminal is small, then a data interface with a mean transmission rate that can be provided through Bluetooth or WBS 802.11 (in the 2.4 GHz band) is sufficient. The advantage is that the frequency range being covered by the MT will be still limited; this will result in low-cost transceivers.

GSM/(E)GPRS is also essential for high-performance portable terminals. In addition, a WBS radio interface operating at 5 GHz will be provided and Bluetooth then eliminated because its capabilities are included with the WBS.

In the time frame being considered, the UMTS radio interface will only be used in high-end mobile terminals in the upmarket price range to provide interference-free mobile radio data operations in local areas with very high traffic volumes "hot spots" to customers with "business tariff" contracts. From time to time other users in the same location will notice that the network is overloaded and only be able to use data rates that are lower than at other locations.

It would be sensible to make the two classes of mobile terminals compatible with one another, e.g., with the small terminal capable of being docked to the big one and taken along when the user wants to travel light. In this case the same user number could easily be used for both devices and the GSM/GPRS interface would not have to be implemented many times.

11.10 Multi-hop and ad-hoc broadband communication in WBSs

The regulatory conditions for WBSs at 5 GHz provide for performance limitations of 200 mW indoors and 1 W outdoors. Due to path loss the range of the radio waves is very limited. At 5 GHz there is very little wave diffraction and the reflected waves are so heavily attenuated that radio supply without line of sight links is only possible in the close vicinity of a station.

WBS standards have functions that are not found in mobile radio systems and therefore can improve radio supply. Stations of WBSss can organise themselves spontaneously into local networks with the stations able to exchange information with one another as needed. If one of the WBS stations has access to the core network, then it can help other stations to reach the network also. This entails the data packets being transmitted many times over radio over the WMS - referred to as multi-hop communication. The participating (or some) stations must then be relay-enabled, i.e., able to buffer data packets arriving over radio links and to forward them over radio links. A WBS does function like an Internet router.

Centrally controlled systems like HiperLAN/2 and IEEE 802.11 (with its central Point Coordination Function) can form area-wide local cellular networks. When mobility management is introduced to a core network part, communication relationships remain intact, even during a cell change, by being forwarded (handover) or re-established under network control. This kind of handover is referred to as a horizontal handover because it is implemented within the same system compared to a vertical handover in which the standard used by the radio interface is changed simultaneously with the handover.

If a terminal leaves the coverage area of a cellular WBS system, then the communications relationship can be continued after a vertical handover if cross-system mobility management that encompasses the WMS and the mobile radio network is available. This relationship can continue even with a reduced quality of service because a mobile radio network typically has a much lower transmission rate than a WBS. For the reason just mentioned, some experts are of the opinion that the introduction of such systems will make the vertical handover superfluous. The problem with vertical handovers is that they produce a drastic degradation in the quality of service for most services, sometimes coming close to an interruption in the service. In the

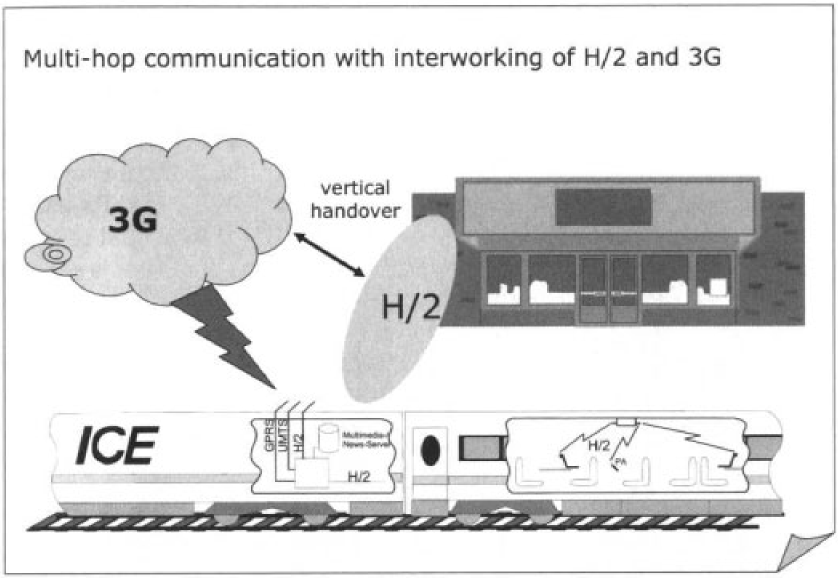

Figure 11.24: Interworking of 3G and HiperLAN using Multi-Hop Communications

other direction of a vertical handover from a mobile radio network to a WMS the suitable time instant for a vertical handover can be selected, e.g., at the end of the communication phase of a session. There is then enough time for a connection to be established in the WMS (without a handover). Another variation of combining the use of WMS and 3G systems with or without multi-hop communication is shown in Figure 11.24.

The best system that airplanes, ships, trains, buses and similar transport carriers with large numbers of passengers can offer their customers "on board" is a WBS operating similar to the wired branch exchange systems in hotels. A connection with the outside world can then either be made from time to time when the MT reaches the coverage area of a M-AP of the WMS or continuously with the quality of service available with the 3G system. In each case a communications relationship exists over two serial radio links (*hops*), one in the transport vehicle and the second in the form of radio access to a fixed network.

The solution selected will depend on the type of service being used: 3G systems will be used for voice communication and important low bit rate data services that essentially have to be available immediately and therefore are also allowed to cost more. For services involving mass data, users will normally wait until (in the case of a train or a bus) they reach a train station or a Media Access Point at an intersection with a traffic light to transmit over a second WBS hop to a M-AP and back. In some cases it is possible to dispense with the first hop in the vehicle and directly search for the contact to the outside M-AP through a control of the terminal (as described above).

The number of M-APs required in a given service area can be minimised through the option of a considerable reduction in infrastructure investment at the cost of a heavier load on the radio medium. With M-APs based on HiperLAN/2 technology wireless base stations (WBS) can be used to increase substantially the coverage area of an H/2 Access Point (AP) that is connected to a fixed network (see Figure 11.25). A *Wireless Relay Station* (WRS), which presents itself like a mobile terminal *vis-à-vis* the AP but acts like an AP to its environment, is used at the boundary of the coverage area of the AP. Consequently, the WRS connects all mobile terminals operating in its coverage area to itself over the first hop and with the AP over a second hop. The principle is cascadable so that a third hop also would be possible.

As a result the available spectrum for a communications relationship with an n-hop connection is loaded n times. This means that the spectrum capacity is divided by n. So long as sufficient spectrum capacity is available locally, the n-hop communication can help towards saving infrastructure investment in the form of fixed network connections that would be needed to connect APs placed at the locations of WRSs.

The selection of terminals with a WRS function can be planned and then provide stationary WRS sites. In special cases some mobile stations could conceivably contain the WRS function, and when they provide relay services for third parties collect credits from WMS operators that they can use later for their own communication in the WMS.

Multi-hop communication makes sense when sufficient spectrum is available locally at a favourable cost. If local traffic increases, the spectrum capacity will reach its limit, more APs will be installed and a migration made to pure one-hop communication to free up spectrum.

In the example shown in Figure 11.26 two-hop communication is provided using H/2 relays (WRS) to illuminate the side streets with the dark shading in the example in Figure 11.23. In this example the H/2 relays are mounted at certain intersections next to an AP and obtain their power supply on site but would have no access to the fixed communications network (this is how a savings is achieved *vis-à-vis* the use of APs at relay locations!). Of course, directional antennas at the relay would contribute much to bridging larger distances between APs and relays. In addition, much higher capacity at the relay is possible then, owing to the higher physical transmission mode applicable, see Figure 11.16.

It is plausible that the principle shown, based on a centrally sited AP supplemented by WRSs, could also be applied to the vertical parallel roads so that

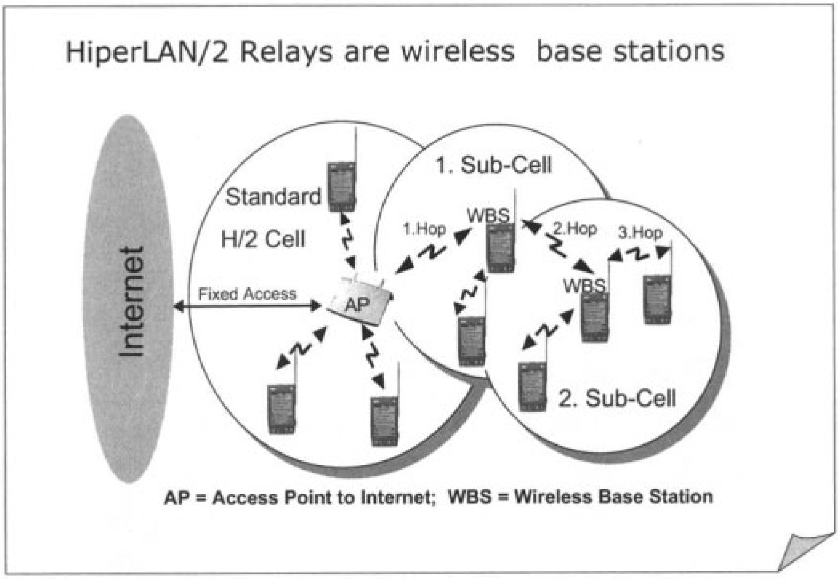

Figure 11.25: HiperLAN/2 relays as wireless base stations

the city road network is provided with a relatively high level of coverage (i.e., presence of WMS).

Private WBS systems would be set up inside buildings and link wireless terminals either directly to the Internet or over an indoor WRS to the outdoor WMS.

11.11 Conclusion

This chapter has described the visions of the next generation of wireless communication systems, yet only from the perspective of a short time period of three to five years.

For instance, no consideration has been given to the fact that 3G systems will continue to evolve and achieve a clearly higher level of spectrum efficiency (kbit/s per MHz and km^2) than is possible today. Technical options based on intelligent antenna systems and systems offering simultaneous transmission of multiple communication relationships over the same radio channel (Multiple Input Multiple Output) are in development right now and have the potential to make this goal a reality. However, the level of processing required of signal processors in mobile terminals is still comparatively much higher than

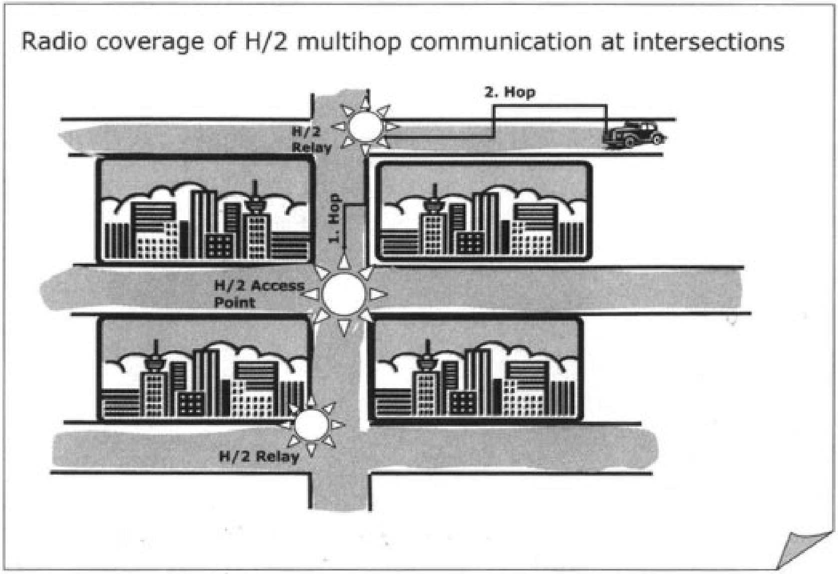

Figure 11.26: HiperLAN/2 radio coverage with multi-hop

now and underlines the urgent need for other technologies for mobile energy supply. Although the batteries available today are being developed further, it is not expected that they will be able to provide the level of power capacity that will be required. Fuel cells have potential in this area but are far from being market-ready, and, therefore, very considerably limit the development possibilities for wireless communication.

What also has not been covered is how new transmission technologies could make mobile radio coverage more omnipresent, thereby achieving a high quality of service (throughput, delay parameters, bit error ratio, and so forth). Coded multi-carrier techniques (*Coded Orthogonal Frequency Division Multiple Access* (COFDMA)) are being looked at again in this regard. This was an option that was not considered at the time the 3G standard was specified due to the linear broadband receiver that is very difficult to implement, but today this problem seems to have been solved. It appears much easier to implement high-speed downlink transmission to highly mobile terminals than high-speed uplink transmission. According to Figure 11.6 this is exactly what the mobile applications would need, so that it can be expected that systems beyond 3G will extend the high-speed service supply from urban to suburban and even rural areas. Spectrum availability will be the dominating parameter to make this possible development happen or not.

The final point to note is that the electromagnetic impact on the environment due to the use of WBS systems will be reduced considerably. There are two main reasons for this: the transmitter power of APs amounts to a maximum of 1 W and is consequently half the amount of that of the mobile terminals being used today and the AP antenna is effective from a greater distance than close to the head of the user. Due to the screens necessary for media services, the associated wireless terminals also will not be operated at the user's ear, and, in spite of higher transmission speeds, the biological effect will be noticeably less.

Acronyms

3GPP	*Third Generation Partnership Project*
3GPP2	*Third Generation Partnership Project 2*
AAL2SAR	*ATM Adaptation Layer 2 Segmentation and Reassembly*
ACF	*Autocorrelation function*
ACTS	*Advanced Communications Technologies & Services*
AICH	*Acquisition Indication Channel*
AMPS	*Advanced Mobile Phone Service*
AMR	*Adaptive Multi-Rate*
AP	*Access Point*
ARIB	*Association of Radio Industries and Businesses*
ARQ	*Automatic Repeat Request*
ASIC	*Application Specific Integrated Circuit*
ATDMA	*Advanced TDMA*
ATM	*Asynchronous Transfer Mode*
BCCH	*Broadcast Control Channel*
BCH	*Broadcast Channel*
BER	*Bit Error Ratio*
BMBF	*Bundesministerium für Bildung und Forschung*
BMC	*Broadcast and Multicast Control*
BPSK	*Binary Phase Shift Keying*
BRAIN	*Broadband Radio Access to IP Network*
BRAN	*Broadband Radio Access Network*
BSC	*Base Station Controller*
BTS	*Base Transceiver Station*
CAC	*Call Admission Control*
CAMEL	*Customised Application of Mobile Enhanced Logic*
CCCH	*Common Control Channel*
CCF	*Cross-correlation function*
CCPCH	*Common Control Physical Channel*
CCTrCH	*Coded Composite Transport Channel*
CDMA	*Code Division Multiple Access*
cdma2000	*Code Division Multiple Access 2000*
CIR	*Carrier to Interference Ratio*
CN	*Core Network*
CODIT	*Code Division Testbed*
COFDMA	*Coded Orthogonal Frequency Division Multiple Access*
COMCAR	*Communication and Mobility by Cellular Advanced Radio*

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CPCH *Common Packet Channel*
Common Part Convergence Sublayer

CPICH *Common Pilot Channel*

CRNC *Controlling RNC*

CSCF *Call State Control Function*

CSMA/CA *Carrier Sense Multiple Access with Collision Avoidance*

CSMA/CD *Carrier Sense Multiple Access with Collision Detection*

CTI *Computer Telephony Integration*

CWTS *Chinese Wireless Telecommunications Standards*

DAB *Digital Autdio Broadcasting*

D-AMPS *Advanced Mobile Phone Service*

DCCH *Dedicated Control Channel*

DCF *Distributed Coordination Function*

DCH *Dedicated Channel*

DECT *Digital Enhanced Cordless Telecommunications*

DFS *Dynamic Frequency Selection*

DPCH *Dedicated Physical Channel*

DPCCH *Dedicated Physical Control Channel*

DPDCH *Uplink Dedicated Physical Data Channel*

DRIVE *Dynamic Radio for IP-Services in Vehicular Environments*

DRM *Digital Rights Management*

DRNC *Drift RNC*

DS *Direct Sequence*

DSCH *Downlink Shared Channel*
Direct Sequence Spread Spectrum

DTCH *Dedicated Traffic Channel*

DTX *Discontinuous Transmission*

DVB-T *Digital Video Broadcasting Terrestrial*

DwPCH *Downlink Physical Synchronisation Channel*

DwPTS *Downlink Pilot Tone Slot*

EDGE *Enhanced Data Rates for GSM Evolution*

EFRC *Enhanced Full Rate Codec*

EGPRS *Enhanced General Packet Radio Service*

E-GSM *Enhanced GSM*

EIRP *Equivalent Isotropic Radiated Power*

ETSI *European Telecommunications Standards Institute*

EU *European Union*

FACH *Forward Access Channel*

FBI *Feedback Information*

FD-CDMA *Frequency Division - Code Division Multiple Access*

FDD *Frequency Division Duplex*

FDM *Frequency Division Multiplex*

FDMA *Frequency Division Multiple Access*

FEC *Forward Error Correction*

FER *Frame Error Rate*

FMA *FRAMES Multiple Access*

FPACH	*Fast Physical Access Channel*
GERAN	*GSM/EDGE Radio Access Network*
GGSN	*Gateway GPRS Support Node*
GMSC	*Gateway Mobile Services Switching Centre*
GMSK	*Gaussian Mean Shift Keying*
GPRS	*General Packet Radio Service*
GSM	*Global System for Mobile Communications*
GTP	*GRPS Tunnel Protocol*
GTP-u	*GRPS Tunnel Protocol User Part*
HiperLAN/1	*High Performance Radio Local Area Network Type 1*
HiperLAN/2	*High Performance Radio Local Area Network Type 2*
HLR	*Home Location Register*
HRC	*Half Rate Codec*
HSCSD	*High Speed Circuit Switched Data*
HSDPA	*High Speed Downlink Packet Access*
HSS	*Home Subscriber Server*
I-CSCF	*Interrogating Call State Control Function*
IEEE	*Institute for Electrical and Electronics Engineers*
IETF	*Internet Engineering Task Force*
IMT-2000	*International Mobile Telecommunications at 2000 MHz*
IMTS	*Improved Mobile Telephone Service*
IN	*Intelligent Network*
IP	*Internet Protocol*
IPv4	*Internet Protocol Version 4*
IPv6	*Internet Protocol Version 6*
IS-136	*TIA Interim Standard 136*
IS-54	*TIA Interim Standard 54*
ISDN	*Integrated Services Digital Network*
ISM	*Industrial, Scientific and Medical*
ISO	*International Organization for Standardization*
IST	*Information Society Technology*
ITU	*International Telecommunications Union*
LA	*Link Adaptation*
LA	*Location Area*
LAI	*Location Area Index*
LCD	*Long Constraint Delay*
LLC	*Logical Link Control*
LMDS	*Local Multipoint Distribution System*
MAC	*Medium Access Control*
M-AP	*Media Access Point*
MBMS	*Multimedia Broadcast/Multicast Service*
MBS	*Mobile Broadband System*
ME	*Mobile Equipment*
MExE	*Mobile Execution Environment*
MGW	*Media Gateway*
MIMO	*Multiple Input Multiple Output*

MIND *Mobile Internet Networking Demonstrator*

MM *Multimedia*

MMI *Man-Machine-Interface*

MMS *Multimedia Message Service*

MS *Mobile Station*

MSC *Mobile Services Switching Centre*

MT *Mobile Terminal*

MTS *Mobile Telephone Service*

OFDM *Orthogonal Frequency Division Multiplex*

OMC *Operations and Maintenance Centre*

OSA *Open Service Architecture*

OSI *Open System Interconnection*

OVSF *Orthogonal Variable Spreading Factor*

PAN *Personal Area Network*

PCCH *Paging Control Channel*

P-CCPCH *Primary Common Control Physical Channel*

PCF *Point Coordination Function*

PCG *Project Coordination Group*

PCH *Paging Channel*

PCM *Pulse Code Modulation*

PCPCH *Physical Common Packet Channel*

P-CSCF *Proxy Call State Control Function*

PCtr-PDU *Period Control Protocol Data Unit*

PDC *Personal Digital Cellular*

PDCH *Packet Data Channel*

PDCP *Packet Data Convergence Protocol*

PDP *Packet Data Protocol*

PDSCH *Physical Downlink Shared Channel*

PER *Packet Error Ratio*

PHS *Personal Handyphone Service*

PHY *Physical Layer*

PICH *Paging Indication Channel*

PNBSCH *Physical Node B Synchronisation Channel*

PRACH *Physical Random Access Channel Payload Unit*

PSC *Primary Synchronisation Code*

P-SCH *Primary Synchronisation Channel*

PSE *Personal Service Environment*

PSK *Phase Shift Keying*

PUSCH *Physical Uplink Shared Channel*

QoS *Quality of Service*

QPSK *Quaternary Phase Shift Keying*

RA *Routing Area*

RACE *Research, Analysis, Communication, Evaluation*

RACH *Random Access Channel*

RAN *Radio Access Network*

RANAP *Radio Access Network Application Part*

RegTP *Regulierungsbehörde für Telekommunikation und Post*

RLC	*Radio Link Control*
RNC	*Radio Network Controller*
RNS	*Radio Network Subsystem*
RNSAP	*Radio Network Subsystem Application Part*
RNTI	*Radio Network Temporary Identifier*
RQCH	*Request Channel*
RR	*Radio Resource*
RRC	*Radio Resource Control*
RRM	*Radio Resource Management*
R-SGW	*Roaming Signalling Gateway*
RWTH	*Rheinisch-Westfälische Technische Hochschule*
SAT	*SIM Application Toolkit*
S-CCPCH	*Secondary Common Control Physical Channel*
SCH	*Synchronisation Channel*
S-CSCF	*Serving Call State Control Function*
SGSN	*Serving GPRS Support Node*
SID	*Silence Descriptor*
SIM	*Subscriber Identity Module*
SIP	*Session Initiation Protocol*
SIR	*Signal to Interference Ratio*
SNDCP	*Sub-Network Dependent Convergence Protocol*
SP	*Signalling Point*
SRNC	*Serving RNC*
SRNS	*Serving Radio Network Subsystem*
SSC	*Secondary Synchronisation Channel*
STP	*Signalling Transfer Point*
T1	*T1 American standardisation committee*
TBS	*Transport Block Set*
TCP	*Transport Control Protocol*
TD-CDMA	*Time Division - Code Division Multiple Access*
TDD	*Time Division Duplex*
TDM	*Time Division Multiplex*
TDMA	*Time Division Multiple Access*
TD-SCDMA	*Time Division - Synchronised Code Division Multiple Access*
TF	*Transport Format*
TFC	*Transport Format Combination*
TFCI	*Transport Format Combination Identifier*
TFCS	*Transport Format Combination Set*
TFS	*Transport Format Set*
TIA	*Telecommunications Industry Association*
TPC	*Transmitter Power Control*
TS	*Technical Specification*
TSG	*Technical Specification Group*
TTA	*Telecommunication Technology Association*
TTC	*Telecommunication Technology Committee*
TTI	*Transmission Time Interval*
UDD	*Unconstrained Delay Data*
UDP	*User Datagram Protocol*
UE	*User Equipment*
UL	*Uplink*

UMTS	*Universal Mobile Telecommunication System*
UPP	*User Plane Protocol*
URA	*UTRAN Registration Area*
USAT	*UMTS SIM Application Toolkit*
USCH	*Uplink Shared Channel*
USIM	*UMTS Subscriber Identity Module*
UTRA	*UMTS Terrestrial Radio Access*
UTRAN	*UMTS Terrestrial Radio Access Network*
UpPCH	*Uplink Physical Synchronisation Channel*
UpPTS	*Uplink Pilot Tone Slot*
VAD	*Voice Activity Detection*
VHE	*Virtual Home Environment*
VLR	*Visitor Location Register*
VoIP	*Voice over IP*
WAP	*Wireless Application Protocol*
WAP-NG	*WAP Next Generation*
WARC	*World Administrative Radio Conference*
WBS	*Wireless Base Station*
WCDMA	*Wideband Code Division Multiple Access*
WG	*Working Group*
W-LAN	*Wireless Local Area Network*
W-MAN	*Wireless Metropolitan Area Network*
WMS	*Wireless Media System*
WRC	*World Radio Conference*
WRS	*Wireless Relay Station*
WSP	*Wireless Session Protocol*
WTO	*World Trade Organisation*
WTP	*Wireless Transaction Protocol*
WWW	*World Wide Web*

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Answers to questions

Answers to questions for Chapter 3

3.1 In addition to GSM mainly IS-95 (CDMA), IS-136 (TDMA) and PDC.

3.2 There was comparatively little distribution of Internet access in Japan at the time i-Mode was introduced. Consequently, the Japanese are doing many things with i-Mode that are being handled over the Internet in Europe. Furthermore, i-Mode uses a flavour of HTML for the content, which simplifies the conversion of contents from the Internet. Finally, additional graphical elements are introducing the possibility of creating fancy portal sites.

3.3 Sophisticated technology, small and reasonably priced terminals, a mass market, early introduction, usable in many countries, very open standard with built-in extension possibilities.

3.4 The following is available based on voice telephony: low-rate *circuit-switched data services* (CSD), 9.6 kbit/s); *high-speed circuit switched data* (HSCSD), up to 54 kbit/s);(*general packet radio services* (GPRS), up to 115 kbit/s); an improved radio interface through *Enhanced Data Rates for GSM Evolution* (EDGE), up to 384 kbit/s).

3.5 More than 50 million customers in Germany are currently using GSM-based services, especially voice telephony. The number of GSM subscribers worldwide is expected to reach 1 billion by 2003. It is not anticipated that these customers will be switching to UMTS in the short term since GSM completely supports most of their requirements. GSM is constantly being developed and along with GPRS and EDGE will be in a position to support most data applications. Consequently, it is expected that UMTS and GSM will coexist for a long time to come and that the GSM radio access network represents an alternative to UTRAN.

3.6 The terms for the licences vary from 2009 to 2016.

3.7 IMT-2000 systems:

- High data rates (144 kbit/s *indoors* up to 2 Mbit/s *outdoors*)
- Symmetric and asymmetric data transmission to support IP-based services

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- Line-switched and packet-switched data transmission
- High level of voice quality, comparable to the fixed network
- High level of spectral efficiency
- Service portability (VHE, Virtual Home Environment)
- Global access

3.8 Cdma2000, DECT+, UWC136/EDGE, TD-SCDMA (now part of UMTS as a low chip rate option of TDD mode in R4).

3.9 Cdma2000 is downwardly compatible with existing IS-95 networks. Cdma2000 will play an important role since a significant basis of IS-95 networks is installed in the US.

3.10 The first systems are being built in accordance with R99. It is unclear to what extent the hardware can continue to be used after a changeover to R4 or R5. A great deal depends on the hardware platform selected by the respective vendor. However, some vendors have hinted at the probability that first generation hardware, except for the 19" cabinets, will have to be replaced in accordance with R99 when a switch is made to R4.

3.11 Some network operators want to start the network as early as the second quarter of 2002. However, the terminals will not be available until late 2002 or perhaps even in 2003, so a widespread start of UMTS is not anticipated until then. The network operators also still need time to build the infrastructure in heavily populated areas.

3.12 The population's increasing criticism of the erection of antenna sites is preventing network operators from acquiring the locations they need for starting networks in heavily populated areas. This is now being regarded as a serious threat to the timetable.

There is also a fear that the vendors of the terminals will have problems due to the high complexity of these terminals. Since end users are used to small, reasonably priced and reliable terminals with a long battery life from GSM, the UMTS terminals will have to offer the same features despite the fact that they are considerably more complex. This can jeopardise the availability of terminals.

Another problem exists because some network operators were not able to shoulder the high financial burden of the licences and infrastructure for long and therefore could not build the networks. Several licences were auctioned off at high prices in the USA in the 1990s, yet the network operators could not start with the networks due to the high costs involved.

3.13 unpaired (TDD): 1900–1920 MHz, 2010–2025 MHz, paired (FDD): 1920–1980 MHz for the uplink, 2110–2170 MHz for the downlink.

3.14 Three extension areas have been identified: below the GSM900 band at 2.5 GHz and at 2.7 GHz. The advantage of the band below the GSM900 band is the lower attenuation of the radio waves due to physical conditions. As a result, a given area can be illuminated with fewer infrastructure elements. A relatively large block exists at 2.5 GHz that is relatively close to the 3G bands used today. This block is therefore also being regarded as an ideal candidate. With continuing growth the neighbouring band at 2.7 GHz can also be used.

Answers to questions for Chapter 4

4.1 Due to the separation of functions in RAN and CN, it is basically possible to replace the RAN and retain the CN. Thus the use of GSM/EDGE RAN (GERAN) is being planned along with UTRAN. The use of Hiperlan/2 or even IEEE 802.11 is also conceivable in the future.

4.2 In the old networks in the USA the identity of a user was permanently associated with his or her terminal. When a user changed terminals the network operator always had to update the databases and face a considerable administrative task. With user-related data and keys being transferred to a SIM card, users can now change terminals without the network operator being involved. Furthermore, network operators are now able to implement their own encryption algorithms in the SIM card.

4.3 The advantage of packet-switched transmission is that users can directly use the capacity not utilised by other active users. This is called “statistical multiplexing”. Since packet-switched services share the same physical transmission medium they influence each others service quality. If a packet-switched connection requires a certain quality of service the protocol used for packet-switched transmission must be able to deal with this problem, e.g. by managing priorities for different connections.

4.4 A RAN consists of Node Bs and RNCs.

Functions of an RNC:

- Admission control
- Autonomous radio resource management
- Call set-up and call clearing
- Outer loop power control
- Channel allocation with packet data services

- Handovers
- S-RNS relocation (change of Iu-reference point)
- Encryption
- Protocol conversion (Iu (RANAP), Iub, Iur (RNSAP))
- ATM, SS7
- O&M tasks.

Functions of Node B:

- Like BTS with GSM, additional inner loop power control, generation of measurement reports for the RNC, softer handovers (microdiversity).

4.5 HLR and VLR are cascaded database systems. The statistical information for a user is filed in an HLR. When the user moves into a cell, the data is copied from the HLR into the VLR responsible for the respective region and stored there temporarily. At the same time a reference to the currently applicable VLR is entered in the HLR. When the location area changes, only the VLR is updated. Access is therefore not constantly required to the HLR, which otherwise would soon be overloaded.

4.6 An MSC is used for the transmission of circuit-switched services, whereas an SGSN is used for packet-switched services. MSCs normally "communicate" between themselves through SS7, whereas an SGSN is addressed over IP.

4.7

- Hard handover: immediate changeover from one cell to another (GSM, UTRA-TDD)
- Soft handover: simultaneous radio connection to two or even three different base stations
- Softer handover: simultaneous radio connection to the different sectors of a base station.

Advantages: Resistance to shadowing, lower fading reserve required, avoidance of near-far effect.

4.8 When the Node Bs involved in a soft handover are attached to different RNCs, the original RNC takes over the control function (SRNC) and also supervises the Node Bs connected to the new RNC over the Iur-Interface and the new RNC. The new RNC (DRNC) forwards the data received to the SRNC. There all incoming data is merged together and sent to the CN (*combining, splitting*).

4.9 With packet-switched services there are more calls (data packets) per time unit that require the set-up of a radio connection than is the case, for example, with circuit-switched voice calls. So that unnecessary signalling can be avoided, the network must have a more exact knowledge of the location of the mobile station than is necessary with circuit-switched services.

4.10 A group of cells is combined into what is called a *location area* (LA). Each cell in the LA beams a code called a *location area code.* As soon as a mobile station recognises that it is receiving a new location area code, it updates the databases. The same applies to packet-switched services (*routing area update*).

The network can also be configured so that a mobile station has to register periodically with the network. This is intended to prevent a situation in which a mobile station leaves a supply area and the network does not register the fact. Incoming calls would otherwise generate signalling at the radio interface to no avail.

4.11 The GTP-U transmits data within a network between GGSN, SGSN and RNC. This enables external and internal IP traffic to be separate from one another.

4.12 A network operator can actually use a purely IP-based infrastructure for an easy and flexible introduction of new services. However, controversy does surround this architecture and some issues still need to be resolved. For example, it is not clear whether the new network nodes are really less expensive or whether the latency for voice connections is sufficient.

4.13

- New network nodes required at the outer boundaries of the IP network
- Possible latency problems for voice telephony
- Security aspects

4.14 SIP is used for the set-up and configuration of IP-based calls. The user data itself is transmitted on a direct path between terminals (via SGSN and GGSN).

Answers to questions for Chapter 5

5.1 In the ISO/OSI layer model each layer supplies the layer above it with certain services. To do so, it accesses the services of the layer below it. The lowest layer is called the physical layer and is linked directly to the physical layer of the communication partner. The other layers can only communicate with one another indirectly over a logical connection.

Through the separation into layers functions are divided into blocks that are easier to manage. This facilitates the orientation and implementation. The alternative would be a large monolithic block that would be more difficult to implement and maintain.

5.2 The philosophers can exchange ideas and information via their staffs. A logical connection therefore does not exist.

5.3 The C-plane or *control plane* takes over the management and evaluation of control information between the layers or between partner entities of the same layers. The actual information data for a user is managed and exchanged in the U-plane or *user plane.*

As a higher ranking part in the protocol stack, the M-plane or *management plane* manages the organisation of a protocol stack. This part is normally not standardised and is implemented differently by the different vendors.

5.4 The UTRA protocol stack comprises layers 1, 2 and 3.

5.5 The physical layer is responsible for the transmission of information over a physical medium. It is therefore responsible for channel coding, forward-error correction and interleaving as well as adaptive power control. It also measures different quality parameters and supplies the measured values to the RRC layer. The physical layer in UTRA-TDD is different from the one in UTRA-FDD.

5.6 Transport channels.

5.7 Priority control/scheduling, monitoring of traffic volume, encryption of data in transparent RLC mode, allocation of data transmitted over so-called *shared channels.*

5.8 Segmentation, flow control, encryption (*acknowledged mode, unacknowledged mode*), error correction.

5.9 In *acknowledged mode* data blocks detected as being defective are requested again. In *unacknowledged mode* defective data blocks are discarded. No error protection occurs at the RLC level in transparent mode.

5.10 Transmission errors can be handled in the following ways:

Error detection merely checks the accuracy of the received data and forwards this as additional information with the data that is unchanged. This requires a check sum for each data block.

With *error correction* transmission errors are actually corrected. This involves an addition to the original data stream of systematic redundancy, which is used in conjunction with appropriate algorithms in the receiver to enable the correction of individual errors.

Repeated transmission using *Automatic Repeat Request* (ARQ) protocols is a procedure that can be attached to an error detection procedure. When a transmission error is detected, it can be used to request and retransmit the defective data block again.

Finally, it is possible for a data block detected as being defective simply to be discarded or for no error detection to be implemented at all. This is sensible when the data being used has to be forwarded soon after the application and the application itself is robust against errors. This is the case, for example, with voice data.

5.11 In transparent RLC mode the MAC layer is responsible for data encryption. In all other cases encryption is a part of the RLC layer.

5.12 The PDCP sublayer adapts the UTRA protocol stack to the packet data protocols supported by the UTRA protocol stack. The functions of the PDCP sublayer include TCP/IP-*header compression*. This gives the PDCP sublayer a special role in mobile Internet access.

5.13 The RRC layer records the measured values of all other layers in the protocol stack. These values are processed in algorithms within the RRC layer. Decisions are made on this basis and forwarded to the individual layers through configuration messages.

5.14 The UMTS standard mainly describes the interfaces but less so the algorithms. Consequently, vendors can use efficient algorithms to increase the performance of their RANs. This especially applies to the *radio resource control* algorithms in the RRC layer (in the RNC). The performance of the radio interface can therefore vary from vendor to vendor.

5.15 The physical layer is embedded in Node B; the other layers belong to the RNC. In between is an ATM line that connects the physical layer with the MAC and RRC layers.

5.16 The UMTS radio interface offers a high level of flexibility. The other layers in the UTRA protocol stack must be able to work with this flexibility so that applications can also benefit from it. Consequently, a complicated but in many areas flexible mechanism was developed for the transmission of transport blocks over the transport channels.

5.17 Transport channels describe how user or control data should be transmitted. The key parameters are data quantity and the time frame when this data arises. The data quantity is determined by the size of the data blocks or *transport blocks* (TB) and the number of them within a *transport block set* (TBS). In UMTS the interval for a TBS is referred to as a *transmission time interval* (TTI). Transport channels are provided by the physical layer to the *medium access control* (MAC) sublayer above it.

Logical channels, on the other hand, are provided by the MAC sublayer to the RLC sublayer. Logical channels describe which type of data is being transmitted thus, for example, the user data of a voice connection over a DTCH or signalling information over a DCCH.

5.18 The MAC layer can transmit a DTCH over various transport channels. In addition to the DCH, these include the CPCH (FDD), the RACH (uplink), the FACH (downlink), the USCH (TDD) and the DSCH. The efficiency of the different transport varies depending on the characteristics of the transmitted data.

Thus a limited quantity of user data can be included with the data transmission over the RACH since the data packet is already transmitted at the time the medium is accessed and no lengthy signalling is necessary. However, due to poor power control of the transmitter interference is higher than with a DCH so that the latter is more efficient when the volume of data reaches a certain level.

5.19 This question naturally does not have a precise answer. Experts operate on the assumption that the complexity for a protocol stack is higher by a factor of 10. However, this excludes the new types of algorithms such as adaptive antennas, etc. Added to this is the additional complexity due to video codecs, graphical user interfaces, etc.

Since complexity is associated with high calculation levels, one of the most difficult tasks is reducing the power consumption of components enough so that a terminal's operating time is acceptable to users.

Answers to questions for Chapter 6

6.1 The duplex technique is used to separate transmitting signals on the uplink and the downlink; the multiple-access technique separates the signals of different users.

6.2 An asymmetric service is a data service in which different quantities of data exist for each of the transmission directions (downlink or uplink). One example is the Internet service with its high volume of data on the downlink.

The TDD duplex technique enables available resources to be shared out on an asymmetric basis through a shifting of the switching point within a TDD time frame. The FDD duplex technique, on the other hand, provides for a paired spectrum with equally large frequency bands for both transmission directions.

6.3 According to the Shannon formula
$D = B * \log_2(1 + R)$
provides the maximum transmission rate
$D = 5\,\text{MHz} * \log_2(1 + 15)$
$D = 20\,\text{Mbit/s}$
for the given values.

6.4 All multiple-access procedures are basically equal when it comes to achievable capacity, i.e., achievable throughput. The advantage of CDMA is its high level of flexibility.

6.5 With the equal distribution in FDMA systems each user receives exactly one-tenth of the data rate, thus
$D_{\text{user}} = {}^{D}/_{10}$.

In TDMA systems each user receives the full data rate, albeit only for one-tenth of the TDMA frame because the remaining portion is allocated to other users, thus
$D_{\text{user}} = D * {}^{1}/_{10}$.

With CDMA systems all users transmit their data simultaneously in the complete frequency band. However, a sufficient number of different codes must be available so that the users can be distinguished from one another. This also reduces the data rate according to
$D_{\text{user}} = D * {}^{1}/_{10}$.

6.6 The modulation scheme affects the amount of user information transmitted per modulation symbol. The transmission rate for the information is generally higher than the modulation rate. UMTS uses QPSK modulation. With this modulation scheme two information symbols are transmitted per modulation symbol. This means that a modulation rate of 3.84 Msymbol/s corresponds to a transmission rate of 7.68 Mchip/s. This is frequently referred to as complex-valued information symbols.

6.7 Since all users within the same frequency band transmit at the same time, several interferers with comparable transmitter power occur for each transmit signal. The resulting S/N is then usually ≤ 1.

6.8 One "bit" is an element of a data stream of user data. One "chip" is an element of a spreading code. With the CDMA technique one bit is mapped to several chips.

6.9 Through the transformation of the bit sequence into a higher rate chip sequence the data signal is spread in the frequency domain. Since this spreading becomes obvious in the frequency spectrum, the expression "spread spectrum" technique is used.

6.10 In CDMA systems "orthogonality" is the characteristic spreading codes have, enabling them to be separated from one another. The product of two spreading codes added up over a period of time equals zero (ideally orthogonal). In this case the value of the cross-correlation function (CCF) of both spreading codes is also zero. The spreading gain enables the user signal to be extracted again from the noise signal. Furthermore, the interference signals of other users are eliminated or at least considerably weakened by the orthogonality of the spreading code.

6.11 In FDMA systems the carrier functions used can be multiplied together and over a period of time produce a zero.

In TDMA orthogonality is found in the time domain. Here individual users transmit one after another so that only one carrier function is ever different from zero when all other carrier signals are deactivated.

6.12 With the spreading process each bit is allocated a chip sequence of constant length. Each ***individual*** bit is therefore mapped to ***several*** chips. The use of orthogonal spreading codes enables an ideal separation of the different data streams.

Scrambling helps to provide a further distinction between the spread data streams. Here quasi-orthogonal chip sequences are used to map *one* chip of the spread data stream in turn to *one* chip according to a certain pattern. In the FDD mode of UMTS this separates the sum of chip streams of the different users. In TDD mode each cell uses its own specific scrambling chip sequence so that the different cells are separated from one another through the scrambling.

6.13 Variable data rates can be achieved through a variation of the spreading factor if the transmission chip rate is maintained at a constant level. With a constant bandwidth and constant modulation, however, it is precisely this transmission chip rate that is fixed. Different data rates can therefore only be achieved through the help of different spreading factors or through the simultaneous use of several code channels.

6.14 Because two participants use spreading codes with the spreading factor SF $= 4$, exactly half of the OVSF code tree is unavailable to the other users. With the spreading factor SF $= 16$ a maximum of only 8 codes can be used, i.e., precisely 8 users can transmit data.

Two participants with the spreading factor SF $= 4$ then always have a transmission rate of 500 kbit/s.

Eight participants with the spreading factor SF $= 16$ then always have a transmission rate of 125 kbit/s.

6.15 Multiple access interference refers to the interference power experienced by a user due to the transmit signals of other users in the cell.

6.16 The power of the user signals ***at the receiver*** must be at the same level for all n users in a cell. This way each user has the same interference signal part of $n-1$ users and always the same level of the user signal part.

6.17 The near-far effect is the effect created due to higher path loss when signals from remote users arrive at the receiver with lower power than the signals from users nearby. Without transmitter power control the multiple access interference is then considerably higher for remote users than for those close by.

Answers to questions for Chapter 7

7.1 Compare Figures 7.3–7.5.

7.2 The transmission bandwidth in FDD systems is $2\times$ 5 MHz – always with one band each for the uplink and for the downlink. The complex chip rate is 3.84 Mchip/s. Taking the QPSK modulation into account, the maximum data rate achieved with a spreading factor of SF = 4 is

$$D_{max} = \frac{3.84 \times 2}{4} = 1.92\,\text{Mbit/s}.$$

The maximum spreading factor, which in FDD mode is different for the uplink and the downlink, is decisive for the minimum data rate. For the downlink

$$D_{min,DL} = \frac{3.84 \times 2}{512} = 15\,\text{kbit/s}.$$

Whereas the calculation for the uplink is

$$D_{min,UL} = \frac{3.84 \times 2}{256} = 30\,\text{kbit/s}.$$

In TDD mode the bandwidth is 5 MHz and the complex chip rate is 3.84 Mchip/s. QPSK modulation is used here too. Since a maximum of 14 time slots may be used for a transmission direction, the following results from a minimum spreading factor of SF = 1

$$D_{max} = \frac{3.84 \times 2}{1} \cdot \frac{14}{15} = 7.168\,\text{Mbit/s}.$$

The minimum data rate resulting from the use of only one time slot for the spreading factor SF = 16 is

$$D_{min} = \frac{3.84 \times 2}{16} \cdot \frac{1}{15} = 32\,\text{kbit/s}.$$

7.3 The advantage of a multiple-switching-point configuration within a TDD time frame is that there can still be a reaction to requests from mobile stations (over the uplink) within the same frame, i.e., data can be transmitted on the downlink.

The disadvantage of this configuration is the additional hardware effort, because switching over the transmission direction means switching off the receiving branches and switching on the sending branches or vice versa.

7.4 The functions of the physical layer include:

- provision of transport channels
- mapping of transport channels to physical channels
- implementation of error protection procedures
- segmentation and multiplexing of data streams
- synchronisation
- transmitter power control
- measurement of channel parameters and connection quality
- execution of soft handovers.

7.5 Transmitter power control reduces the energy consumption of terminals, which increases the operating time. Furthermore, there is a reduction in the radiated power and consequently system interference. An appropriate adjustment of the transmitter power prevents the near-far effect.

7.6 There is no difference in the PC techniques between UL and DL.

7.7 A different technique is defined in TDD mode than for FDD mode or the TDD downlink. Only on the TDD uplink is the control loop between base and mobile station open. This means that no control information that is used directly to control the transmitter power for the uplink is sent from a base station to a mobile station. In this case the control loop algorithm itself estimates the transmitter power to be set for the mobile station from measured values and target guidelines.

7.8 A target value for the *Carrier-to-Interference* (C/I) ratio is specified by the outer control loop. This enables the transmission quality to be adapted to the requirements of different services (voice, video, data), the current propagation environment (city, countryside) and the transmission mode used (spreading factor).

7.9 The *Coded Composite Transport CHannel* (CCTrCH) is the designation for a virtual channel that merges together all transport channels of a user within the physical layer so they can then be distributed again to the available physical channels.

7.10 *Discontinuous Transmission* (DTX) is when a transmitter is switched to mute if there is no user information for transmission on the physical channel. DTX is already being used in GSM systems where the transmitter power is always regulated downwards during voice pauses.

7.11 On the FDD UL the DTX function is not found within a frame. Instead, either the entire frame is used for the transmission or all time slots remain unused (during voice pauses).

7.12 On the FDD DL zero values are set in the otherwise two-value data stream during voice pauses. After the bit interleaving these DTX indicators are found distributed unevenly over the frames. Depending on the combination of transport channels, a second insertion of DTX indicators may be necessary in case whole voice data blocks are empty.

7.13 The physical layer provides different channels with varying gross data rates using different spreading factors. In addition, an adaptation to variable data rates is possible on the transport channels through rate adaptation using puncturing or repeated transmission.

Answers to questions for Chapter 8

8.1 A physical channel is a communications channel between two physical layers. It can be uniquely referenced through physical values, e.g., through a mid-frequency and a frequency bandwidth, through a point in time or period of time, through a code, and so forth.

8.2 Physical channel:

- In FDD: in FDD mode a physical channel is characterised by the spreading code, the scrambling code and, on the uplink, additionally through the phase position of the carrier (CDMA).
- In TDD: in TDD mode a physical channel is characterized by the spreading code, the scrambling code and by a time slot (CDMA/TDMA).

In both modes a physical channel can additionally be characterised by a mid-frequency if more than one frequency channel is used (FDMA components).

8.3 Dedicated physical channels are used for communication between the physical layer of a Node B and that of a UE. A dedicated channel is used exclusively for such an individual connection and is therefore allocated to it. Therefore, a Node B can conclude from the physical channel to the UE and vice versa.

8.4 Shared physical channels are used for communication between the physical layer of a Node B and several UEs. This applies, for example, to *broadcast* channels and to packet-data channels, the capacity of which is shared by several UEs. When shared channels are used, an identification of the connection to which the sent data belongs is required in the data stream.

8.5 The key difference lies in the necessity for identification in the case of jointly used physical channels.

8.6 $15 \times 40\,\text{bit} = 600\,\text{bit}$.

8.7 $15 \times 80\,\text{bit} = 1200\,\text{bit}$.

8.8 Because a UL burst with BPSK modulation (2-value) is being transmitted but the DL burst uses QPSK modulation (4-value).

8.9 Variable data rates are achievable either through a change to the spreading factor or through multicode transmission. A maximum of 6 DPDCH with the same spreading factor can be transmitted simultaneously.

8.10 Variable data rates are achievable either through a change to the spreading factor or through multicode transmission. Certain conditions relating to the use of OVSF codes have to be taken into account.

8.11 Minimum spreading factor: 4.
6 DPDCH simultaneously for each 960 kbit/s gross = 5740 kbit/s gross
3 DPDCH are transmitted in the in-phase branch and 3 DPDCH and the DPCCH in the quadrature branch of the modulation.

8.12 Minimum spreading factor: 4.
A maximum of 3 DPDCH with spreading factor 4 can be used, because other codes from the remaining branch of the code tree are already being used by other physical channels.
$3 \times 1872\,\text{kbit/s} = 5616\,\text{kbit/s}$.

8.13 Minimum spreading factor: 32.
Maximum number of bursts: 30.
$30 \times 80\,\text{bit} = 2400\,\text{bit}$.

8.14 The aximum number of users is generally limited by the number of available codes. For the FDD downlink the number of simultaneous transmissions is restricted to 512, in TDD to a maximum of 16 which corresponds to the maximum spreading factor, respectively. In practice, the maximum number of available physical channes in the FDD downlink is much lower since common and dedicated signalling has to be transmitted in parallel to the user data. In addition to that a certain amount of the spreading codes can be allocated for shared use packet transmission. In the FDD uplink, the maximum number users is limited by the respective interference power affecting the base station's receivers.

8.15 Minimum spreading factor: 1.
$14 \times 441{,}6\,\text{bit/s} = 6182\,\text{bit/s}$ (gross).

8.16 Minimum spreading factor: 1.
$14 \times 441{,}6\,\text{bit/s} = 6182\,\text{bit/s}$ (gross).

Answers to questions for Chapter 9

9.1 In CDMA systems many participants can use the same frequency channel at the same time due to spectral spreading. All users of the same frequency channel produce the effect of co-channel interferers on the other users. The frequency reuse intervals in cellular CDMA networks are typically very small due to the fact that communication is possible even in conditions with low signal interference.

9.2 Spreading increases the signal-interference ratio in receivers.

9.3 FDD coverage range:

EIRP	21 dBm
Noise power	-103 dBm
Interference power	0
Total effective interference and noise	-103 dBm
Required C/I	-20.9 dB
Receiver sensitivity	-123.9 dBm
Gains and losses at BS	11 dB
Maximum path loss	155.9 dB
Fading margin	11.3 dB
Handover gain	3 dB
Link budget	147.6 dB
Range	2.28 km

9.4 Cell size with normal load:

EIRP	21 dBm
Noise power	-103 dBm
Interference power	-98
Total effective interference and noise	-96.8 dBm
Required C/I	-20,9 dB
Receiver sensitivity	-117.7 dBm
Gains and losses at BS	11 dB
Maximum path loss	149.7 dB
Fading margin	11.3 dB
Handover gain	3 dB
Link budget	141.4 dB
Range	1.52 km

9.5 Cell size with higher interference:

EIRP	21 dBm
Noise power	-103 dBm
Interference power	-95
Total effective interference and noise	-93.8 dBm
Required C/I	-20.9 dB
Receiver sensitivity	-114.7 dBm
Gains and losses at BS	11 dB
Maximum path loss	146.7 dB
Fading margin	11.3 dB
Handover gain	3 dB
Link budget	138.4 dB
Range	1.25 km

9.6 Size of GSM/UTRA cells:

$$R/\mathrm{km} = 10^{\frac{147.6-135}{35.2}} = 2.28$$

9.7 The coverage range of 2.28 km is already the maximum achievable range, i.e., it does not allow for any interference. UMTS cells must therefore be reduced in size.

9.8 UL: capacity reduces as interference increases, maximum transmitter power is constant. DL: capacity reduces as interference increases, maximum transmitter power per user decreases as number of users increases.

9.9 In soft handovers a UE has a connection to two or more different Node Bs that are typically separated from one another physically. In softer handovers the UE has a connection to two or more sectors or cells that are supplied by the same Node B.

9.10 A softer handover is an intra-controller, intra-Node B handover. A soft handover is an inter-controller handover when the same controller controls all participating Node Bs. Otherwise it is an intra-controller handover.

9.11 A soft handover essentially produces two effects. First, the sensitivity of the receiver to fast and slow fading decreases due to spatial diversity. As a result, there is a decrease in the signal-to-interference ratio needed for sufficiently effective data transmission. Second, the soft handover prevents a UE from moving from one cell to another while it is still being supplied and controlled by the original Node B, thereby causing strong interference in the other cell. Soft handovers ensure that the transmitter power of the UE is always controlled by the Node B to which the lowest propagation loss exists. This prevents a near-far effect.

9.12 On the UL the signal sent by the UE is decoded into two or more Node Bs and the individual data streams are merged in the RNC. On the DL the data stream is duplicated in the RNC and always transmitted with specific scrambling codes over two or more Node Bs. The UE decodes all copies of the data stream and merges them.

9.13 In UMTS FDD the number may not be greater than three. The RNC decides which base stations are involved.

9.14 Handover:

(a) Softer handover

(b) Soft handover

(c) Hard handover

(d) Hard handover

(e) Hard handover

(f) Hard handover

9.15 UTRA-FDD stations transmit without interruption. However, such interruptions are required when seeking possible target frequencies.

Answers to questions for Chapter 10

10.1 Participants can access personalised services independently of the network (for example, abroad) and terminal used. This enables users to use the initial knowledge they have acquired on how to use services.

10.2 User data, program, execution environment, communication control. They develop applications usable for different mobile stations through the open interfaces provided by different developers. This simplifies the development of frequently used applications (***killer applications***). The same history of success incidentally applies to personal computer architecture. Many developers were able to develop applications on the basis of disclosed interfaces and therefore contribute to the continuing success of a system.

10.3 SAT applications are run on a processor on the SIM card, whereas MexE applications are executed within the terminal.

10.4 Some examples:

- Bearer services: Transmission channel with error correction or detection with e.g., 20 kbit/s, transmission channel with error correction with e.g., 144 kbit/s.
- Teleservices: Voice telephony, Internet access.
- Supplementary services: Call diversion when number busy, advice of charge.
- Value-added services: Call diversion when lines congested telematics), financial services.

10.5 Data transmission rate, jitter, latency, bit-error rate, quality of service.

10.6 Since there is no money to be made over the long term from purely transmitting bits, network operators try to extend their value-added chains and implement other applications on the basis of their networks. It is hoped that attractive services will be developed and customers will pay for information or entertainment and not purely for the transmission service.

10.7 Maximum12.2 kbit/s, minimum 4.75 kbit/s.

10.8 8000 values per second.

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