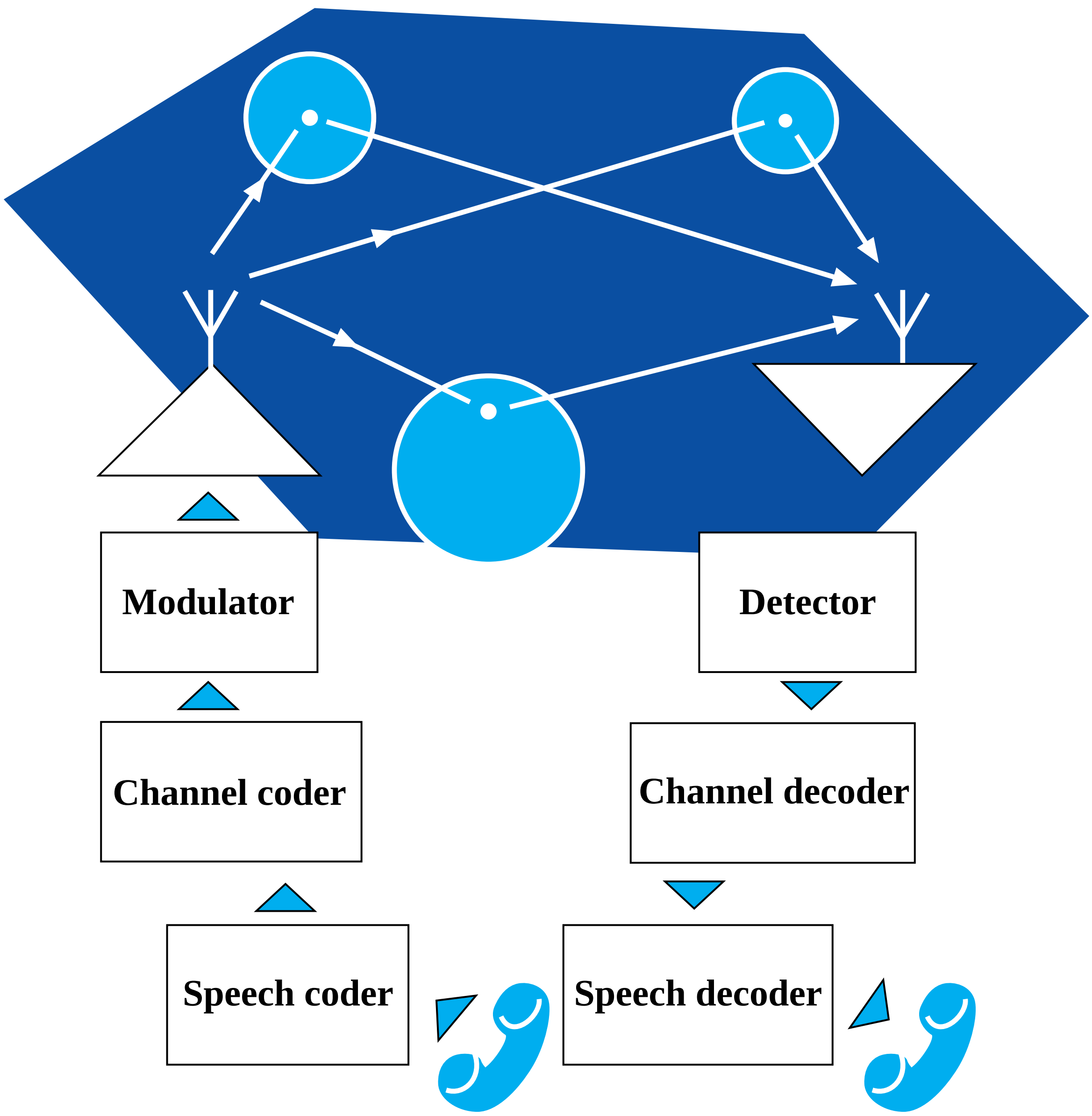
Radio School

DM3. Digital Mobile Telephone
Further developments



R C U R

Core Unit Radio Systems and Technology

ERICSSON

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DM3. Digital Mobile Telephone
Further developments

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1. Introduction

The **second generation of cellular systems** (first generation of digital cellular systems) was originally optimized for **speech services**, even if the objective for GSM was that eventually those of the ISDN services should be offered that were possible, taking into account the user rate limitations of GSM. One of the main reasons for the introduction of digital speech was improved frequency economy through reduced radio bandwidth per speech channel and/or reduced cluster size. The main example (see module DM1) is **GSM**. A less detailed discussion of the corresponding US and Japanese systems are given in module DM2. All three systems are based on **TDMA**. These systems are called **channel-limited**, as the traffic capacity is limited by the available number of radio channels within each cell.

Another system, belonging to the second generation of cellular systems optimized for speech is, is the US standard **IS-95**, proposed by Qualcom. See appendix 1. It is based on **DS-CDMA** (Direct-Sequence Code Division Multiple Access), which is an **interference-limited** system. In such a system, the maximum number of simultanious connections is determined by the fact that the transmission quality will be unsatisfactory, if the total cochannel interference within the system becomes too large.

An advantage of an interference-limited system is improved frequency economy due to efficient **averaging of cochannel interference**. The required cluster size is in the limit (extremely large number of coexisting connections) determined by the average interference level instead of the wost-case interference situation. Another advantage is fast dynamic allocation **of channel capacity** (dynamic demand assignment or ”bandwidth-on-demand”). However, this is not very important as long as the dominating service is speech.

A system according to the U.S. standard IS-95 gives a certain improvement of the frequency economy in comparation with a normal, channel-limited GSM system. However, the basic GSM system can be modified to an interference-limited system, by taking full advantage of the advanced frequency-hopping structure, included as an option in the specification. Such an interference-limited system is called **FH-CDMA** (Frequency Hopping CDMA). Extensive simulations have been made by ERA of the performance of GSM when operated as FH-CDMA. See appendix 2. Even if the frequency economy seems to be considerably improved and at least equal to the IS-95 system, the capacity improvement by operating GSM as a FH-CDMA system has not been considered sufficient to motivate the large efforts, needed to establish a commercial system.

However, the advantages of interference-limited systems are much more important in connection with the third generation of cellular systems **(UMTS/IMT-2000)**, which must offer a wide range of data and multi-media services. DS-CDMA has therefore been studied in the European research program **CODIT** as a possible MA-alternative for UMTS. See appendix 3. A parallel research program has studied a TDMA alternative. See appendix 4. A follow-up research program has been FRAMES. The results and conclusions from FRAMES have been used as background for the ETSI evaluation of several MA options for UMTS. Of these remained at the end of 1997 two main candidates, on of which is CD-CDMA. ETSI decided in February 1998 that the main UMTS air interface specification should be based on Wideband DS-CDMA ("WCDMA"). UMTS is discussed in section 7.

Another system approach, optimized for high-traffic areas, **is cordless telephone**. See section 2. The first digital system was CT-2, which was developed in England and standarized by ETSI. Due to limited system capabilities, only a few systems have been installed. The main cordless telephone system in Europe is **DECT** (Digital European Cordless Telephone - new name Digital Enhanced Cordless Telecommunications).

As for GSM, the objective of DECT is that it shall handle most of the services available within ISDN. In Japan, **PHS** (Personal Handy Phone System), is used extensively in areas with high traffic density.

Cordless telephone systems could be developed to form one part of an integrated UMTS system, suitable for the smallest cell sizes. A more short-terms solution is combined GSM/DECT networks. The advantages with DECT in comparation with GSM is improved frequency economy and reduced traffic charges.

Due to the rapid expansion of the mobile telephone market, large improvements are necessary with respect to frequency economy and geographic availability. One important system feature is **hierachical cell structures**, incl. good hand-over capabilities between different hierachical layers. The highest layer is **satellite cells**, the lowest layer is **indoor pico cells**. Cell structures are discussed in section 4.

The satellite service is considerably more expensive than terrestrial cellular systems, but anyway a considerable market is forseen, mainly as gap-fillers to the terrestrial systems (using dual-mode terminals). The first wordwide land-mobile satellite network IRIDIUM started commercial service in late 1998, and before year 2000 additional systems will be established.

A parallel development, which will give more efficient cell configurations, is the introduction of advanced **antenna structures with phased arrays**. These will improve the frequency economy (by reduction of the average level of cochannel interference) and the link budget (higher antenna gain). A short overview is given in section 9.

For indoor use, an alternative to UMTS mainly for movable data terminals is **W-LAN** (Wireless Local Area Networks), operating at considerably higher frequencies than UMTS. A first generation system, HIPERLAN (1) has already been standardized. W-LAN might operate as separate systems or be integrated with UMTS to extend the coverage (double-mode terminals). ETSI in cooperation with ATM Forum is developing the **HIPERLAN 2** standard for W-LAN. The transport layer of Hiperlan 2 will probably be based on ATM. It is therefore in line with another development to extend the ATM networks to include also mobile terminals (W-ATM). The objective is that W-ATM shall also permit the usage of outdoor terminals in micro cells. Hiperlan 2 and W-ATM will offer data rates up to 20 - 25 Mb/s. See section 8.

The key driving force, which motivates the development of the third generation of MTS, i.e. UMTS, is the rapidly **expanding market for data and multi-media services**. An important package of different services is based on IP-technology. IP stands for Internet Protocols. It will be a large demand to access also these services through wireless terminals.

IP-technology and ATM are examples of the trend to replace circuit switched networks with **packet oriented networks**, motivated by the need to handle a wide range of different multi-media and data services. Packet transmission is overviewed in sections 5.4 and 5.5.

UMTS will handle a wide range of transmission modes with user data rates up to 2 Mb/s. For several reasons, **the highest data rates could only be used in the smallest cells** in the hierarchy. In connection with macrocells, the maximum data rate will probably be below 400 kb/s. Services which need more than 2 Mb/s will be handled by W-LAN and W-ATM.

A considerable part of these new services will not need 2 Mb/s. Therefore there is a large interest **to extend the capabilities of the second generation of MTS**, e.g. GSM, to handle data services up to around 100 kb/s or possibly 400 kb/s. Therefore, there are two lines of development:

a. Extension of the capabilities of the second generation of MTS to handle packet transmission with low to moderate data rates. An example of this development is the General Packet Radio Service (GPRS) using GSM, see appendix 1 of module DM1. Advantages with extended generation-two MTS services are that these will probably be considerably less expensive than UMTS and give full wide-area coverage:

b. Standardization and development of the third generation MTS. In Europe the detailed air interface specification for UMTS is ready early 1999. ITU-R (previously called CCIR) has studied the third generation for many years under the name of FPLMTS (Future Public Land Mobile Telecommunication Systems). The name has recently been changed to IMT-2000. In Japan, the air interface for part of the corresponding standard has already been decided. Wideband DS-CDMA shall be used. (The need to cope with up to 2 Mb/s user data rate and multi-media means that the system differs considerably from IS-95. ITU will probably decide on the basic air interface specification for IMT-2000 at the end of 1998.)

As UMTS networks will for a long time be established only in high-traffic areas, there will a need to combine GSM and UMTS based on double-mode terminals.

In parallell with the development and standardization of UMTS/IMT-2000 there are several other lines of development, which in different ways influence the structures and markets for the future mobile communication networks:

a. The wide-spread **replacement of circuit oriented networks by packet networks** in the fixed telecom networks. **ATM** with fast packet switches might become the universal transport network of traditional telecom systems. A parallel development is **IP-technology**, coming from the data communication sector. This technology uses a somewhat different network structure, based on ”routers” which are a less complicated type of packet switch. However, the transport function could also be based on ATM. Long-term it would be possible and very competetive to use IP-technology also for toll-quality speech. The key IP-technology functions recide in the higher network layers. For the lower layers and wireless access many different technologies are competing (Ethernet, W-LAN, ATM, W-ATM, UMTS, DECT).

   The cellular networks must be integrated with the new packet-oriented transport networks. Therefore one important consideration is to what extent ATM and IP-technology can be used in connection with fading mobile propagation channels without too much reduction of the strict transmission quality standards (QoS) of the fixed network. Another question is to what extent the future telecom market will be taken over by IP systems and operators. It is possible that cooperating **ISP** (Internet Service Providers) can offer similar types of services as the telecom operators of today, incl. international roaming and access by wireless terminals.

b. Local-area-networks with wireless terminals **(W-LAN and W-ATM)**. The maximum user data rate would be much higher than for UMTS. **Hiperlan 2** might become a world standard using frequencies in the 5 GHz band for primarely indoor applications. A possibility could be to combine the two systems, so that users could swich between high data rate W-LAN indoors and medium rate UMTS outdoors. W-LAN technology can also be extended to outdoor point-to-multipoint applications (Wireless Local Loop).

c. **Wideband satellite services, i.e Teledesic.** The main application will be a high-speed service to fixed terminals. A proposed secondary application is a moderate rate data service (i.e. full ISDN service) to portable terminals.

d. **Digital Broadcasting (DAB and DVB).** One possible mobile application is an asymmetric service, using GSM in the inward direction and high-speed DAB/DVB channels in the outward direction.

e. **Universal, integrated access networks** comprising different types of access methods and servicing both mobile and fixed terminals.

2. Cordless Telephone

2.1 System alternatives

Cell-based digital mobile telephone systems have been discussed in modules DM1 and DM2. Another type of mobile telephone is cordless telephone. The main characteristic of cordless systems is their local coverage. Each system covers only a small area through very small cells (micro and pico cells). There are originally no requirements for operation also with large cells (macro cells). The basic version cannot handle roamers between local areas, even if different local coverage areas can cooperate to make limited roaming possible.

These characteristics and limitations result in reduced costs both for terminals and for the fixed part of the radio network, and also much improved frequency economy through the very tight frequency reuse.

The development of cordless telephone has gone through several steps. To start with, several types of extremely simple analog systems were introduced as an extension of fixed domestic telephones (CT-1).

The first system, based on digital speech transmission, was **CT-2**. It is based on FDMA and operates in the 900 MHz band. The main application is to connect cordless terminals to **Tele Points** but the system can also be used as Radio PABX. Telepoints are small base stations that have a typical range of 30 to 50 m and are located on walls or lamp posts in business centres, railway stations, airports and the like. The terminals can access the public telephone network through these telepoints. A **Radio PABX** is an office exchange to which cordless terminals are connected.

The radio access for CT-2 is based on time duplex and 100 kHz radio channels. See Fig. 2.1, which summarizes the radio parameters. The same arrangement for antenna diversity is used as for DECT, see section 2.2.

Figure 2.2 describes the time division duplex (TDD) structure. In the figure are shown two types of frames: one is used mainly for time synchronization in conjunction with the initial setting-up of calls from terminals. The other type of frame is used during calls. 64 bits is transmitted in each direction during each 2 ms frame. This corresponds to a 32 kb/s traffic channel.

Digital Cordless Telephone CT-2

Frequency range: 864 - 868 MHz

FDMA: Time duplex

Channel spacing: 100 kHz

Speech coder: 32 kb/s ADPCM

Base station: Antenna diversity for both directions

No need for echo control

Figure 2.1

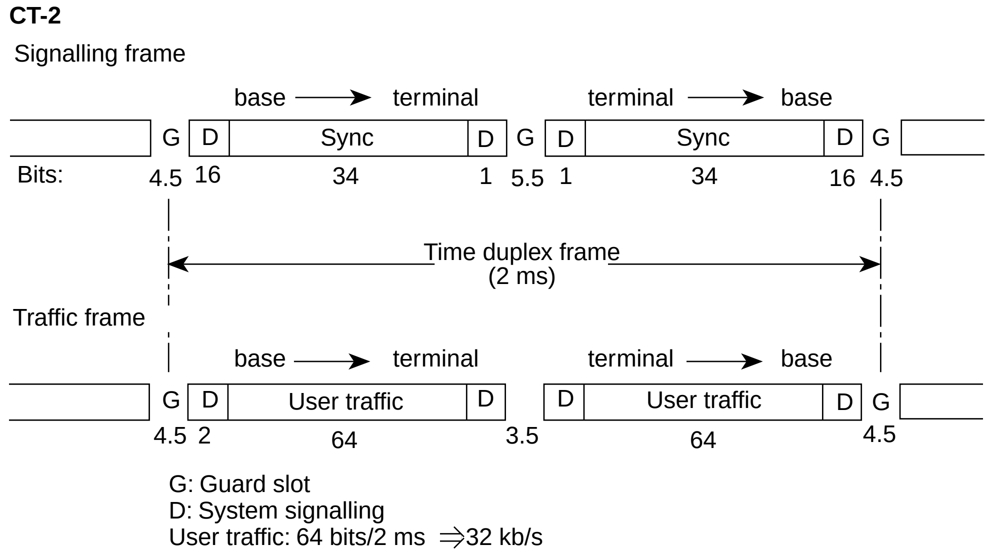


Figure 2.2

A more advanced system is **DECT** (Digital Enhanced Cordless Telecommunications), for which is used a combination of TDMA and TDD. The overall system concept was verified in 1988 through a 900 MHz test system, which was set up in Sweden. Based on the results from this test system, ETSI has developed the specification for a pan-European system operating in the 1800 MHz band. The specification was adopted as an official EC standard in 1992. The DECT system is mainly used for Radio-PABX applications. A recent application is point-to-multipoint systems. (The range can be extended to around 5 km without any problems due to time dispersion, as narrow-lobe antennas are used at both ends).

The key radio parameters are listed in figure 2.3. Figure 2.4, describes the TDMA/ TDD structure used for traffic. The TDMA frame can be structured so that several time slots can be allocated to users which need wide bandwidth "multislot".

DECT Digital Enhanced Cordless Telephone (DECT)

European (ETSI) standard 1991

Frequency band:	1,880 - 1,900 MHz
Channel spacing:	1.7 MHz
System data rate:	1.15 Mb/s
Modulation:	GMSK (BT=0.5)
TDMA-time duplex:	TDMA frame 10 ms 2x12 time slots
Transmitter power: mean peak	10 mW 250 mW
No FEC channel coding	
No channel equalization	
Speech coder:	32 kb/s ADPCM
Antenna diversity at base station for both inward and outward directions.	
Echo suppression required in some network configurations.	
ISDN compatible	

Figure 2.3

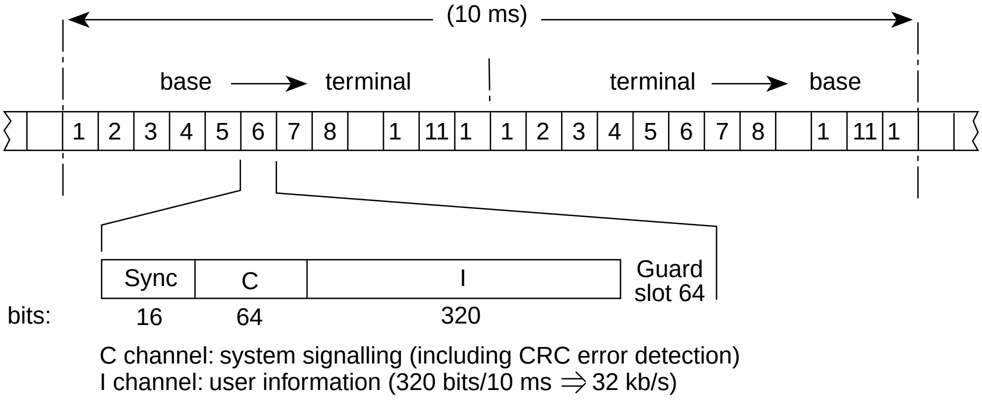


Figure 2.4

A parallell development to DECT is the Personal Handy Phone System (PHS) in Japan. PHS has some similarities to DECT, but more limited capabilities as the goal was to establish a low-cost system with extremely small portable phones and with speech as the dominating service.

PHS uses a TDMA frame with only 4 time slots, three for traffic and one for signalling. Also the system data rate is three times lower than for DECT, which makes it more tolerant to time dispersion, but only allows low rate data services. A more advanced (coherent) detector is used than in GSM, which gives somewhat better receiver sensitivity, but requires very high carrier stability. The small number of time slots in a frame makes the system unsuitable for dynamic channel allocation. A comperation of the basic system parameters for DECT and PHS is shown in figure 2.5

	PHS	DECT
Access technique	MC/TDMA/TDD	MC/TDMA/TDD
Carrier spacing	300 kHz	1728 kHz
LO stability	3 ppm	25 ppm
Modulation	π/4 DQPSK	GFSK
RF power (peak/average)	80/10 mW	250/10 mW
Sensitivity for 0.1% BER	-95 dBm	-89 dBm
Speech codec	32 kb/s ADPCM	32 kb/s ADPCM
Tolerance to time dispersion	500 ns	200 ns
Frame duration	5 ms	10 ms
Time slots/frame	4 +4	12 +12
Speech channels/carrier	3	12
Packet data	24 kb/s	552 kb/s
Seamless handover	no	yes

Figure 2.5

2.2 DECT

Speech coding, echo control

The ADPCM 32 kb/s (G.726) speech coding standard is used, as the change-over to micro and pico cells has resulted in such a large improvement in frequency economy that the resulting larger bandwidth per speech channel is acceptable. The advantages are improved speech quality, lower cost and lower power drain in comparation with the low-rate coders for generation-two cellular systems.

Compared to CT2, the larger transmission delay at DECT due to TDMA is a draw-back, when DECT is connected to the public telephone network. In this case echo control is generally needed to suppress echos.

Network structure

A typical DECT-network, connected to a PABX, consists of a number of base stations, which together form a pico cell structure with fairly large overlap regions between nearby cells. The placement of the base stations is not critical besides the requirement for complete coverage of the service area. See figure 2.6.

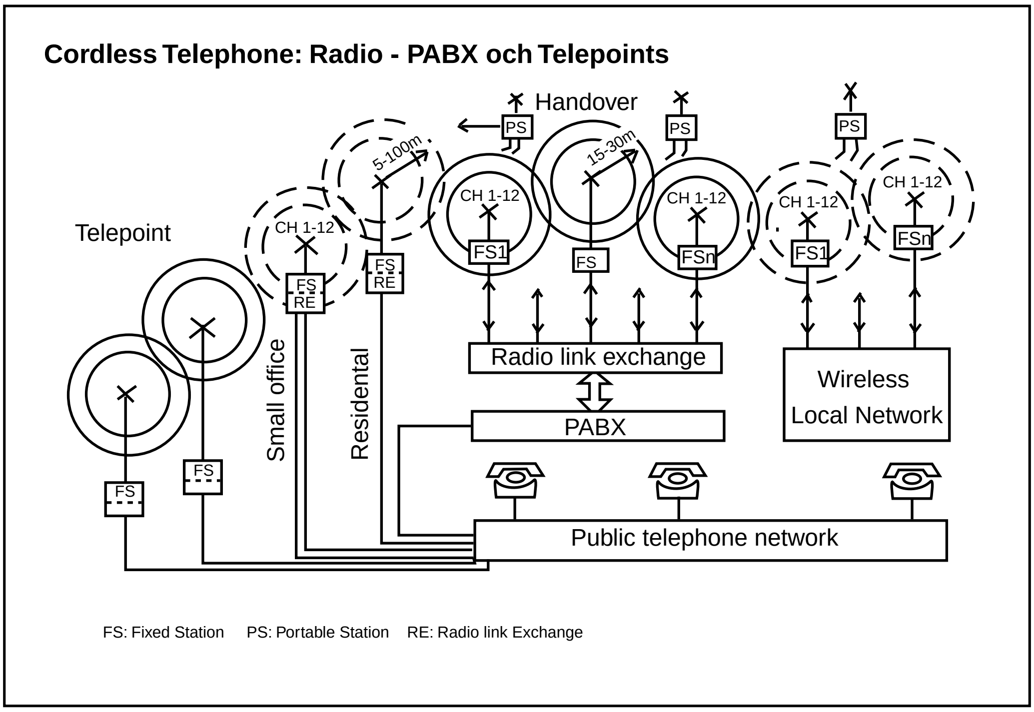


Figure 2.6

Diversity, dynamic channel allocation

The radio part of the DECT system differs from the GSM in two essential respects: the diversity arrangement against multipath fading, and the dynamic adaptive channel allocation.

The DECT system combines TDMA with TDD, i.e. the same radio channel is used for transmission in both directions. This means that the multi-path fading has precisely the same time and spatial structure in the up and down directions. The **antenna diversity** at the base station receivers is therefore effective also in the outward direction. The same diversity antenna is used for the two time slots in a frame, which constitute a two-way traffic channel. The prerequisite is that the time between the two time slots is short enough so that fading conditions are unchanged.

The antenna diversity in the DECT system provides roughly the same diversity gain as what is achieved through channel coding, interleaving and frequency hopping in the GSM system. Therefore, the speech channels are not protected by FEC. The main reason, why antenna diversity is better than FEC, is that channel coding in this case gives only a small improvement, even if supported by interleaving. The reason is that frequency hopping cannot be utilized. As most terminals are quasi-stationary, the error bursts during fading dips are therefore too long to be handled by interleaving only.

Another advantage of antenna diversity is that the time dispersion is reduced to some extent. The reason is that the time dispersion is most prominent when the dominent propagation path fades down during fading dips. Then weak secondary reflections with larger delays will be of larger relative importance, and the delay spread will be increased.

The combination of the antenna diversity and the small time dispersion in micro and pico cells allows modulation bandwidths of about 2 MHz without any need for equalization. DECT can tolerate delay spread up to 200 nS. However, the high system data rate and lack of equalization make DECT unsuitable for macro cell networks, with mobile terminals.

The other important feature of DECT is decentralized **dynamic channel** allocation. Each base station determines together with the involved terminal a suitable channel (time slot) for the call. The decision is based on quality data for the available time slots, which have been measured and stored by both the base station and the terminal A duplex channel is chosen that provides an acceptable C/I (and C/N) for both transmission directions. It might happen that a call in progress suffers strong cochannel interference. If that should happen, the call is switched to another more suitable time slot.

This dynamic channel allocation gives a **large improvement of the frequency efficiency**. The explanation is that with traditional fixed frequency planning it is necessary that the reuse distance is large enough to handle the worst propagation case. In contrast, when dynamic channel allocation is employed, it is possible to utilize the fact that the propagation geometries for C and I and also the shadow effects are considerably more favourable for the average case than for the worst case. The result is a considerable reduction of the average reuse distance. It might even happen that the same time slot can be used for connections in adjacent cells. See figure 2.7.

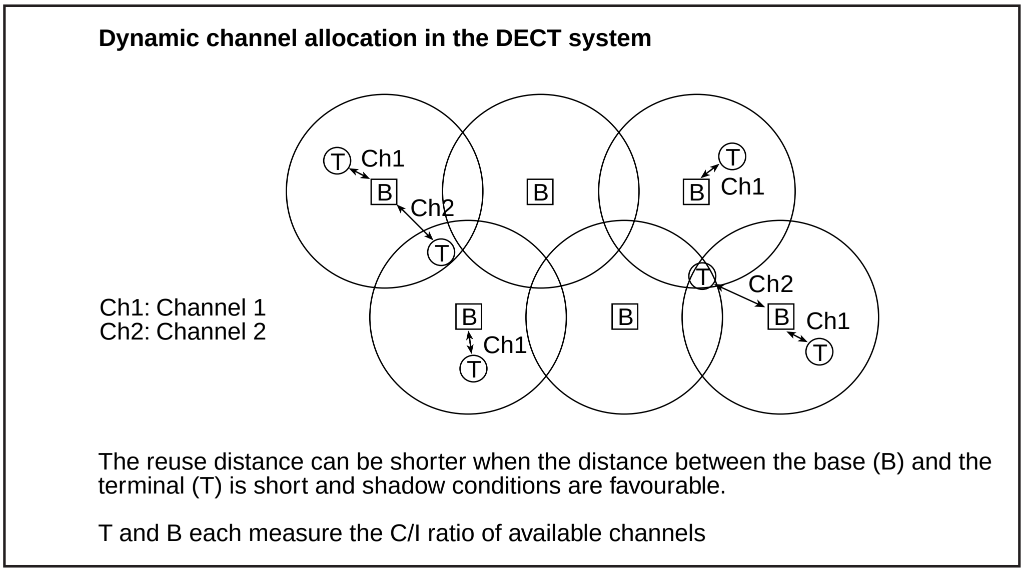


Figure 2.7

A further important advantage of dynamic adaptive channel allocation is the much **simplified installation planning**, since there is no need for traditional frequency planning with allocation of channels to different cells.

Applications

DECT and GSM networks can be combined by means of double-mode terminals and cooperating network structures. GSM gives wide-area coverage. However, when a subscriber is within range of one of the local DECT networks, a switch-over to DECT results in lower call charges and improved frequency economy. Also special subscription conditions often apply to radio-PABX networks such as DECT. A dual mode terminal will be only little larger than a GSM terminal. It is planned to introduce a similar arrangement for domestic applications, giving the user the advantage that there will be no radio charges when using the radio terminal in his home.

DECT is a bearer service for radio access. It can cooperate with many different fixed networks, through different gateways at the interface. It can service many types of terminals. The maximum user rate has been extended by multi-slot. An other possibility is to introduce more bandwith efficient modulation. One possibility is (2B+D) ISDN, another is packet transmission with data rates up to 552 kb/s. The multi-slot structure is very flexible, including the possibility to adopt to asymmetric traffic patterns (by allocating most of the time slots in a frame to one traffic direction).

DECT can also be used for wireless local area networks (W-LAN) with moderate user rates.

Another line of development is to use DECT technology for point-to-multipoint networks. In this application with fixed terminals directive antennas are used both at the base station and at the terminals. The delay spread is therefore reduced considerably and the useful range is therefore not degraded by time dispersion (5 km range possible). An example is DRA 1900.

3. Further development of the second generation MTS

Improved speech coders

The first versions of GSM, D-AMPS and PDC were optimized for speech, using full-rate speech coders. One line of further development is the introduction of **half-rate speech coders**. The main reason is that the existing operators in the 900 MHz band need increased traffic capacity. (In several countries, the present 900 MHz operators are not allowed to apply for frequency allocations in the 1800 - 1900 MHz band.) Half-rate coders have been introduced in the Japanese mobile telephone system PDC. In Europe, ETSI has developed a corresponding standard for GSM, but the interest from operators is small.

The speech quality of the half-rate coders is fairly similar to the previous full-rate speech coders during good conditions. However, they are less robust, i.e. the speech quality is considerably degraded by acoustic background noise (car and office operation, several speakers) and the coders also distort other signals than speech (such as back-ground music while waiting for connection in a telephone exchange).

Another line of development is to take advantage of the continuous developments in the areas of speech coding algorithms and VLSI to improve the speech quality. The overall objective is that the speech quality in all respects shall be at least equal to the 32 kb/s ADPCM speech coder, which for a long time has been a secondary standard (G.726) within the public telephone network. Besides the requirement on good speech quality under good conditions, there are requirements on low transmission delay and acceptable speech quality, even with relative high bit error rates and acoustic background noise.

A new full-rate speech coder with improved performance has recently been standarized for GSM (EFR: Enhanced Full Rate). The first practical application in an Ericsson system was introduced in Hongkong in September 97.

Studies and standardization work are also going on in ITU study groups. Several new speech coder standards have been issued. One of them (G.729) gives the same performance as G.726 but at 8 kb/s.

Data services

Another line of development is to make the digital mobile telephone systems of today more suitable for different **data services**. Higher capacity per traffic channel must be offered for intermediate data rate applications and there is also a need for better adaption to varying source data rates. The solution is a combination of **multi-slot** and modification of the basic TDMA protocol to permit **packet transmission**. Multi-slot means that one connection can be allocated several time slots in the TDMA frame.

In Europe, ETSI has developed extended versions of the GSM and DECT standards. Use of GSM for different types of data services is covered by Appendix 1 of module DM1. The General Packet Radio Service of GSM will provide date rates up to 115 k/bs. ”Evolved GSM” can offer up to 384 kb/s (on of the UMTS data rates) both for circuit switched and packet connections by replacing GMSK with linear modulation and, above all, utilizing high-level modulation (i.e 16QAM). Corresponding developments are going on in USA and Japan. See also appendix 4 about ATDMA.

In the US, new versions of D-AMPS include short message service (similar to what has been introduced for GSM), slow data services and fax. The new standard IS-136, which was issued late 94, breaks away from the coupling to analog AMPS, i.e. a complete mobile telephone system is specified, including a complete set of signalling channels (as at GSM) and system facilities to set up connections to and handle roamers. See Appendix 1 of module DM 2. D-AMPS extended by IS-136 is used for 1900 MHz US PCS networks.

In Japan, the revision B specification of PDS allows for fax and data with 9,6 kb/s. The following revision C (1996) introduced half-rate speech coders. The system advantage of the new spech coder is both improved radio capacity and lower transmission cost in the fixed network. (A 64 kb/s channel can be shared between several speech channels.) Further revisions cover fast modems and fax (revision D) and packet data transmission (revision E).

The possibilities to accomodate fast data services through further development of the present systems are limited. International study groups have proposed that the next generation (UMTS/IMT-2000) shall accomodate services with data rates up to 2 Mb/s. This is considerably higher than what is possible with the present TDMA cellular systems, even in the extreme case where one connection is allocated all the timeslots in a TDMA frame. DECT allows somewhat higher user rate, but not up to 2 Mb/s.

Furthermore, there are requirements on more flexible system arrangements, which will accomodate, with reasonable cost and frequency economy, many different services with different demands on bandwidth and quality (i.e. multi media). Therefore, a new generation of cellular systems is needed with wideband capabilities and improved frequency economy.

The different lines of development are summarized in Fig. 3.1. The ETSI specification for UMTS comprises both WCDMA and TDCDMA.

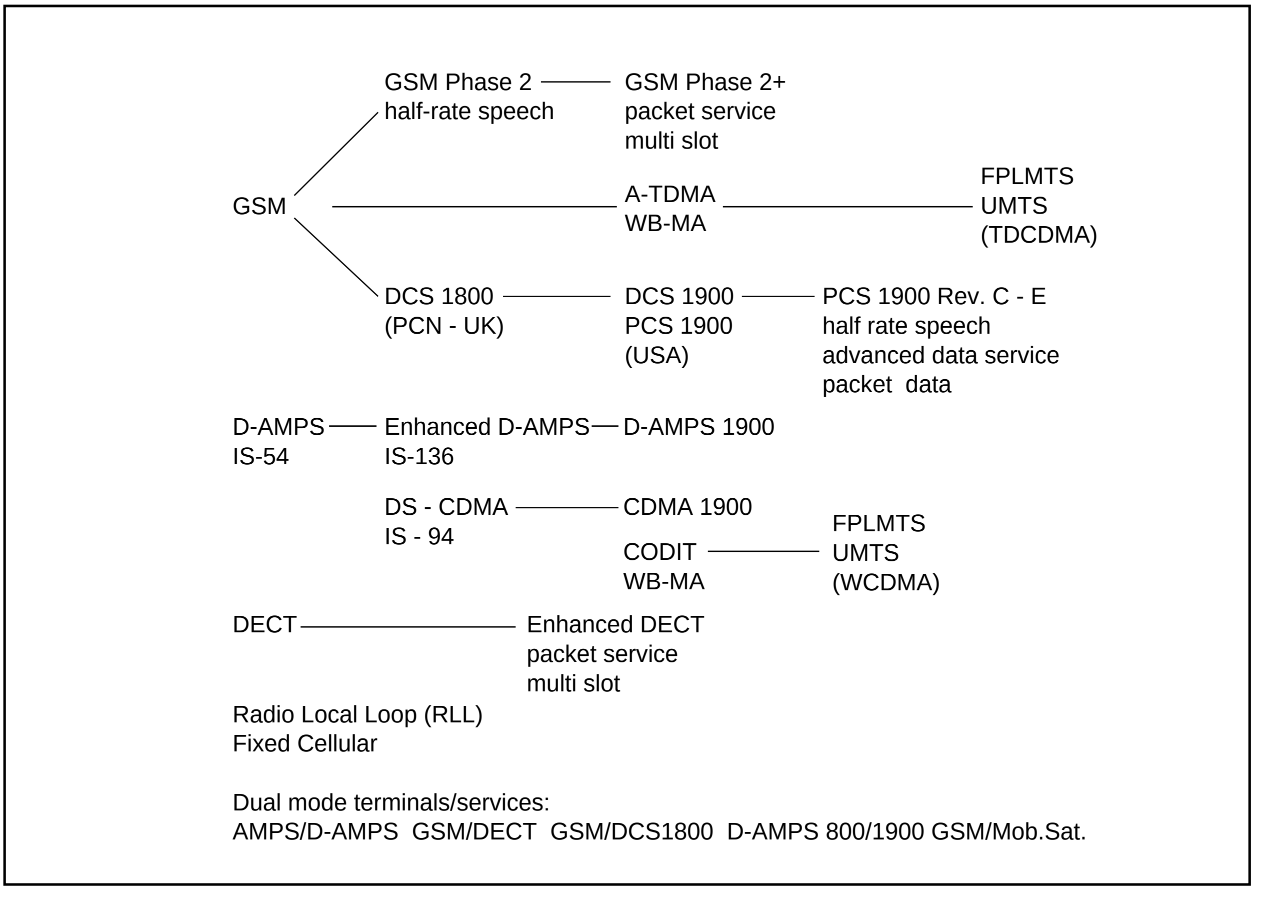


Figure 3.1

4. Cell structures

4.1 Additional frequency allocations to digital mobile telephone systems

The rapid increase of the number of mobile telephone subscribers has led to insufficient traffic capacity in several countries in the allocated frequency bands around 900 MHz. Most of the new subscribers join one of the digital mobile telephone networks. In the Nordic countries, the number of subscribers in the NMT system does not increase further. According to EC directives, the frequency bands for the present analog 900 MHz systems shall gradually be given over to GSM. In Sweden, the first step was taken in March 96, when each of the three GSM operators were allocated an additional 0.6 MHz, taken from the NMT band. See Fig. 4.1a.

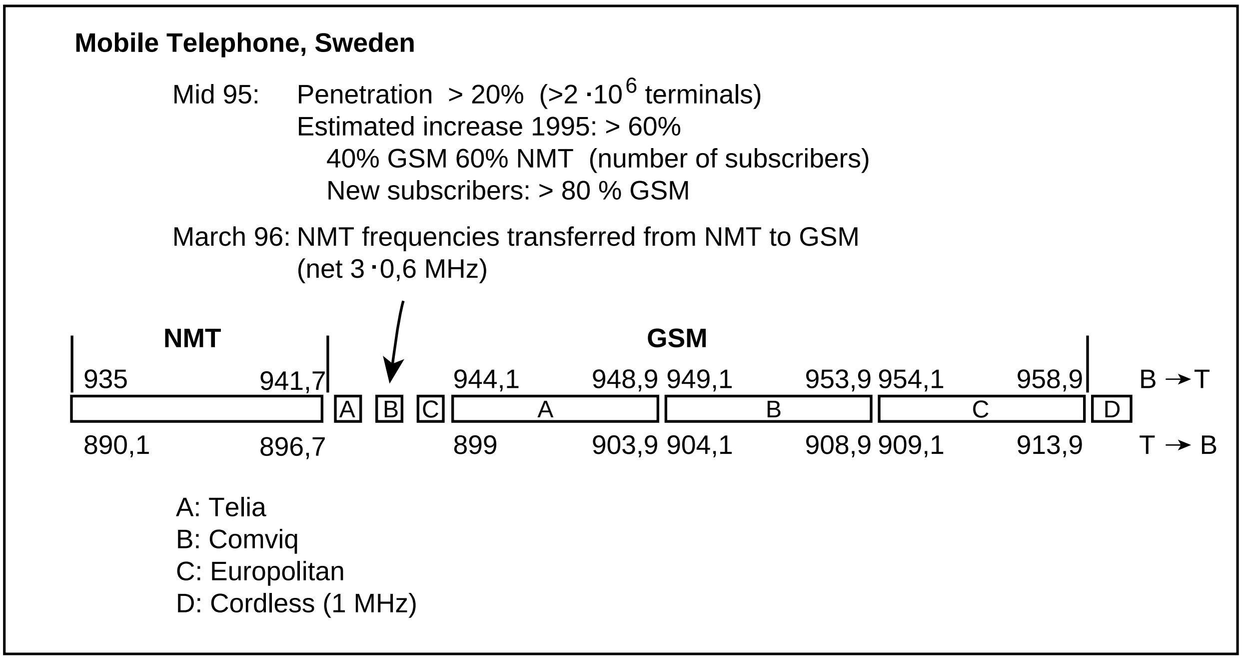


Figure 4.1a

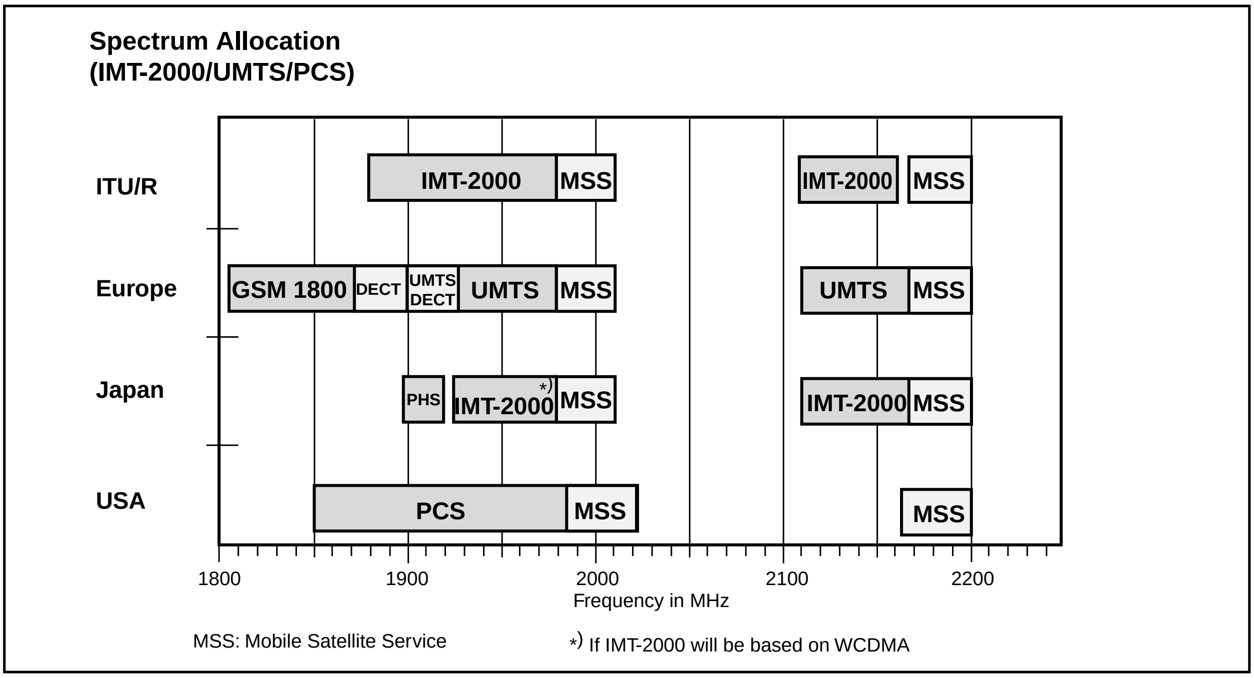


Figure 4.1b

On the other hand there seems to be good reasons to continue and improve the NMT 450 system due to the better propagation characteristics, which allow larger maximum cell sizes to be used. Radio Design AB in Kista was first in offering a multi-lobe base station antenna for NMT. The antenna gives 96 antenna lobes, which results in both improved link budget (increased range and/or reduced transmit power) and improved frequency economy, especially if frequency hopping is used (reduced cochannel interference). According to Radio Design the frequency economy is improved 8 times.

It is doubtful if it is economic viable to extend the 900 MHz GSM networks to give full coverage of the northern part of Sweden. The cell sizes will be even more restricted at 2000 MHz due to difficult propagation conditions. (The propagation loss is roughly 16 dB larger at 2000 MHz than at 500 MHz.) That means that the 2000 MHz UMTS networks will be limited to areas with high traffic density where small cells can be used. In scarsely populated areas, UMTS must therefore be complemented by 900 MHz GSM or by satellite networks.

The fast expansion of the mobile telephone market has already made it necessary to utilize part of the new 1800 - 2000 MHz band assigned to UMTS/IMT-2000. In Sweden combined 900 and 1800 MHz networks started operation late 97 (double-band terminals are needed).WARC 92 allocated frequency bands with a total width of 230 MHz for these systems. Europe and Japan still can find a common band, but this band is no longer available in the US (used by PCS 1900). Japan has already allocated the bands 1900 - 1980 MHz and 2110 - 2170 MHz for IMT-2000. See figure 4.1b.

4.2 Hierarchical cell structures

One of of the requirements on the future cellular systems is that they shall handle both densely and scarsely populated areas in an optimum way. This requirement also applies to the digital mobile telephone systems of today. Therefore hierarchical cell structures have already been introduced to some extent.

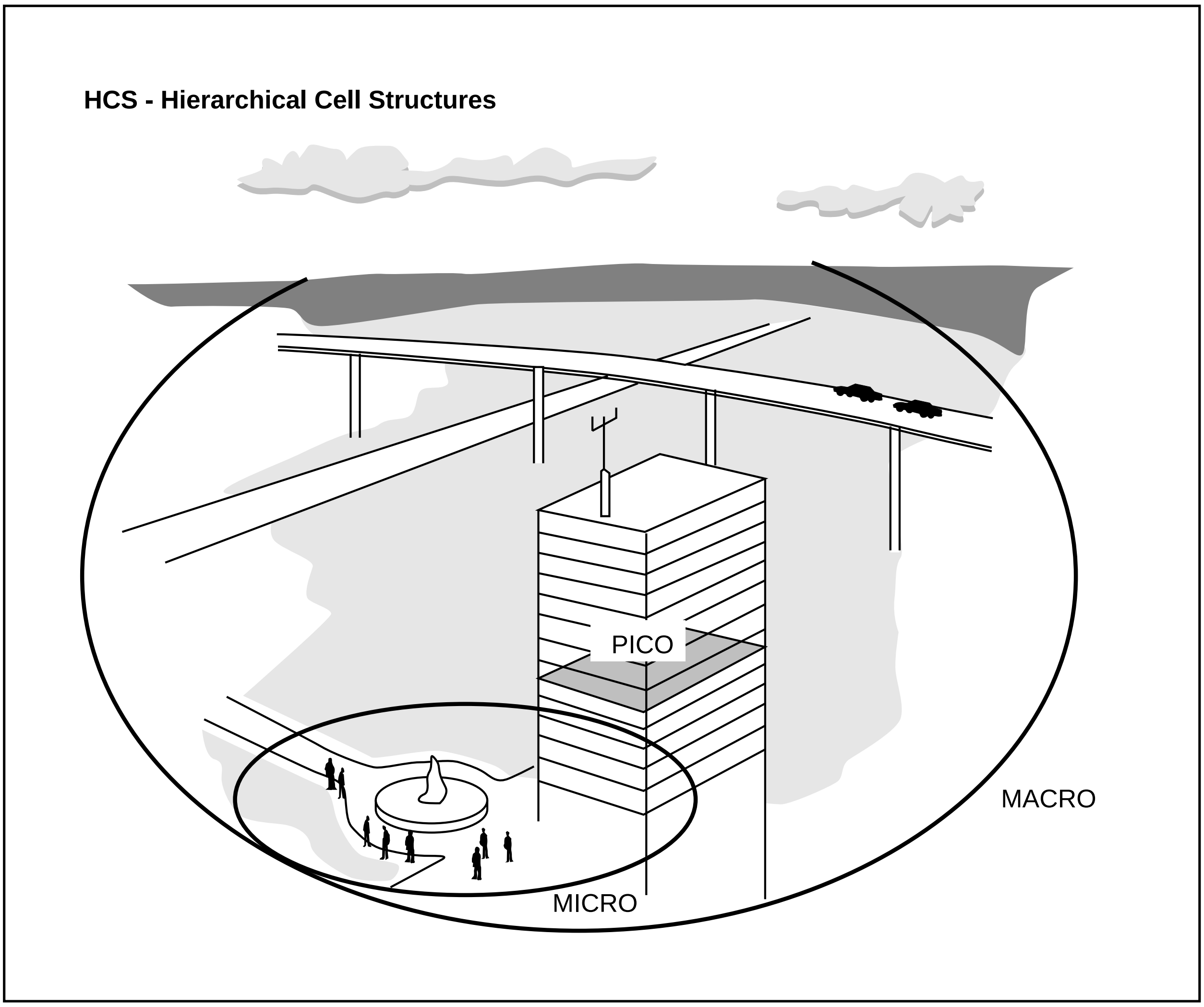


Figure 4.2

A hierarchical cell structure comprises cells of different sizes. See Fig. 4.2. Areas with considerably different traffic densities are covered by structures with different cell sizes, i.e. the larger the traffic density, the smaller cells are used. Without the hierarchical structure, the traffic capacity will be unsufficient in areas with high traffic, and the cost for the fixed network per terminal excessive in rural areas.

The highest traffic density can be found indoors, for examples in large offices and conference centra, assuming that most of the wire-connected telephones will be replaced by radio-connected ones. In this case, pico cells are used with dimensions of about 30 m. A pico cell might be fed by a base station placed in a corridor and with a coverage area consisting of the corridor and the adjoining rooms. Outdoors, traffic concentrations exist in shopping streets, sports and business centra. These are covered by micro cells with typical dimensions of 100 meters. A microcell can have a base station on a lamp post or house wall and cover a length of street.

Even in major cities there are areas with so low traffic density that it is not economic viable to use micro cells. Therefore, a complement is macro cells with approximately 1 km size. They are often called umbrella cells, as an important function is to fill in gaps between the high traffic areas, serviced by micro and picocells. As before, larger cells with sizes up to around 30 km (large macro cells) are used in rural areas.

The highest level in hierarchy are satellite cells, serviced by landmobile satellite systems.

The main part of the traffic will come from traffic concentrations at densely populated areas, outdoors and indoors. The major demand for radio spectrum will come from these traffic concentrations, which should be covered by micro and pico cells. This is a precondition for enough total traffic capacity in the future cellular systems.

The introduction of hierarchical cell structures gives certain complications on the system level. Even if the frequency economy is improved, there is probably a need for different allocation of radio channels to the different levels of the hierarchy. This could be a problem due to unsufficient spectrum allocation to an operator. One reason for the use of different frequencies for differernt layers is that the system control shall be able to force the terminals to use the lowest hierachical layer, that gives an adequate connection. However, a limitation is that micro cells cannot serve fast moving terminals, as this would result in such a high handover rate that the system signalling would be overloaded.

There are also additional requirements on signalling and control logic in order to handle handover between cells on different levels, incl. MAHO (Mobile Assisted Hand Over). The principle of MAHO as used at GSM is shown in figure 4.3.

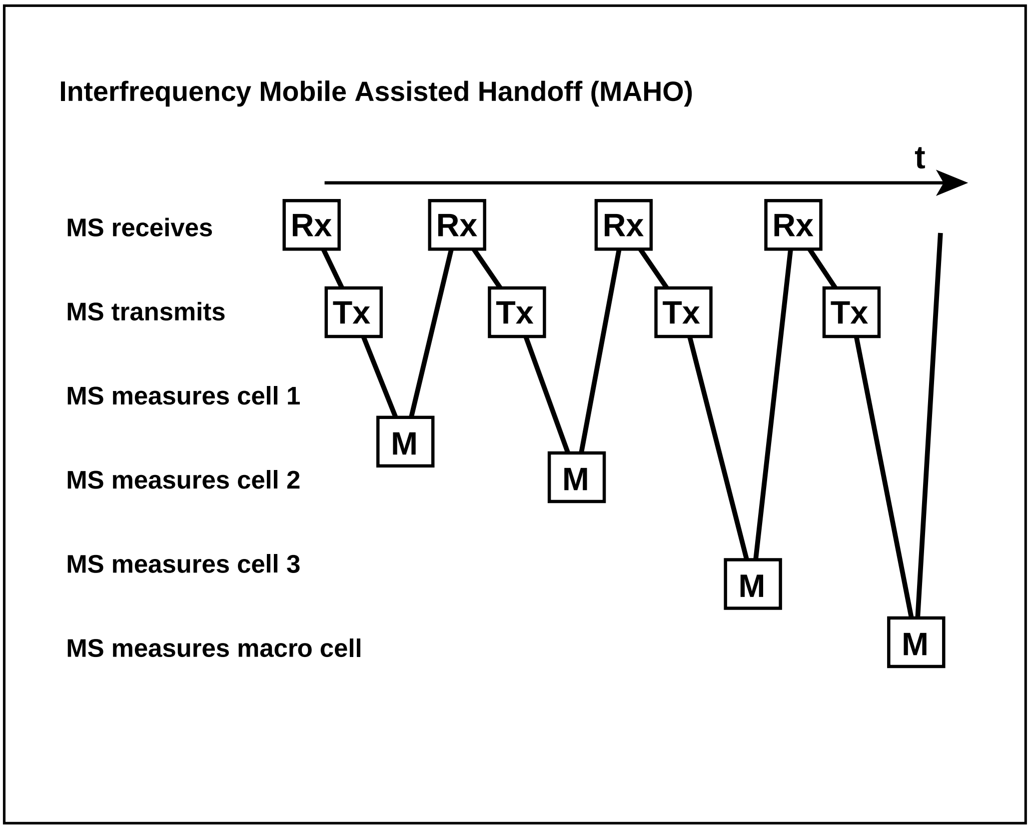


Figure 4.3

4.3 Propagation in micro and pico cells

Going over to micro and pico cells covering streets or indoor areas, the propagation is much more influenced by reflections from and attenuation in adjoining bodies such as floors, walls, vehicles, persons etc. than propagation in macro-cells (mainly above buildings). Therefore, the propagation models, suitable for micro and pico cells, deviate considerably from the traditional propagation models used for macro cells. This is discussed below.

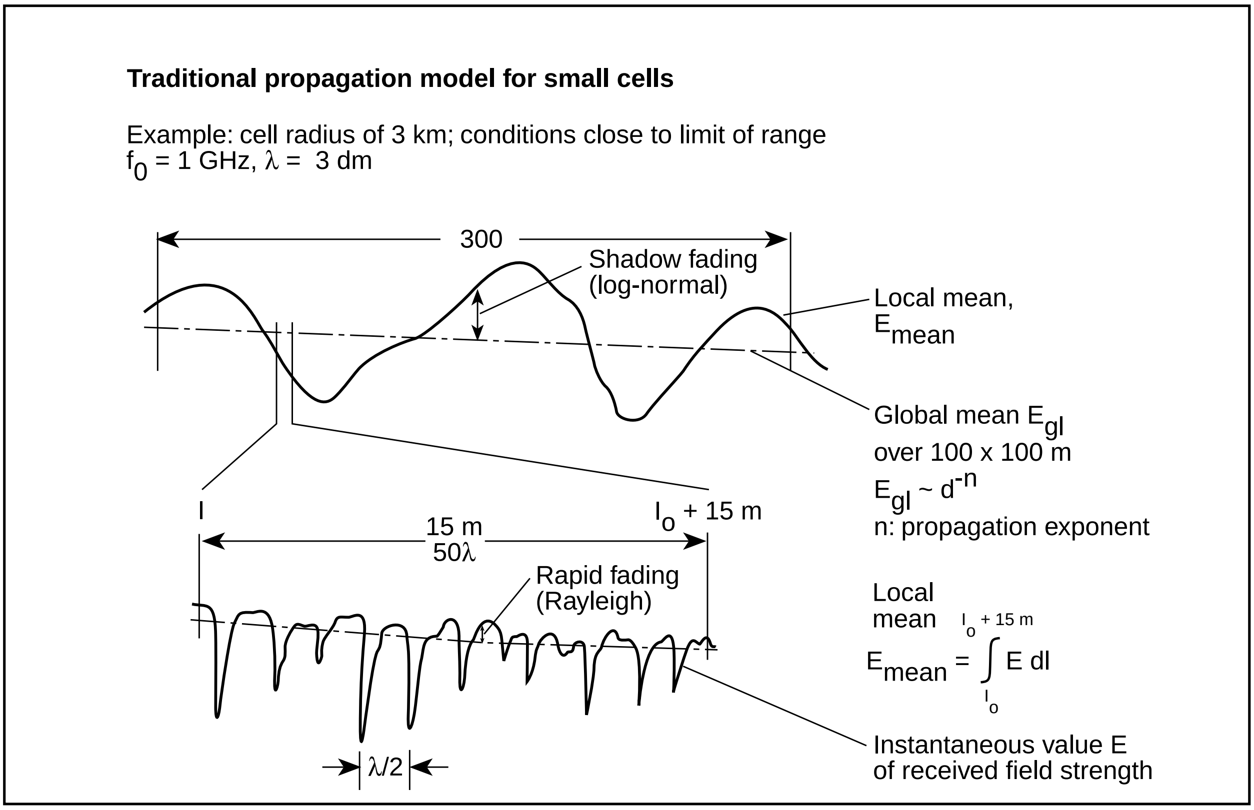


Figure 4.4

The traditional propagation model for large cells is summarized in Fig. 4.4 for a typical case: 1 GHz radio frequency and a cell radius of 3 km (10 000 wave lengths). Due to multipath propagation, fast fading (typically Rayleigh fading) occurs with fading dips approximately λ/2 (1,5 dm) apart. The local mean, E_{mean}, is obtained by integration of the field strength over many wave lengths, e.g. over 10 meters. This gives a stable local average over the multipath fading with small statistical uncertainty. The scale of the shadow fading is much larger than the multipath fading. Therefore, the shadowing varies very little over the integration interval used to calculate the local mean.

The deterministic distance dependance (global average) of the propagation attenuation can be distinguished from the shadow fading in a similar way. The global average is almost constant over a large enough area (e.g. 100 x 100 m) to enable the statistical characteristics of the shadow fading to be determined (the shadow fading can therefore be modeled by a stationary process).

These characteristics are the basis for the normal model for the design of cellular structures by means of adding fading margins to the global average of the propagation loss.

This model can be utilized only to a limited extent for the analysis of propagation in micro and pico cells. It is difficult to distinguish the distance-dependence of the global average from the shadow fading and also to discriminate between the shadow fading and the multipath fading. See Fig. 4.5.

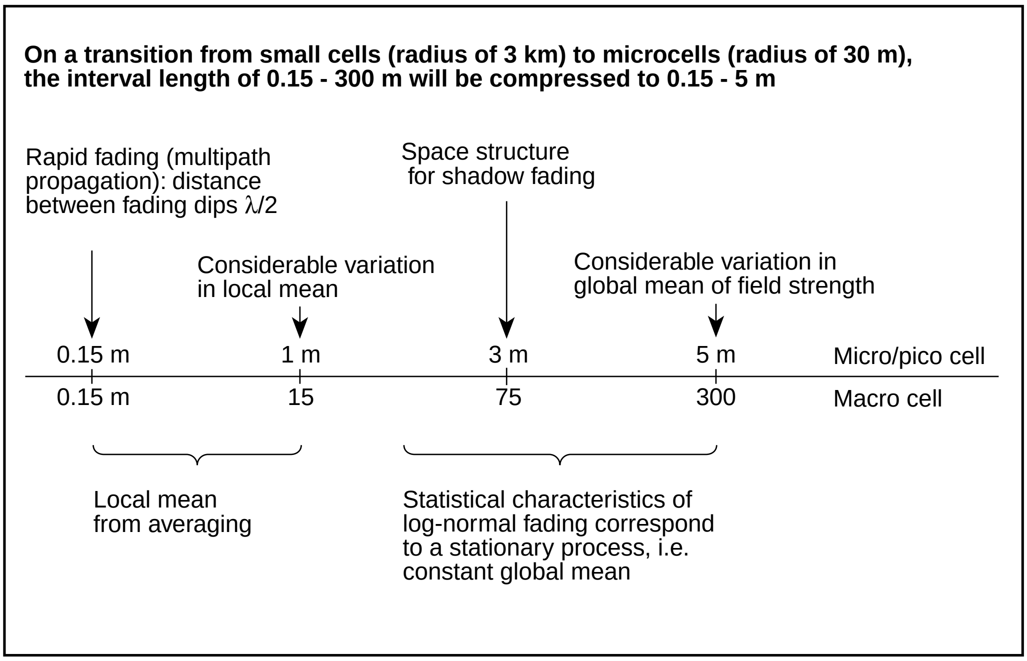


Figure 4.5

The picture is further complicated for indoor propagation in pico cells in confined areas as the total received signal consists of multiple reflexes from walls, floors and ceilings. Even if there is a direct path, the sum of these reflections is often of a similar level. The propagation resembles to some extent that in an over-sized wave guide.

Calculations and measurements show that, under these conditions, the distance between fading dips due to multipath can amount to several wave lengths, see figure 4.6. This can influence the system design considerably, e.g suitable diversity arrangements against multi path fading. Similar conditions apply to some extent to propagation along streets with high-rise buildings on both sides, if the transmitting and receiving antennas are sited close to the street level.

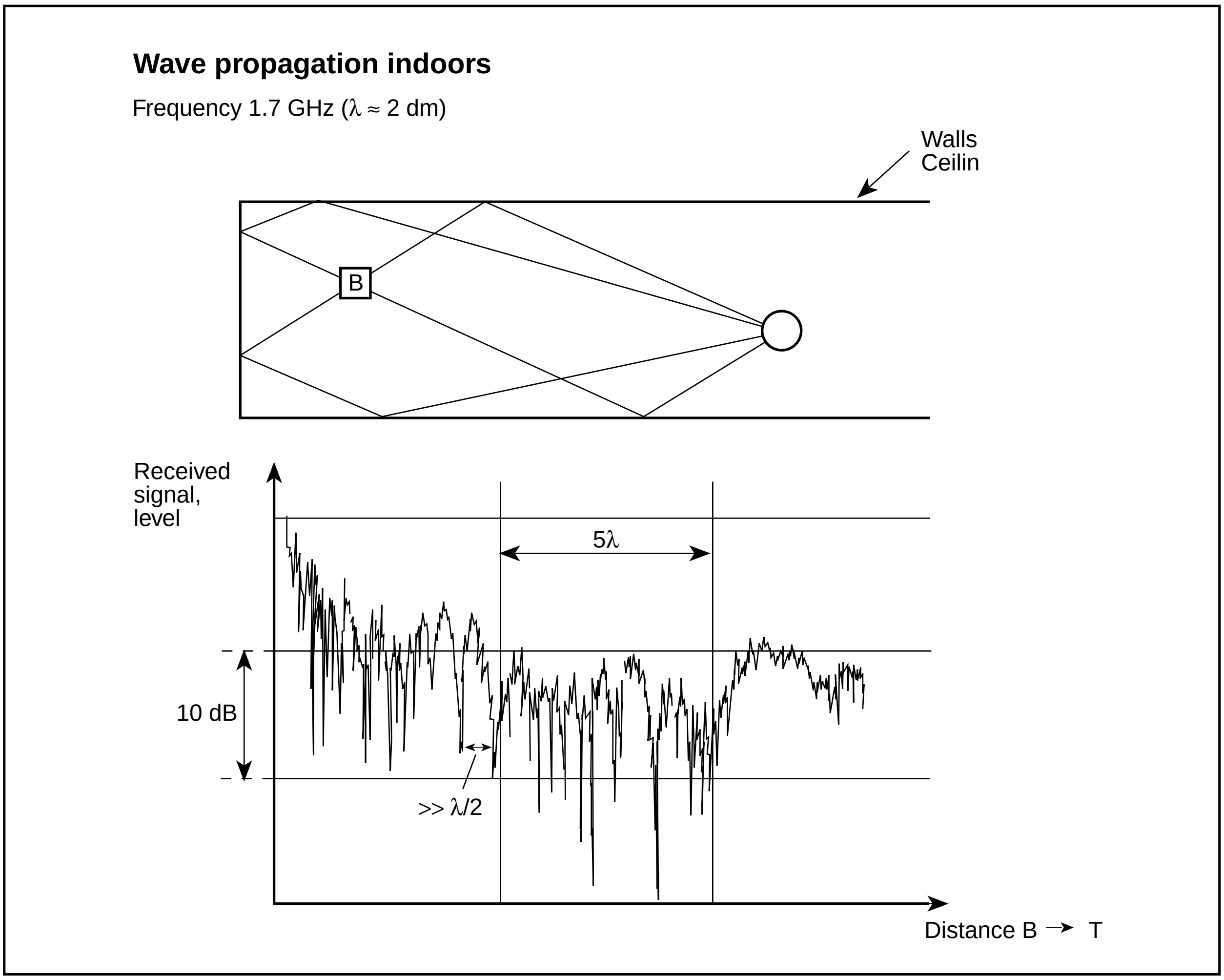


Figure 4.6

Another characteristic of propagation in micro and pico cells are the more pronounced shadow effects. When the terminal user turns the corner of a street, the propagation path can change from line of sight to obscured, causing a sudden increase in the propagation loss of 20 – 30 dB, see figure 4.7. The same condition can occur indoors. This could cause disturbing interruptions of the connection unless the handover is extremely fast or macro diversity with soft handover is introduced. A last resort could be to set up the connection via an umbrella cell during a short time interval in order to make the handover fully seamless.

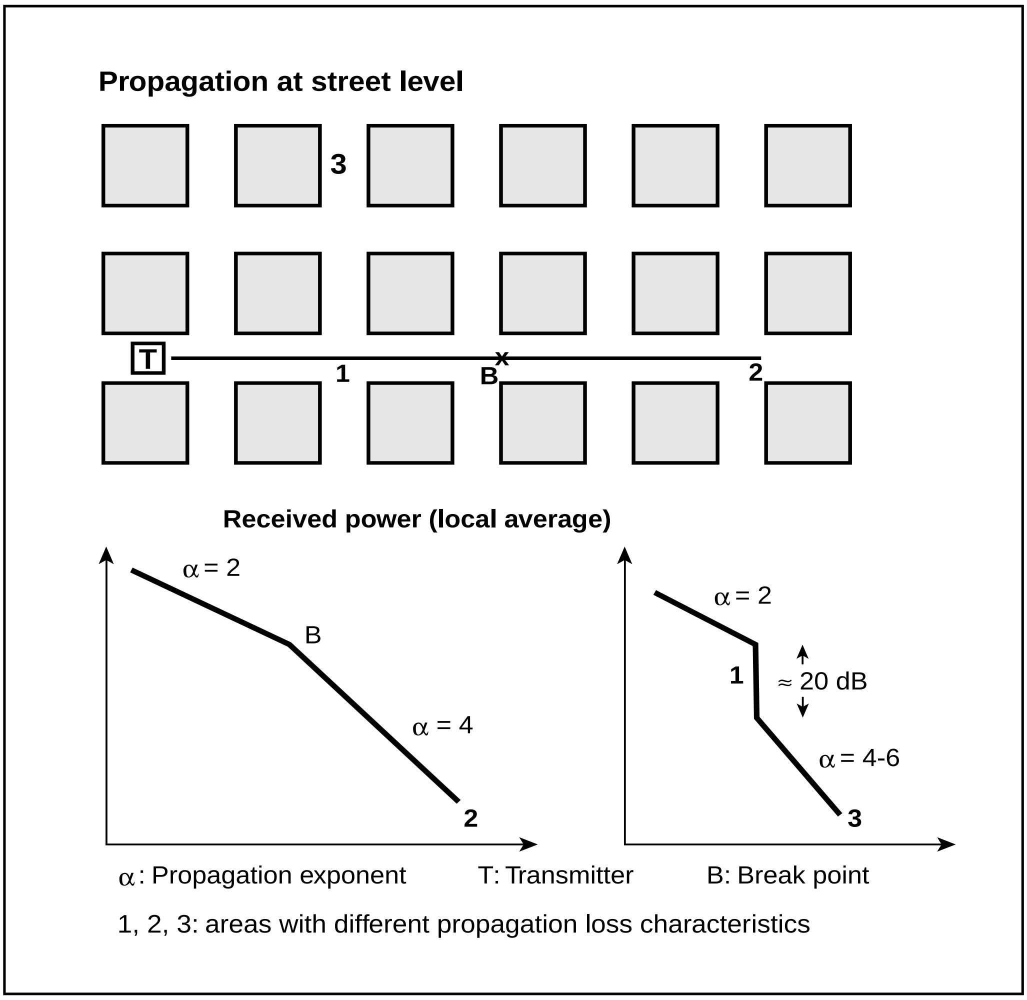


Figure 4.7

In figure 4.8 are summarized the characteristics for micro cell propagation.

Propagation in microcells

Problem 1.	*The global mean varies almost as rapidly as slow fading*
Problem 2.	*The local mean varies almost as rapidly as rapid fading*
Problem 3.	*The fading structure is strongly influenced by nearby objects. Multipath propagation creates a complicated fading structure.*
Implication:	*Analysis of micro and macrodiversity and of radio coverage must be based on new propagation models.*

Figure 4.8

4.4 Land mobile satellite communication

World-wide satellite communication has been used for many years for traffic to ships, using frequencies close to 1 GHz. The operator is Inmarsat, which is jointly owned by a large number of telecom administrations. Geostationary satellites are used. These are placed in an equatorial orbit 36000 km above the earth. The difficult link budget has been a limitation, which means that high performance terminals must be used with fairly high EIRP and G/T (G: gain of receiver antenna, T: noise temperature of receiver system). See figure 4.9.

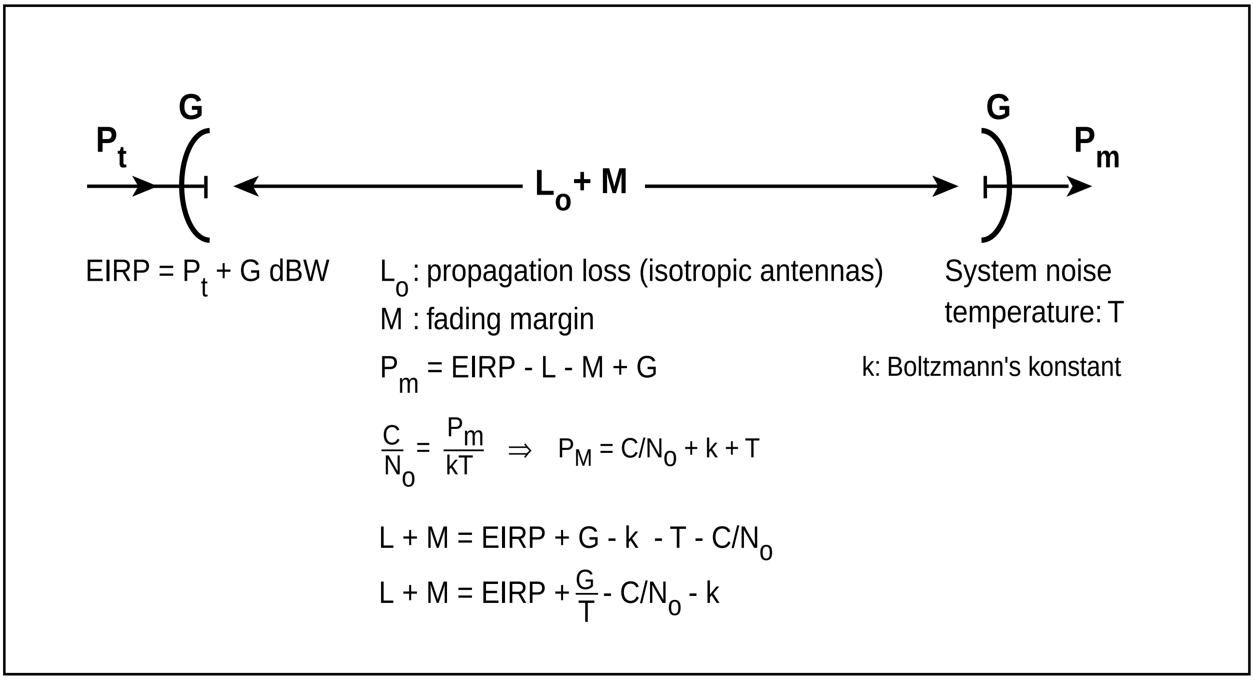


Figure 4.9

As the satellites must be placed over the equator, a further drawback is that the polar areas are not covered. If the service is extended to land mobile terminals, there would also be problems with strong shadowing in hilly terrain or in metropolitan areas, especially if the elevation angle to the satellite is small.

The same basic satellite system has recently been used also for slow data services, e.g. fax and short data messages. The combination of low data rate and low required Eb/No means that terminals with much reduced EIRP and G/T can be used, even if the output power from the satellite per traffic channel must be increased. The dimensions and cost of the terminals could therefore be reduced considerably. A large number of terminals have been installed on fairly small ships and even on vehicles. A portable version has about the same size as a large briefcase. Free sight is necessary, and the directive antenna must be pointed towards the satellite.

A new generation of geostationary Inmarsat satellites with high gain, multi-lobe antennas started operation in 1997. The satellite EIRP and G/T are considerably improved, which makes it possible to use fairly small mobile terminals for digital speech, even if their radio performance (EIRP and G/T) still must be considerably better than for terminals for terrestrial networks.

The fast expansion of terrestrial mobile telephone systems has caused a large interest to establish also land mobile satellite networks. Even if the cost for terminals and calls will be several times larger (estimate 3 times) than for the corresponding terrestrial services, it has been estimated that there would be a potential satellite market amounting to a few percent of the total mobile telephone traffic. This has been enough to motivate the large investments needed to establish such satellite systems, several of which are under development. The main applications are given in figure 4.10. The main application is probably as gap filler, i.e. the satellite network is used outside of the coverage area of the terrestrial network, that is normally used by the subscriber.

Land mobile satellite networks, which can be rapidly installed (if the satellites with earth support are already in place), could also complement the primitive public telephone networks in many developing countries.

Use of Mobile Satellite Communication

- Extend coverage to areas not covered by any terrestrial radio network
- Extend coverage to areas serviced by other, non-compatible terrestrial networks

Problems:

Difficult link budget

small shadow margin (indoor coverage very limited)

high cost per call

larger, more expensive terminals

Inferior frequency economy

large cells

Advantages:

Large (world wide) coverage by one flexible system

Figure 4.10

A satellite system, that shall be an extension of the terrestrial mobile network, must be designed so that the size of the dual-mode terminals (terrestrial + satellite service) is only moderately larger than a standard terminal for the terrestrial network. This requirement could be complied with by using advanced geostationary satellites with very large multi-lobe antennas. If a lower orbit is used, the size of the satellite antenna can be considerably reduced. (If the orbit height is changed, the link budget remains the same for the same area of the footprint of the satellite antenna on earth.)

That is one reason why the land mobile satellite systems being designed, are based on satellites in lower orbits than the geostationary, either low flying satellites in orbits with around 1000 km height (LEO: Low Earth Orbit) or in intermediate orbits (ICO: Intermediate Circular Orbit) about 10 000 km above the earth. See Fig. 4.11 and 4.12. The height interval between the LEO and ICO orbits is unsuitable due to a belt of intensive radiation (Van Allen belt), which would rapidly damage the satellite electronics.

Other advantages using the LEO and ICO orbits are considerably lower propagation delay than for the geostationary orbit and that good coverage is obtained also of regions close to the poles (less shadowing if the satellites are well above the horizon).

Mobile satellite communication - portable terminals

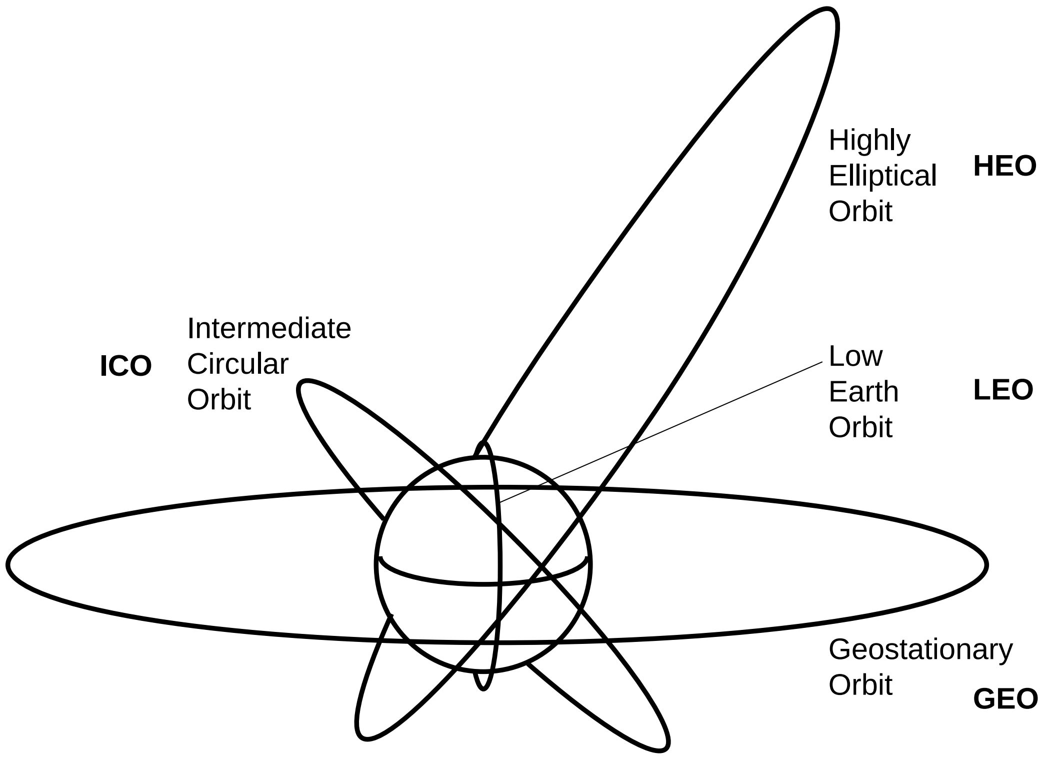


Figure 4.11

Land mobile satellite communication

	Iridium	Globalstar	Odyssey	ICO G.C.
Start of operation	98	99	99	99
Type of orbit	LEO	LEO	ICO	ICO
Orbit period (hours)	1.7	2	6	6
Satellite height (km)	800	1 400	10 000	10 000
Frequency band (terminals)	1.7 GHz			1.9/2.2 GHz
Number of satellites	66	48	12	10
One-way delay	100ms			200 ms
Cost 10^9 $ (excl. terminals)	3.4	1.8	1.7	2.4

Figure 4.12

Examples of systems using LEO and ICO respectively are the Iridium and the ICO Global Communication.

Iridium

The first proposal for a land mobile satellite service was the Iridium system from Motorola. The system comprises 66 satellites, which are placed in 6 polar orbits, each with 11 satellites. The satellites have advanced antenna arrangements, which generate a large number of lobes. The whole earth surface is covered with about 4000 cells. See Fig. 4.13. 4.8 kb/s speech coders are used. The required terminal EIRP is about 2 W (1W transmitter power and 3 dB antenna gain).

The LEO-satellites move fast relative to a fixed point on earth (orbit period around 100 minutes) and the moving cells are fairly small. A terminal is therefore within the coverage area of a cell only for a few minutes. Thus handover between cells is needed, even if the terminals are not moving.

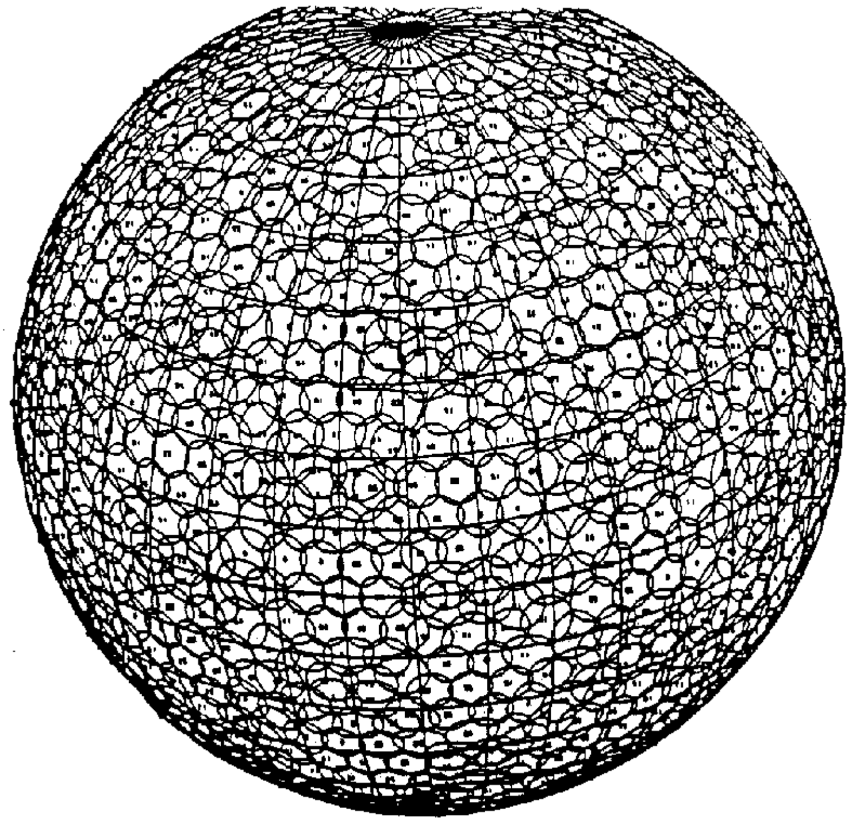


Figure 4.13

A further problem is that each of the earth stations, which connect the satellites with the fixed public telephone network, only can see a few of the 66 satellites at a certain time.

A large number of earth stations would therefore be needed, if it would have been necessary that all satellites at all times would be within sight of an earth station. However, only a small number of earth stations will be needed in the system, as the satellites operate both as an access network and a transport network. Independent of the position of the terminal to be connected, an earth station only needs to establish a connection with any suitable satellite above the horizon. From this satellite, the call can be relayed via several satellite-to-satellite links to that satellite, which is best situated to service the terminal. These connections (the transport network) consist of microwave links between nearby satellites. The arrangement evidently requires switches in the satellites. The fixed terrestrial infrastructure is based on GSM technology.

Commercial Iridium services will start late 1998. Therefore, the Iridium consortium has made roaming agreements with many terrestrial operators (i.e. using GSM and D-AMPS). Dual-mode terminals will be used.

ICO Global Communication (Inmarsat P)

The complications mentioned above with the LEO orbit are avoided, if the satellite network is based on the ICO orbit. The satellites are in a considerably higher orbit, and therefore move much slower (orbit period around 6 hours), also the cell sizes are somewhat larger. Handover between cells would not be necessary. Due to the higher satellite height, each earth station can service satellites over a fairly large part of the earth. Therefore, there is no need for a satellite transport network. INMARSAT (strictly speaking the affiliary ”ICO Global Communications”) will use 12 Satellite Access Nodes, each with 5 antennas to establish the contacts between the satellites and the fixed part of their network.

10 satellites will be placed in two orbits with 45 degrees inclination to the equator. No signal processing is performed in the satellites (”bent pipe”).

Networks compatible with UMTS

The land mobile satellite networks, mentioned above, are designed for speech as the dominating service. They belong to the same generation as GSM and D-AMPS. Only low-speed data services can be offered. A new generation of satellites is planned, which will give moderate rate data service to both mobile and fixed terminals. They could complement UMTS/IMS-2000. One example is Teledesic, which is discussed in section 7.4.

5. System technologies

5.1. Introduction

The dominating problem, when designing a cellular system is the mobile propagation channel, which must be shared between many simultanious connections (multiple access or multiplex). A detailed description of the mobile radio channel can be found in module G2. The key concepts are summarized in section 5.2.1.

The ongoing system development is based on gradual improvements of several system features and building blocks ("enabling system technologies"), which are stongly coupled from a system point of view. Better system performance is based to a large extent on the combined impact on the overall system of these features, including important mutual synergy effects. Section 5.1 gives an overview of a number of system technologies. Sections 5.3 to 5.5 go into more details. One of the most important technologies is interference-limited systems, see section 6.

As a complement to this chapter there are also a number of appendices, which give examples of concrete system configurations, which in different ways combine these system features.

Diversity (section 5.2.2 and 5.2.3)

One of the key objectives at the system optimization is to introduce more and more effective means to counteract the performance degradations due to the multipath and shadow fading, which are typical for the mobile propagation channel. At the system design, these are accounted for by applying fading margins in the link budget. Different diversity arrangements can be combined in order to reduce the fading margin by combining several types of diversity. The main goal is to permit a reduction of the required average value (global mean) of the protection ratio, thus allowing a smaller value of the normalized reuse distance D/d, i.e. the ratio between the distance D between adjacent cochannel sites and the cell radius d. D/d determines the necessary cluster size.

The main motivation for change over from analog to digital radio transmission was improved frequency economy. **GSM** is a good example. The improvement is primarely due to the reduction of the cluster size from 21 for NMT to 9 for GSM (a necessary condition for cluster size 9 also for quasi-stationary connections is the use of frequency hopping). The improvement is caused by the reduction of the required protection ratio (local average over multipath fading) from 18 dB for NMT to about 10 dB for GSM. This improvement is due to a combination of channel coding, interleaving and frequency hopping, which together give very effective micro diversity. Also, the modulation bandwidth at GSM is sufficient for a modest additional decrease of the multipath fading margin through path diversity. This is handled by the equalizer.

The above-mentioned signal processing against multipath fading (to get a diversity gain) also reduces $(C/N)_{min}$ about as much as the protection ratio $(C/I)_{min}$. **The noise-limited receiver sensitivity** is thus improved, which means that larger cell sizes can be used in scarsely populated areas, where there is no frequency shortage (assuming that the maximum permitted transmitter power is the same). The result is reduced cost for the fixed part of radio network. Alternatively, the terminal transmit power can be reduced, which means less battery drain.

Macro diversity (section 5.2.3)

In addition to the fast fading due to multipath propagation, there is also **shadow or log-normal fading**, which means that a shadow fading margin must be applied to move from the required local mean of C to the required global mean. (It is the minimum acceptable value of the global mean of the protection ratio $(C/I)_{min}$ that determines the cluster size.)

The minimum acceptable value of the global mean of $(C/I)_{min}$ and $(C/N)_{min}$ can be reduced by **macro or base station diversity**, in the overlap regions of nearby cells. The gain through macro diversity can be further increased by soft or seamless handover, even if it means somewhat higher system complexity.

MAHO

An important advantage with TDMA compared to FDMA and basic DS-CDMA is more efficient **handover procedures**. As only part of each TDMA frame is used for reception or transmission, the terminal has time available during idle time slots to measure the signal level or the signal-to-noise ratio on the radio channels, which are used by adjacent cells. This forms part of the information used by the base controller to prepare suitable actions, if the quality of the presently used channel suddenly drops so much that handover becomes necessary (Mobile Assisted Hand Over).

Interference-limited systems (section 6)

Frequency hopping breaks up the quasi-stationary fading pattern, which results in reduced length of the error bursts. An additional advantage of frequency hopping is the averaging of cochannel interference, as random frequency hopping spreads out the interference from each cochannel transmitter over many radio channels.

In the limit with averaging over a very large number of channels we obtain a perfect interference-limited system, for which the average value of the cochannel interference instead of the worst case determines the transmission quality and the necessary cluster size. A key idea is that simultanious connections use different random hopping patterns determined by different codes. Such a system is called a **FH-CDMA** system. CDMA means Code Division Multiple Access. Another CDMA alternative is **DS-CDMA** (DS:Direct Sequence). It is easier to obtain a fully interference-limited system using DS-CDMA.

The other advantage of an interference-limited system, beside efficient interference averaging, is fast dynamic allocation of the total transmission capacity between different users **(demand assignment or bandwidth on demand)**. This is a vital characteristic for future systems, which must handle large variations in user data rates, including variations during a connection (bursty sources). Also this feature can easier be implemented using DS-CDMA than FH-CDMA.

Requirements on third generation cellular systems

As already mentioned above, future personal communication systems must comply with important new requirements in order to accomodate a wide range of different services with low costs and good frequency economy.

The requirement that UMTS /IMT-2000 shall accomodate user rates up to 2 Mb/s, means a considerable system complication. 2 Mb/s is about 100 times more than what a traffic channel can handle in the original version of present digital mobile telephone systems. As a common, wide band radio channel shall service several simultanious traffic channels by means of a suitable multiple-access arrangement, the required system data rate becomes about 20 Mb/s. This leads to several consequences, which make the system design more difficult (see figure 5.1.)

Less traffic channels available within the system

Reason:

Services with large bandwidth demands

Possible actions:

Wide band services allowed only in micro and pico cells

Dynamic allocation of channel capacity

Base station antennas with high directivity

Increased transmitter power needed

Reason:

Higher system data rate requires increased input power to receiver

Possible actions:

Smaller cells - wide band services only in micro and pico cells

Base station antennas with high directivity

More complicated equalization

Reason:

Decreased duration of radio symbols

Possible actions:

Smaller cells - wide band services only in micro and pico cells

New type of multiple access (OFDM)

Figure 5.1

The need to accommodate traffic channels with much higher data rates than what is offered by the present systems, means that much larger modulation bandwidth must be used. This complicates the frequency planning as each operator is allocated a limited frequency band, often less than 20 MHz (2x10 MHz). This might be a problem, especially for DS-CDMA, which in addition needs a fairly large bandwidth expansion to get adequate isolation between connections. As the number of 2 MHz channels available would be quite low, wideband services could only be used in connection with **high frequency reuse** (small cells, small cluster size).

It is also expected that wide band transmission is requested only during short time intervals, as otherwise the call charges would become excessive. A consequence of this **is packet transmission** will be common in the future. Packet transmission can also give demand assignment of channel resources, even if it is less efficient than DS-CDMA. See section 5.4.3.

The other problem with high system data rates is that the **required receiver power is proportional to the data rate**, as $(Eb/No)_{min}$ is more or less the same for different services (depends on the required transmission quality). The main remedy, also for this problem, is to allow wide band services only in the smallest cells. This makes the required transmitter power reasonable even for wide band services.

The third problem is increased requirements on **equalization** to counteract the effects of time dispersion. The complexity of the equalizer depends primarely on the width of the equalizing window expressed in symbol intervals. Already for GSM, the need for equalization was a difficult limitation, which had a considerable impact on the system design - above all the maximum possible system data rate. As the width of the impulse response of the propagation channel is roughly proportional to the maximum useful range (cell radius), the equalizer becomes less complicated, if high system data rates are used only in the smallest cells. An example of such an arrangement is ATDMA in appendix 4.

Different maximum data rates will therefore be allowed in the different layers of the hierarchical cell structure. A suggestion is given in Fig. 5.2.

maximum data rate for traffic channel

Satellite cells	16 kb/s
Large macro cells	128 kb/s
Small macro cells	128 kb/s
Micro cells	500 kb/s or 2 Mb/s
Pico cells	2 Mb/s

Figure 5.2

For UMTS it is planned to use 2 Mb/s mainly indoors and restrict the maximum user rate to 384 kb/s for wide-area applications.

5.2. Propagation effects, diversity

5.2.1 The land-mobile propagation channel

The propagation in macro cells can be described by a combination of three factors: the distance dependance, which determines the global average of the received signal, the impact of shadowing, which is described by the slow lognormal fading, and the multi-path effects, which in the worst case are described by the fast Rayleigh fading. See figure 5.3. The distance dependence is given by the **propagation exponent** α, which for macro cells is about 4. The fading characteristics have already been mentioned in figure 4.4.

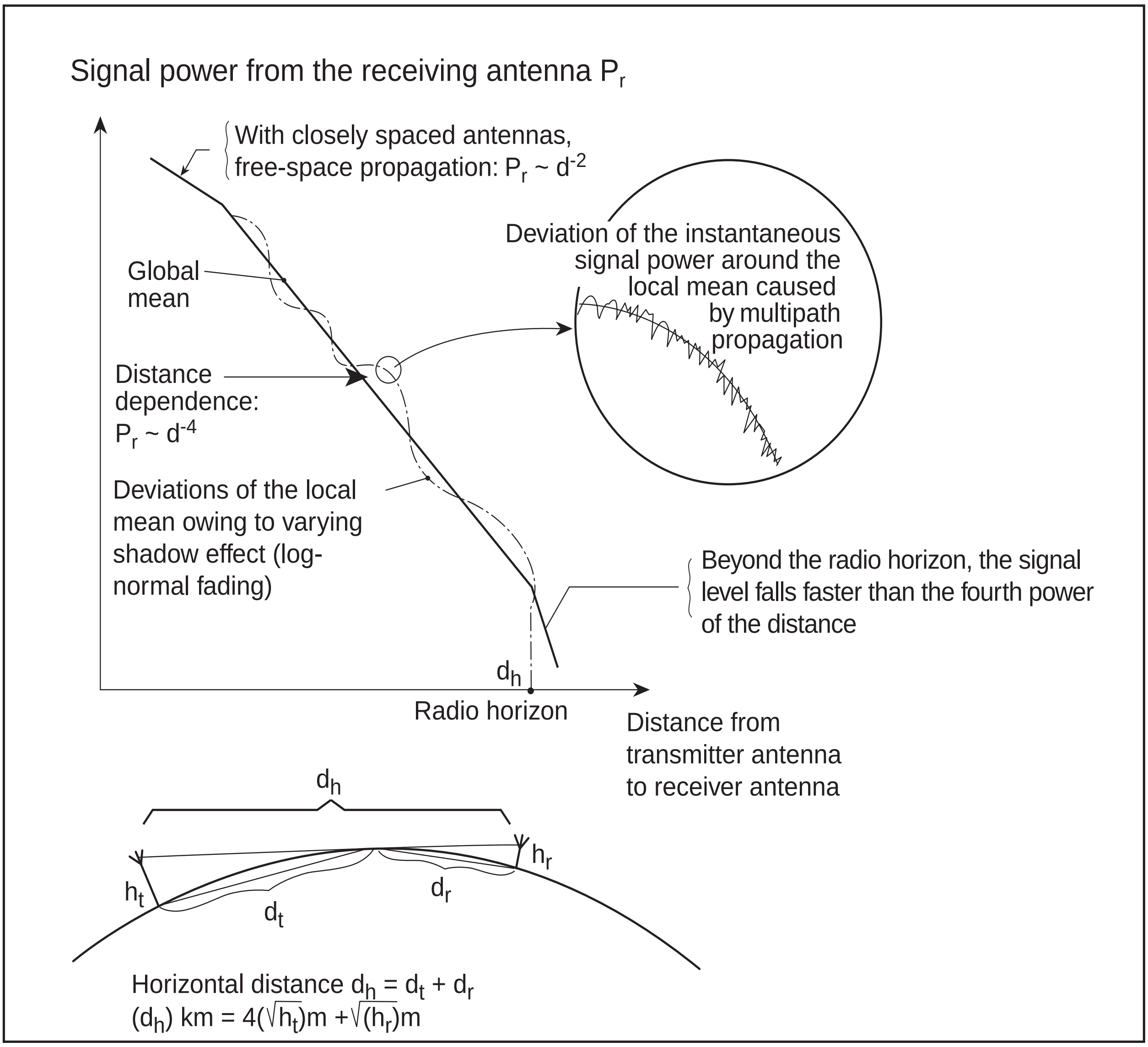


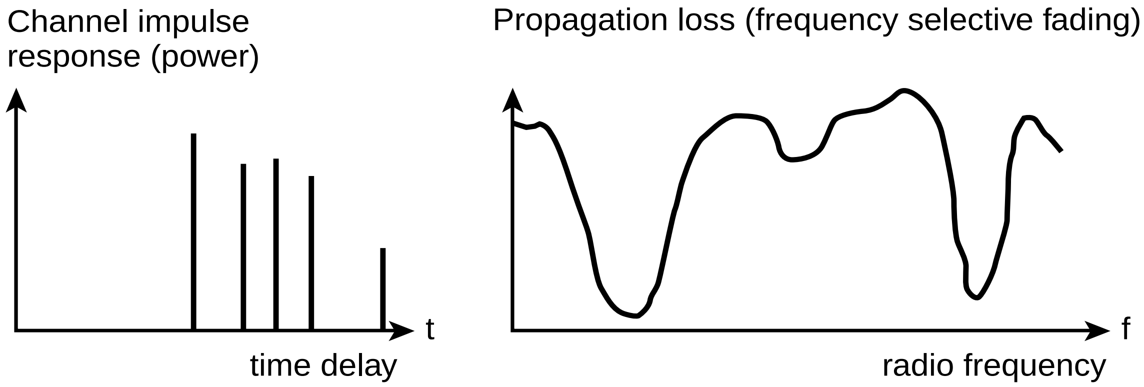
Figure 5.3

Another effect of multipath propagation is **time dispersion** due to varying propagation delays. The effect is that the impulse response of the propagation channel is spread out. The amount of time dispersion is roughly described by the **delay spread**. Fourier transform of the impulse response gives the **frequency correlation function** that describes the characteristics of the frequency selective fading. This is roughly characterized by the coherence **bandwidth**. See figure 5.4a. Large delay spread can be an disadvantage, as channel equalization might be necessary, but on the other hand could make frequency or multi-path diversity possible. See next section.

Another important Fourier pair is the **doppler spectrum** and the **time variations of the impulse response**. The doppler spectrum is roughly characteriszed by the **doppler width**. The rate of the time variations is roughly described by the **coherence time**. A disadvantage with large dopplerwidth might be that the coherence time becomes so short that it might be necessary to adapt the channel equalizer to the variations of the impulse response during each time slot.

According to the general characteristics of Fourier pairs, the coherence bandwith is inversely proportional to the delay spread and the coherence time inversely proportional to the doppler width. See figure 5.4b.

Time dispersion or frequency selective fading



Disadvantage:
Adaptive channel equalization needed of wideband signals

Advantage:
Makes frequency diversity possible

Figure 5.4a

Impulse response (Delay spread s) $\Longleftrightarrow$ **Frequency correlation function (Coherence bandwidth B_c)**

Doppler spectrum (Doppler width B_d) $\Longleftrightarrow$ **Time correlation function (Coherence time T_c)**

$$B_c \sim \frac{1}{s} \qquad T_c \sim \frac{1}{B_d}$$

Figure 5.4b

At the cellular design, the required protection ratio without fading must be increased first with multi-path fading margin to get the protection ratio with respect to the local mean. Then an additional shadow fading margin must be applied to get the minimum value of the global average of the protection ratio.

These propagation characteristics determine the worst case both with respect to the minimum value of the wanted signal C and the maximum value of the total cochannel interference I (global average). The minimum C and the maximum I determine the required global average (C/I) and thus the reuse distance, see figure 5.5.

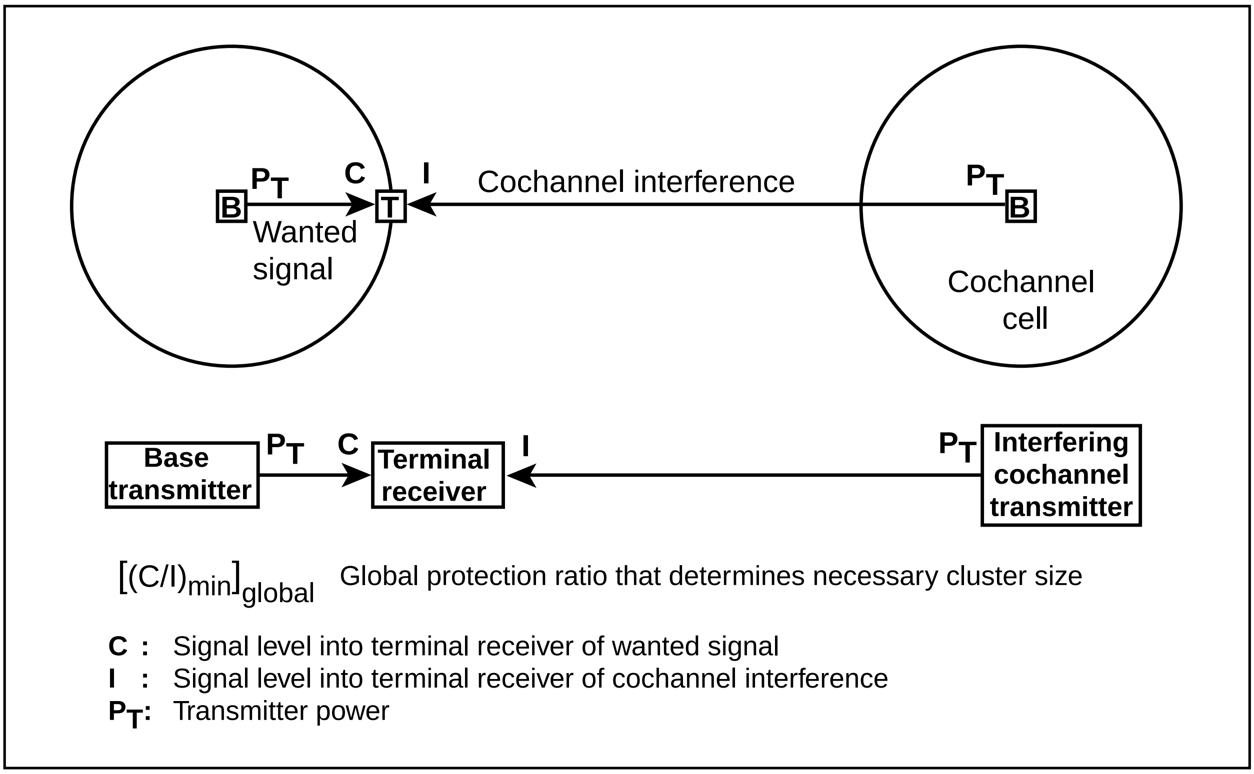


Figure 5.5

The minimum value of C for a certain value of the average C value can be increased by diversity, which gives reduced fading margin (diversity gain). See figure 5.6.

The mobile propagation channel

Distance-dependence of the Global Average
- received signal decreases with 4:th power of antenna separation
- makes frequency reuse possible in cellular systems

Fading of the local average due to shadowing
- large fading margin necessary for high geographic availability
- the fading margin can be reduced by macro diversity (diversity gain)

Fading of the instantanious signal level due to multi-path
- large fading margin necessary for good transmission quality
- the fading margin can be reduced by micro diversity (diversity gain)

Figure 5.6

The maximum value of I can be reduced by interference averaging, power control, discontinuous transmission (DTX). As mentioned above, these means are more effective at interference-limited systems. They are therefore discussed later (section 6). An overview of the situation is given in figure 5.7.

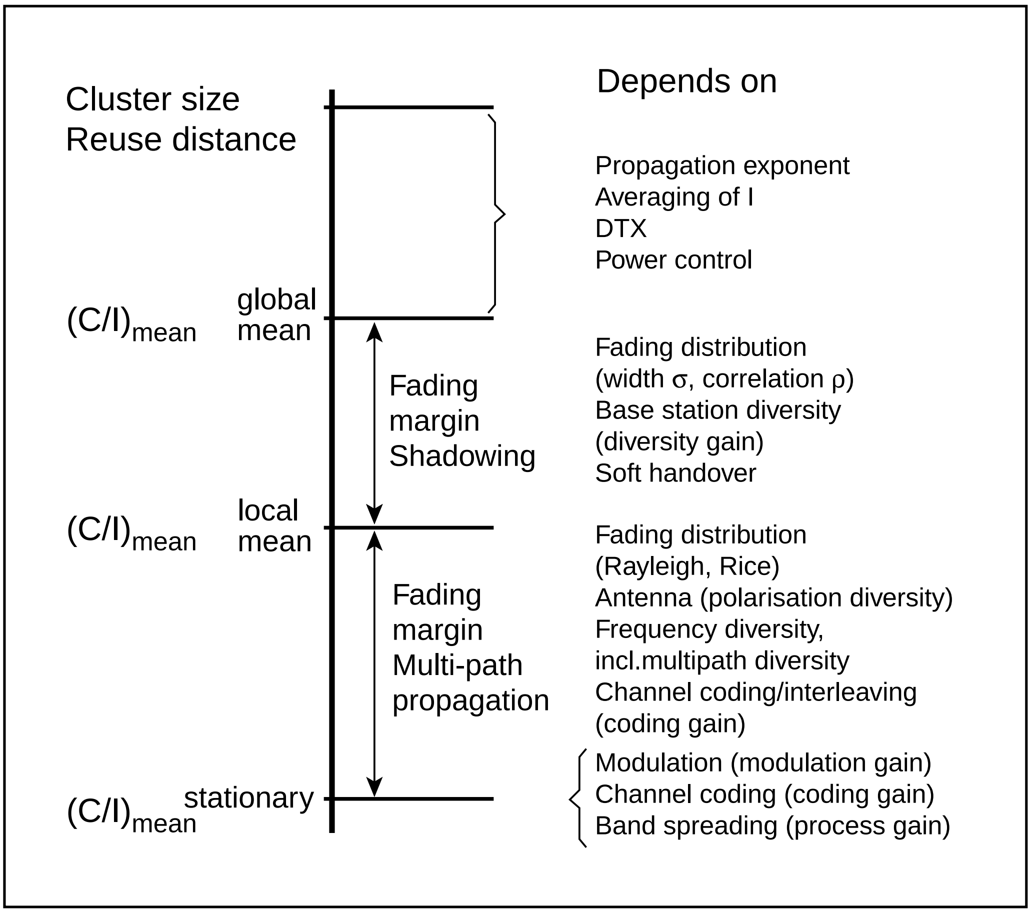


Figure 5.7

5.2.2 Micro diversity

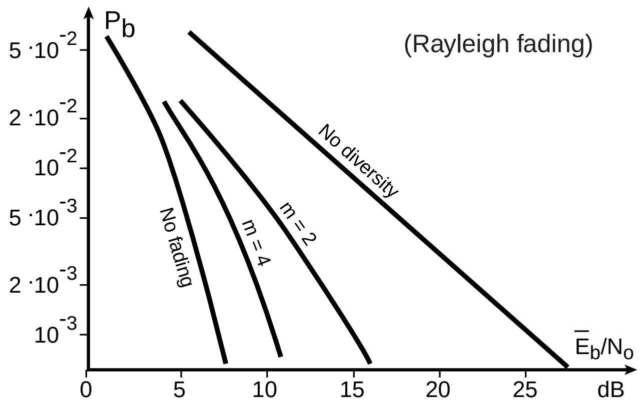
The effective diversity gain is improved by using many diversity paths. See figure 5.8. Due to the large variations of the propagation channal between different propagation situations (amount of time dispersion, terminal speed), it is also advantagous to combine several types of diversity, as one type of diversity might be more efficient for certain channel conditions and another type of diversity for other conditions.

The effective diversity gain is improved by combining several diversity arrangements:

Antenna diversity
Frequency diversity
Multi-path diversity
Time diversity (Channel coding)

Reasons:

1. Increased number (m) of diversity paths



2. Different characteristics of the propagation channel (delay spread, max doppler frequency) permit different types of diversity

Figure 5.8

For instance, channel coding is not very efficient against burst errors, frequency diversity only works if the time dispersion is sufficiently large to give low enough correlation bandwidth.

Also, multipath diversity is only effective, if the delay spread is enough so that the signal processing in the receiver can discriminate between the different paths (requires relatively large modulation bandwidth).

As can be seen from the figure, most of the diversity gain is obtained already with two diversity paths, the gain by adding more and more paths becomes less and less.

From a system point of view, the micro diversity bridges over the fading dips, thus reducing the rate of bit or block errors (improved transmission quality). The result is that the fading margin for a given error rate can be reduced. The fading margin relates the protection ratio without fading with the protection ratio with respect to the local mean of C and I. Also the fading of the interfering signals has a certain influence on the fading margin. However not very much, as the upfading is much less than the downfading, and the moderate variations are fairly well averaged out.

The combined effect of the modem arrangement and micro diversity is described by the detection characteristic. If we consider cochannel interference, we have the principle relation scetched in figure 5.9. The detection characteristic gives the relation between the local average of C/I and a suitable quality measure, such as the bit error rate after the signal processing in the receiver. A similar curve gives the relation between C/N or Eb/No and the transmission quality.

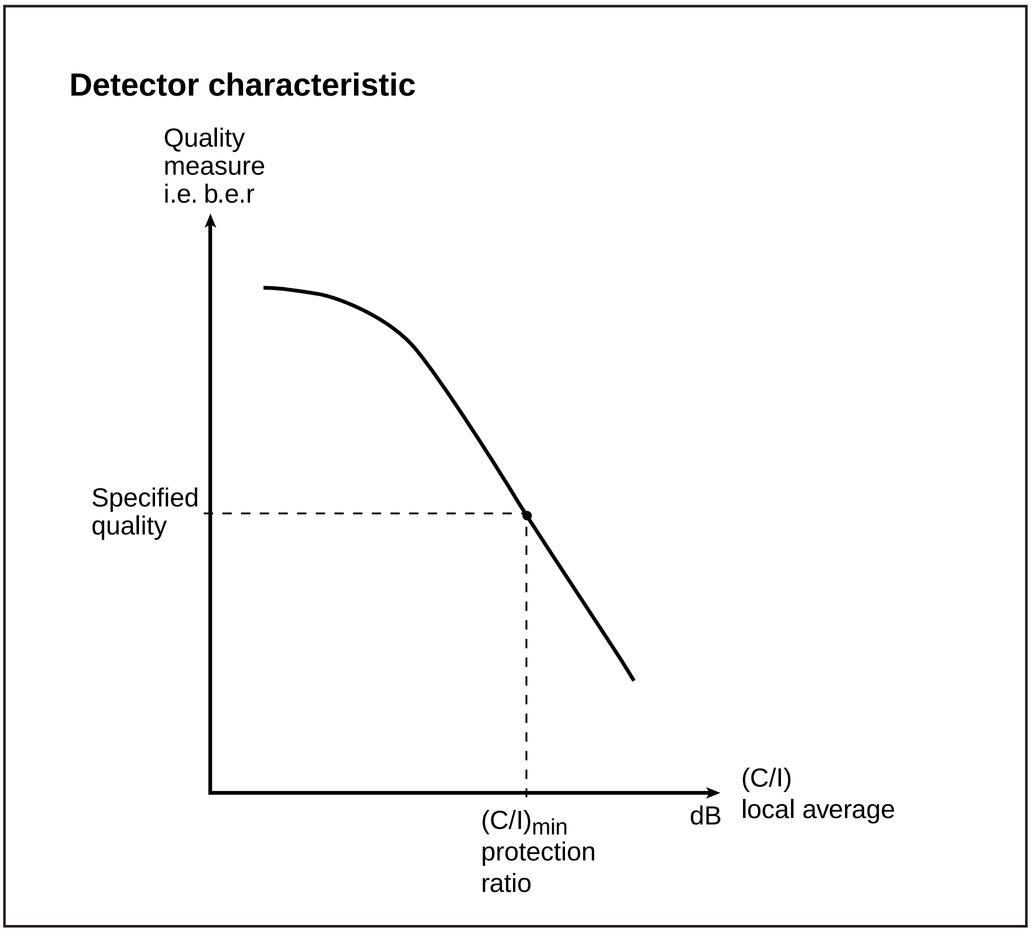


Figure 5.9

Another measure of the transmission quality, used in connection with GSM, is the frame erasure rate. Frame erasure occurs when the FEC (channel coding for error correction), protecting the most critical bits in a speech frame, is unsuccessful. When the required transmission quality is specified, the protection ratio with respect to the local mean follows from the detection characteristic. The protection ratio is the value of C/I that corresponds to the specified transmission quality.

The situation is somewhat more complicated in connection with data transmission. In this case, the allowed ber is often much lower than for speech. Therefore the most important use of channel coding is error detection in connection with ARQ (backward error control). When the C/I or C/N at the receiver input goes down, the retransmission rate due to block errors increases, which results in reduced throughput. Another effect of ARQ is increased transmission delay. With ARQ, additional quality measures are the transmission delay and the throughput.

5.2.3 Macro diversity, soft handover.

Geographic availability

As mentioned above, a probable future scenario is that a considerable part of the wire connected telephones are replaced by wireless personal terminals. However, this development is only possible if both the frequency economy is much improved (i.e. by introduction of micro and pico cells and reduction of the cluster size), and the availability of the connections are nearly as good as for the fixed network. The blocking probability must be small and the cell coverage (geographic availability or service area reliability) high.

Therefore, only a very minor part of the nominal coverage area of a cell should be isolated from the network due to excessive shadowing (outage or radio hole). The shadowing has an impact both on the noise-limited performance (link budget) and on the interference limited performance. To simplify the situation these are often treated separately at a rough analysis. Presently 90 - 95% coverage is generally acceptable, but in the future 98 - 99% would probably be required.

Macro diversity

Implementing a system with high cell coverage and seamless handover becomes more difficult with micro and pico cells. The shadow effects are more pronounced and handovers for mobile terminals must be made more often. The consequences are:

- Increased handover rate and increased requirements on fast (seamless) handover results in a large increase of the system signalling which takes up transmission capacity.
- The frequency economy is decreased, as the requirement of improved coverage leads to higher fading margin in respect of shadowing and therefore larger reuse distance (cluster size).

In order to comply with increased requirements both with respect to service area reliability and frequency economy, future mobile telephone systems must include **macro diversity**, preferably in combination with **soft handover**. Soft or seamless handover means that the macro diversity is arranged so that simultanious connections are established between the two base stations, involved in the handover.

The basis for **macro diversity** is that adjacent base stations have partly overlapping coverage areas and more or less uncorrelated shadow fading so that holes in the coverage from one base station in most cases can be filled in by a nearby base station. See figure 5.10.

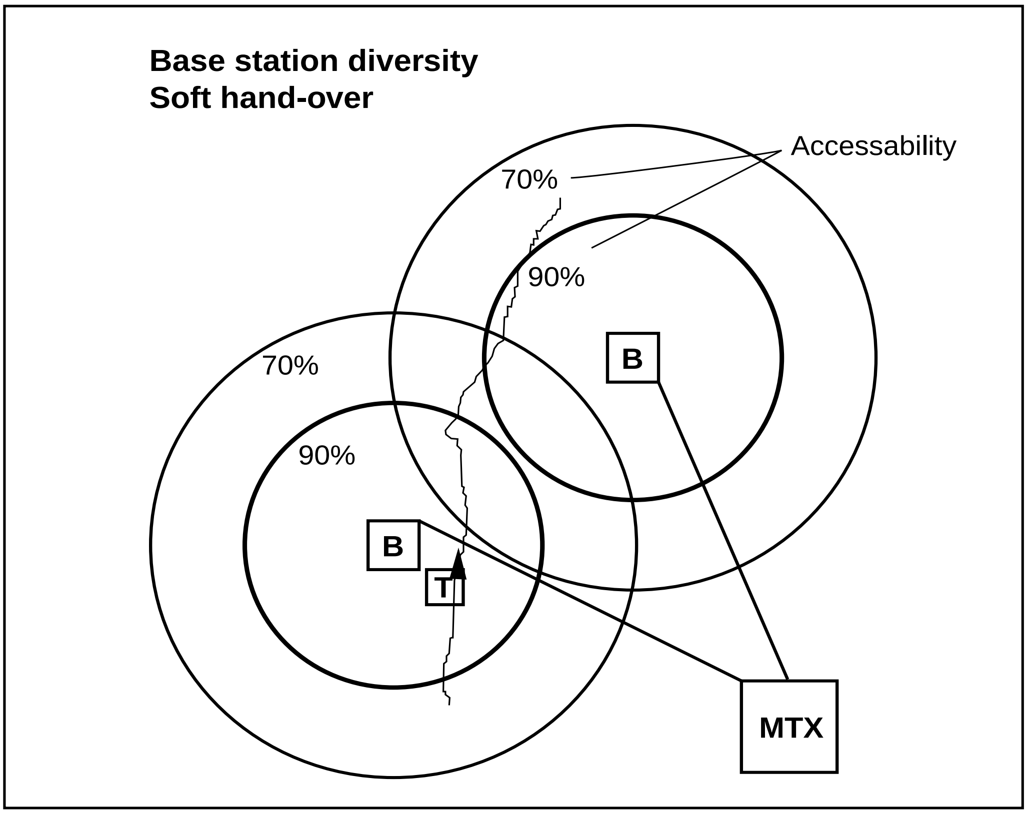


Figure 5.10

The effect of macro diversity is that the fading margin to obtain a specified availability can be reduced. See figure 5.11.

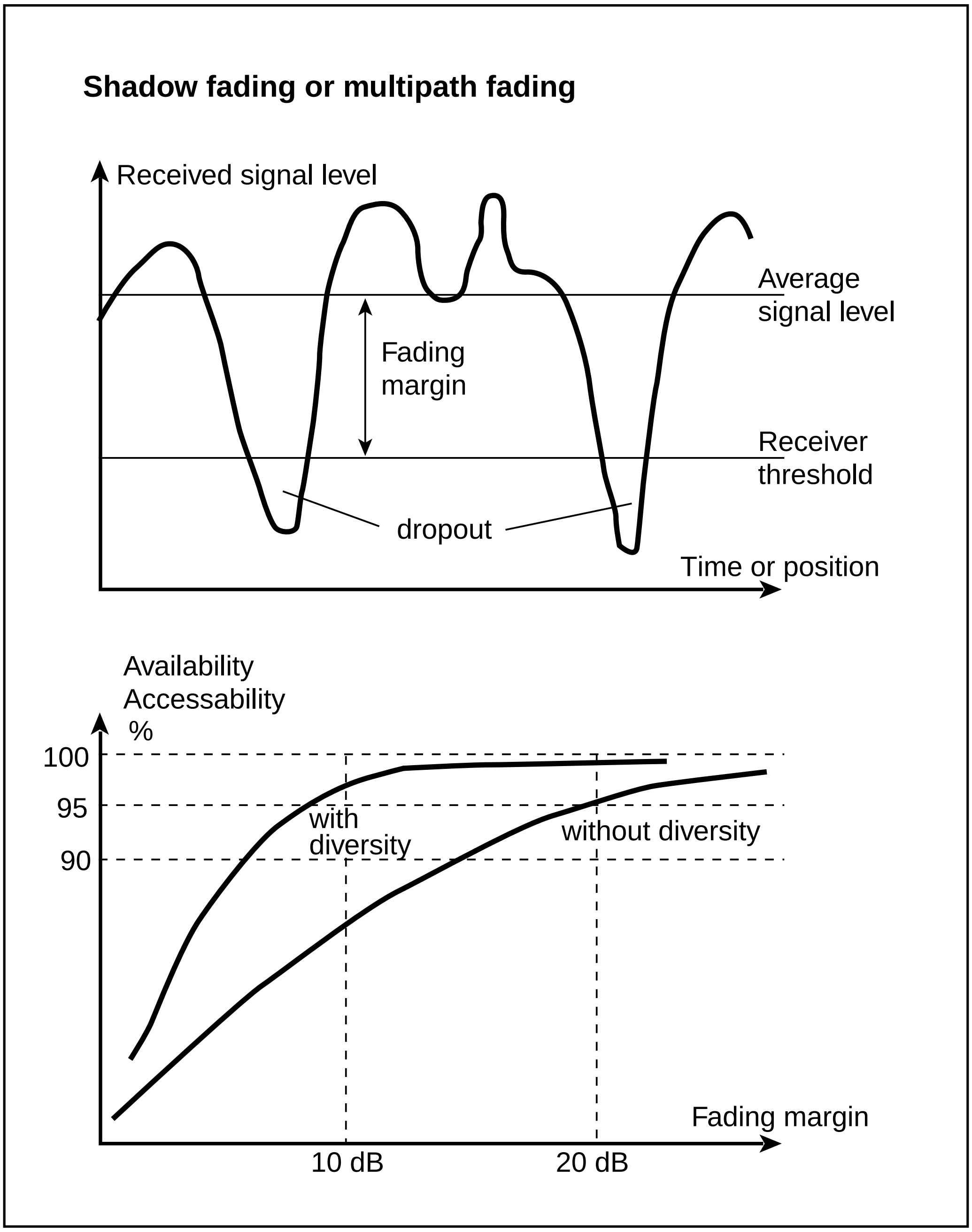


Figure 5.11

The system architecture must allow the connection between a terminal and the fixed network to be established via two or more base stations. The simplest arrangement (hard handover) is very fast handover, which at each instant selects that base station, which gives the best transmission quality (best local average of C/N or C/I). This corresponds to switching diversity. (The handover algoritm at GSM is fast enough to give a moderate amount of diversity gain against shadowing in a system with macro cells.) However, the requirements on handover algoritms and associated signalling are considerable especially in connection with pico and microcells. Connection dropouts at handover cannot be completely eliminated.

More effective macro diversity and possibly also less system signalling might be obtained through **soft handover**.

This is fairly simple in the inward direction, i.e. from terminal to the network. The principle is the same as the traditional voting arrangement, according to which a large geographic area is covered by a network of fixed receivers, connected to a selector at the MTX, see figure 5.12. The quality of the signals from the different receivers is monitored. The receiver (cell), which gives the best quality, is selected and at the handover the new connection is established before the old one is released.

More advanced diversity arrangements are also possible. The signals from several cells can be combined before the channel decoding or even before the data detection. Macro diversity in the inward direction gives no additional loading of the spectrum (but requires additional transmission capacity in the fixed part of the radio network).

Voting arrangement to increase coverage area

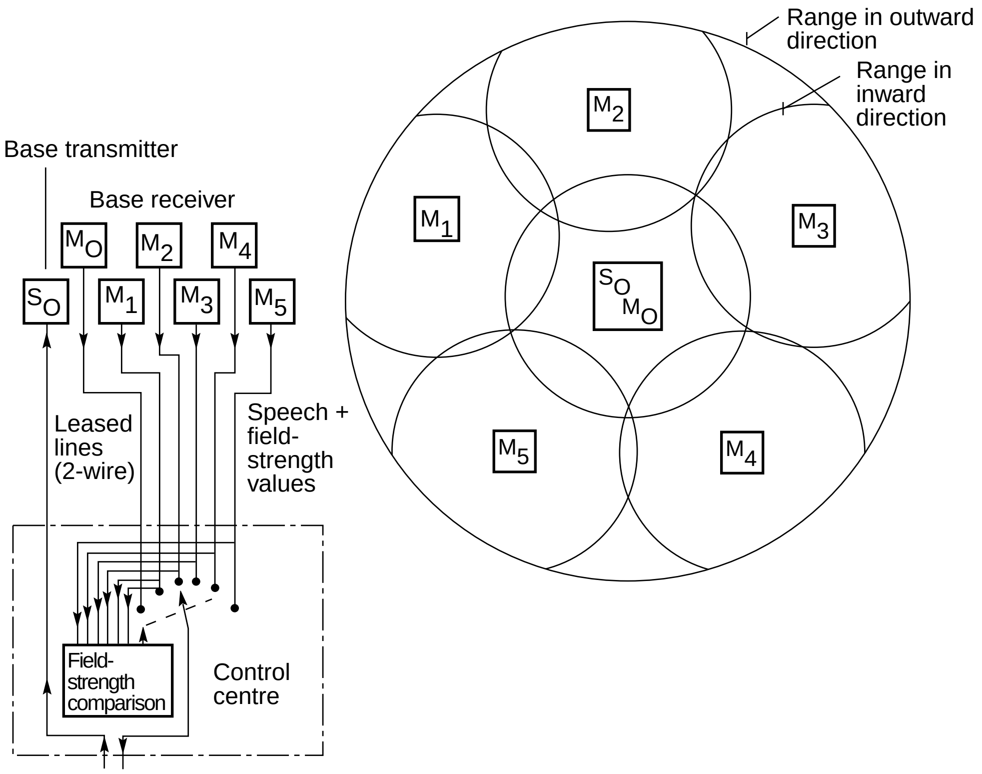


Figure 5.12

In the **outward direction** the over-all network control must determine which combination of base transmitters that shall be engaged to etablish the connection with the terminal. As in the inward direction, different diversity arrangements can be utilized in the terminal. Soft handover can be implemented by packet radio or TDMA (the same information is carried by two or more packets or time slots from different base station sites), or by Direct Sequence CDMA (RAKE-arrangements, see section 6).

A further possibility is multicast using relatively wide radio channels (i.e. in connection with TDMA) so that the modulation bandwidth exceeds the correlation bandwidth of the propagation channel. This means that the equalizer, in addition to discriminate between signals corresponding to different propagation paths from the same transmitter, can also discriminate between signals from different base stations, which transmit identical signals. In this way both micro diversity and macro diversity can be obtained.

A drawback of macro diversity with soft handover in the outward direction is increased loading of the radio spectrum. Macro diversity should therefore only be used during those time intervals, when otherwise insufficient transmission quality is obtained at the terminal.

The different diversity arrangements, discussed above, decrease the differerence between the minimum value of C, and the corresponding global average. Most of the signal variations due to fading is compensated for. To reduce the cluster size further, it is also important to reduce the impact on the system from the variations in the level of the total cochannel interference I. This is possible through suitable averaging of the interference from many cochannel transmitters. This is discussed in section 6 in connection with interference-limited systems.

From a system point of view, macro diversity and averaging of cochannel interference increase the availability for a given cluster size and required local average of the protection ratio. The situation is very complicated and these relations must generally be evaluated by means of Monte Carlo simulations. The placement of cochannel transmitters are varied, and for each propagation path a random value of the shadow fading is applied. If power control and DTX are used, this is also taken into account in the simulations. The result of the simulation is that for each cluster size a curve is obtained, which relates the availability and the required protection ratio (local average). See figure 5.13.

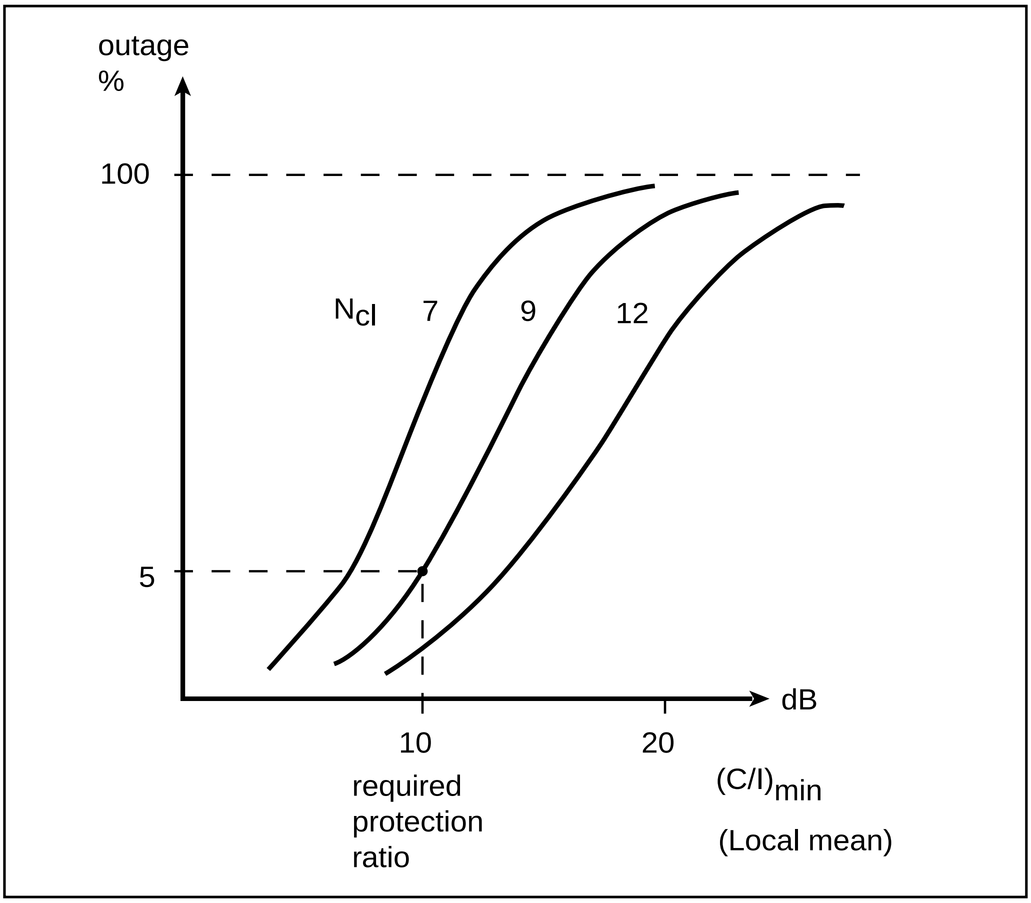


Figure 5.13

The required protection ratio is obtained from the detector characteristics (figure 5.9). For a system, whose performance is given in figure 5.13 and in addition requires a protection ratio of 10 dB, a cluster size of 9 would be required if the required availability is 95%, i.e. 5% outage is permitted.

The result of simulations of a specific case is shown in figure 5.14.

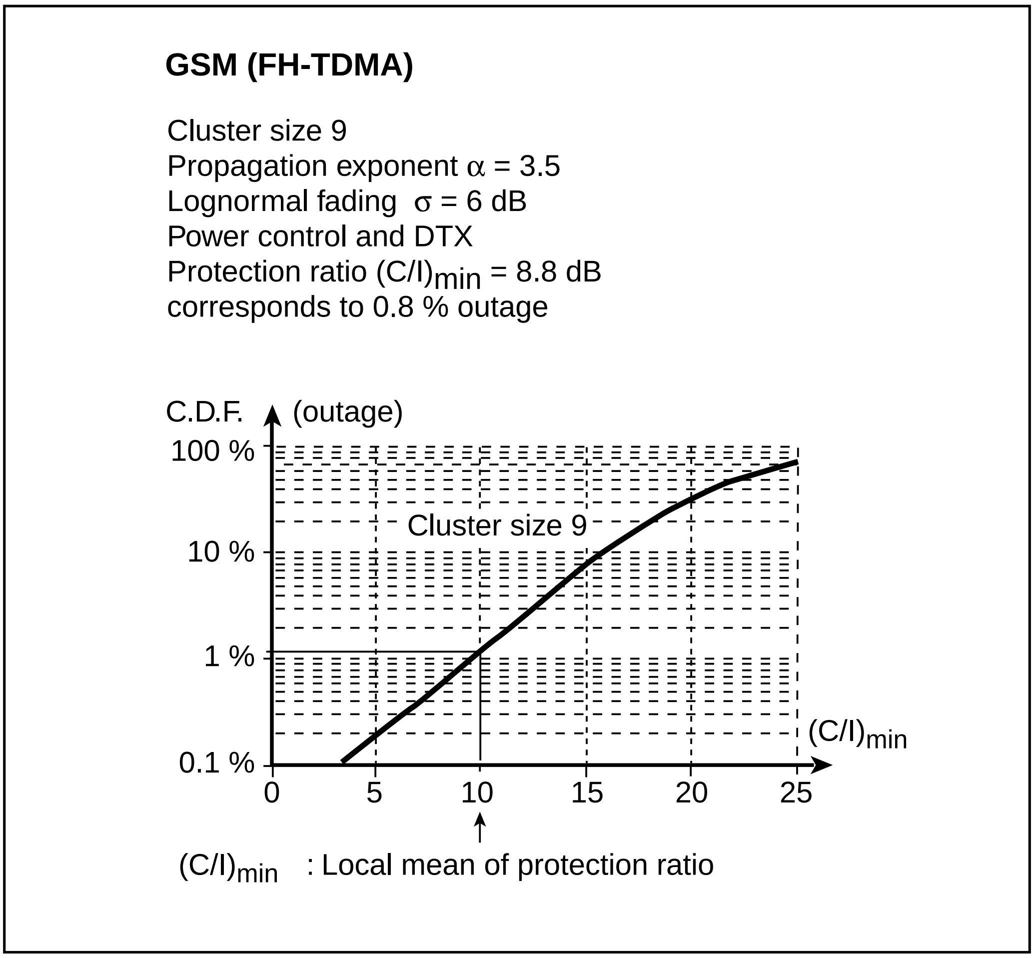


Figure 5.14

It follows from the figure that a C/I (local average) of 10 dB gives an outage of about 1% (availability 99%) with a cluster size of 9.

5.2.4 Summary

The effect of multipath propagations and shadowing and corresponding diversity arrangements are summarized in figure 5.15.

Effects of multi-path propagation and shadowing

Multipath propagation - Rayleigh fading
- Fading margin
- Micro diversity
  - Number of diversity paths
  - Pre- or post-detector combining
  - Diversity gain

Doppler spectrum - coherence time of channel
- Spectrumwidth proportional to radio frequency and terminal speed
- Determines required adaption rate of equalizer

Multi-path propagation - time dispersion, frequency-selective fading
- Increased ber (without equalization)
- Multi-path diversity (if enough modulation BW)
- Frequency diversity (if system bandwith is larger than the coherence bandwith)

Delay spread - correlation bandwith
- Depends on cell size
- Determines if equalization needed
- Small correlation bandwidth ⇒ frequency diversity possible

Channel coding handles short fading dips
- Coding gain, interleaving
- Hard and soft decoding
- Long dips need support of: interleaving, frequency hopping

Base station (macro) diversity against shadowing
- Soft or hard hand-over
- Diversity gain

Figure 5.15

This is followed up in figure 5.16 and 5.17, which summarize different means to reduce the cluster size. Measures mentioned in figure 5.16 are also processing gain through band spreading and high-gain antennas (narrow lobes). These measures are discussed in sections 6 and 9.

Means for reduction of D/d (Cluster size)

1. Means, which reduce the fading margin for multi-path fading
- a. Increase of the number of diversity channels
- b. Independent fading of diversity channels
- c. Optimum combining of diversity channels
- d. Matching to variations of the propagation channel
  Use several types of diversity:
  channel coding
  antenna separation or polarization diversity
  frequency or multipath diversity

Characteristics of multi-path channel:
fading structure: Rayleigh or Rice
impulse response: delay spread - coherence bandwidth
width of doppler spectrum: correlation time

2. Means, which reduce the fading margin for shadowing (macro diversity)
- a. Increase of the number of diversity channels
- b. Optimum combining of diversity channels
- c. Soft handover

Macro diversity in outward direction gives additional loading of spectrum

3. Means, which reduce the influence of cochannel interference
- a. Effective averaging of interferers
  (incl. DSI/DTX and dynamic power regulation)
- b. Bandspreading for process gain
  (requires increased system bandwidth)
- c. Advanced high-gain antennas at base stations

Figure 5.16

The normalized reuse distance and the cluster size are directly related to the required $(C/I)_{mean}$ (global mean)

$(C/I)_{mean}$ is determined by the condition

$(C/I)_{min} > K_I$ (protection ratio without fading)

which must be satisfied for almost all connections

K_I: lowest C/I value for acceptable transmission quality

Early cellular systems:

$(C/I)_{mean} >> (C/I)_{min} = C_{min} / I_{max}$

That is, large cluster size
low frequency economy

This was caused by

shadow fading of C
multi-path fading of C
large variations of I

Measures to decrease the required $(C/I)_{mean}$:

1. Combinations of several diversity arrangements (reduced fading margin)
2. Statistical averaging of many interference signals
   or
   Dynamic channel allocation based on C/I measurements
   (decreased I_{max})

Figure 5.17

5.3 Dynamic adaption of system parameters

Power regulation

Dynamic transmitter control continuously adjusts the output power so that C/N and C/I are limited to what, which with a certain safety margin corresponds to the required minimum value for acceptable transmission quality. This reduces the average value of the total cochannel interference. If suitable averaging of the cochannel interference is introduced, e.g. through frequency hopping, a certain reduction is also obtained of the worst case interference, which the system must tolerate with adequate transmission quality. Another advantage is reduced power drain from terminal batteries.

Bandwidth on demand

As mentioned above, it is not enough to adapt the system to the maximum data rate demanded by different type of services. A further important capacity gain can often be obtained through dynamic adaption of the offered channel capacity to the strongly varying demands during a call (dynamic demand assignment or bandwith-on-demand). This is discussed in section 5.5. One example of such variations is a two way speech connection, for which each direction is active during less than 50%. Many types of data services have much higher burstiness.

For a speech connection a further gain in capacity would be possible by adaption of the data rate to the varying need during an active speech interval. For certain radio systems with extremely fast dynamic capacity allocation (DS-CDMA), speech coders with variable data rate are used.

Adaption to system traffic load

The main limitation within areas with concentrated traffic is unsufficient capacity due to limited system bandwidth. The digital transmission should therefore be optimized for the best frequency economy (combination of channel spacing per traffic channel and cluster size), even if this means increased requirements on $(C/N)_{min}$. This might not be a serious limitation, as the linkbudget is less stringent for micro and picocells (must be used to cover areas with large traffic density).

The situation is the opposite in rural areas, where there is no spectum shortage. Here, the main system limitation is the link budget, which determines the maximum useable cell sizes. In this case it would be advantageous to adapt the transmission characteristics for the best noise-limited receiver sensitivity, even if this means larger bandwidth per traffic channel. One possibility is use FEC with low rate codes. Also modulation methods which maximizes the transmitter efficiency can be used (e.g. constant envelope modulation).

Adaption to channel quality

This is a similar case to the one discussed above. For a high-quality radio channel (high E/No and C/I), low error rates is obtained even with high level modulation, i.e. 16QAM, and without FEC. When the channel quality becomes marginal, it is suitable to introduce FEC and use 4QAM.

Examples are the new data services of GSM for which a future option will be more spectrum efficient modulation (see appendix 1 of module DM1). For a very good radio channel (high C/N and C/I) the highest throughput is obtained with 16QAM and no FEC. When the channel quality gradually detoriates, it is optimum to keep the retransmission rate fairly low even if this means reduced system data rate. 4QAM is used instead of 16QAM and more and more FEC is introduced.

Adaption to maximum user data rates (cell sizes)

The fairly low user date rates offered in macro cells could motivate the use of fairly low system data rates. Much higher modulation bandwidths are used in micro and pico cells. An example of this adaptions of system parameters is the ATDMA system, see appendix 4.

5.4 Dynamic demand assignment

5.4.1 Introduction

In the first generationen of digital mobile telephone systems, a connection with sufficient peak capacity is allocated continuously during the call. This arrangement gives bad capacity utilization, if the average data rate of the source signals is considerably lower than the peak value (the signal has high burstiness). As an example, a two way speech connection is active in each direction only about 40% of time, i.e. the maximum data rate is 2.5 times larger than the average. If we also take into account that a radio channel cannot be allocated to calls 100% of the time (would lead to excessive blocking), the result is that a one way trunk connection (radio channel) cannot be utilized more than about 1/3 of the time.

The time utilization can be much less at interactive data traffic between a terminal and a host or data base. Another difference between speech and data transmission is that many data connections are unsymmetric, i.e. the information flow is much less in one transmission direction. The average rate from the terminal is in many cases much less than the rate from the server.

Therefore, the utilization of the total transmission resource, when servicing bursty signals, can be considerably improved by letting a number of connections share a common channel resource by means of fast dynamic channel allocation (demand assignment). During intervals when a connection needs less than the allocated peak capacity, another connection can use the excess capacity.

A limiting case, corresponding to perfect averaging, is that the required total transmission capacity of a cell is given by the sum of the average capacities used by the different users. It is possible to get fairly close to this limiting case, if a very large number of users with independent traffic structures share a common transmission resource. However, the optimum combination of fast dynamic channel allocation and high channel utilization can only be obtained in an interference-limited system. See section 6. In this section other, less perfect possibilities are discussed.

If a typical connection has an unsymmetric data flow, the bandwidth utilization can be further improved if the available bandwidth can be split up unequally between the two directions. This is especially simple to implement if TDMA is combined with TDD. An example is DECT. However, a limitation with TDD is reduced capacity due to the need for large guardbands between the outward and reverse time slots, if the variations in propagation delays between different connections are of the same order as the slot length. That might limit the use of TDD to small cells. (Another system complication with TDD is that time synchronization between nearby cells would be required).

5.4.2 DSI

The utilization of a group of trunks in the public telephone network is sometimes improved by means of DSI (Digital Speech Interpolation). See Fig. 5.18. If DSI is introduced for a large trunk group, the effective trunk utilization is roughly doubled, i.e. the number of trunks can be reduced by a factor of two. However, the DSI gain is smaller with small trunk groups due to less efficient statistical averaging. The reason is that if the number of trunk lines (radio channels) are decreased, the blocking probability will increase due to increased risk that more connections are active than the number of trunks.

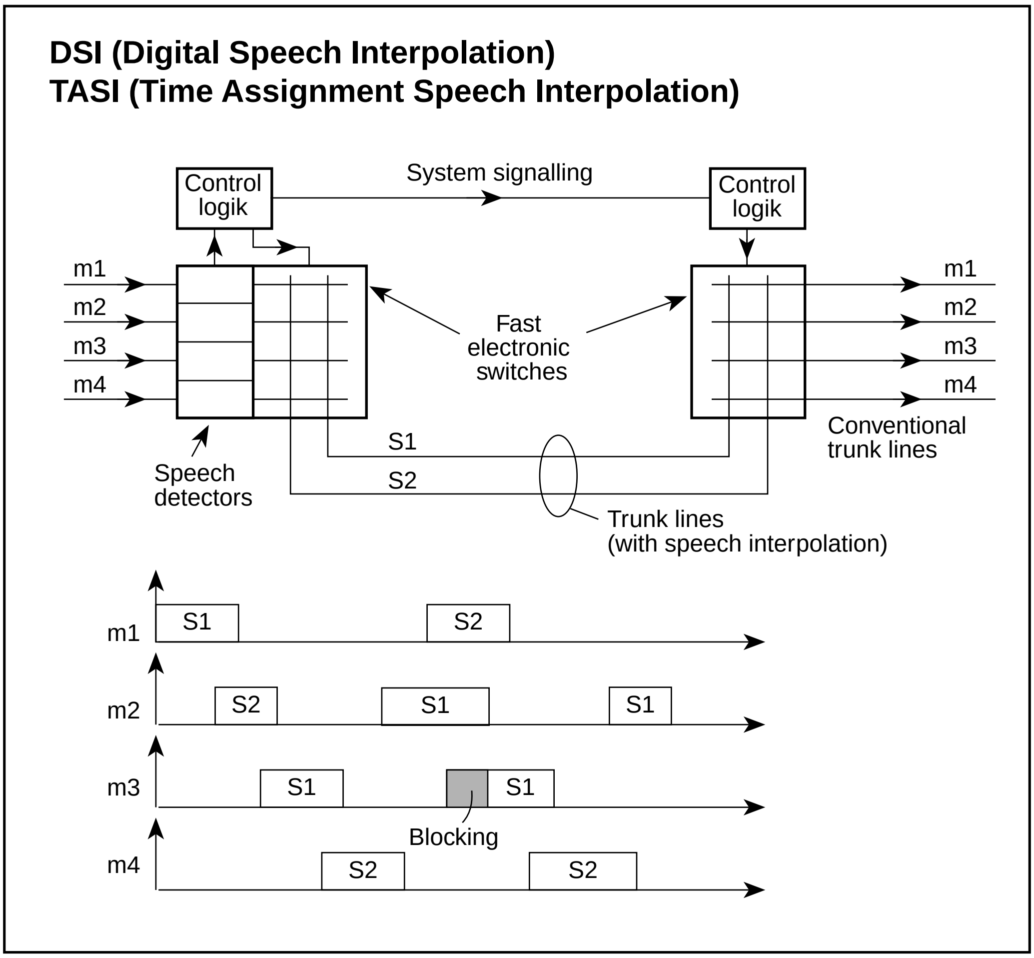


Figure 5.18

A similar concept, used in connection with packet transmission is statistical multiplexing. The performance can be improved by buffering the incoming data blocks, if the outgoing line is fully loaded. The drawback with buffering is increased and variable transmission delays.

At a digital mobile telephone system based on TDMA (or TDM in the outward direction) the corresponding improvement is possible by allocating time slots dynamicly after the varying demands from the different connections (demand assignment, statistical multiplex or asynchronous TDMA). The allocation is controlled on the basis of information from speech activity detectors. The detectors indicate during which intervals each source is active. It is especially important to indicate immediately the start of an active speech interval (”talk spurt”) so that a time slot can be allocated. Otherwise the first part of the talk spurts will be cut, which leads to a considerable quality degradation.

In principle, the procedure is simple in the outward direction (single access), but additional signalling must be introduced to inform each terminal which time slots to receive. In the inward direction (multiple access) it is considerably more complicated to reserve capacity to the different terminals after the rapidly varying demand. New multiple access methods, such as packet radio must be used, see section 5.4.3.

The GSM terminals make use of Discontinuous Transmission DTX, which has certain similarities with DSI. Speech detectors determine intervals with speech pauses, and the transmitter is switched off during these intervals. The main reason is the need to minimize the current drain from the batteries. However, if frequency hopping is used for averaging of cochannel interference, DTX will give a certain decrease of the maximum interference level, which might permit a reduction of the cluster size. This is discussed in connection with interference-limited systems below (FH-CDMA).

5.4.3 Packet radio

Introduction

TDMA gives in principle considerable flexibility in matching the capacity allocations to the peak capacity demands of different users. A traffic channel with large peak capacity is obtained by allocation of several time slots per frame (multi slot). A user, which demands very small capacity, can be offered time slots with large mutual separation (i.e. the half-rate traffic channel at GSM). However, the allocation procedure is fairly slow. Therefore, the basic TDMA arrangement does not permit fast dynamical allocation of transmission capacity to the varying need during a call. This requires further development of TDMA to statistical multiplexing or **packet radio** (see the GPRS of GSM in module DM1). Even with these modifications, compromises are necessary between channel loading (frequency economy) and the speed of the dynamic allocation.

Packet radio, based on a common transmission channel for several simultanious users is a multiple access arrangement, which is related to TDMA. One of the main advantages of packet radio is the possibility to match the offered capacity to the varying needs of the users, incl. the burstiness of the source. Packet radio is related to the corresponding development within the fixed telephone network, where a transport network with packet transmission of both speech and data is used (ATM: Asynchronous Transfer Mode).

There are two major alternatives for packet transmission in a multiple access situation (i.e. in the direction from terminals to the network.) Packet transmission from a base station, which has exclusive right to the channel, to terminals is much easier, as the base station has full knowledge of the total traffic situation.

A **contention arrangement** (risk for collisions) gives fast demand assignment of channel resourses, but the maximum loading of the common channel is only about 30% with one of the simplest arrangements, slotted Aloha. This might be acceptable for short messages or connections with very high burstiness and low average data rate. A **reservation arrangement** permits high loading of the channel without risk for collisions, but the reservation procedure takes time and gives signalling overheads. It cannot give fast dynamic demand assignment. Different combinations of contention and reservation protocols are therefore used in practical applications.

Contention arrangements

Different contention protocols are shown in Fig. 5.19. The simplest arrangement is **pure Aloha**. In this case, a terminal transmits a packet, as soon as enough bits to fill a packet has been delivered by the source. The packet is transmitted without any coordination with the other terminals, sharing the radio channel. Therefore, there is a certain probability that packets from different terminals overlap in time.

The packets collide, and it is then quite probable that none is received correctly due to strong mutual cochannel interference. Therefore, it might be necessary to transmit packets several times. The consequence is both considerably increased and variable transmission delay and reduced throughput.

An important advantage is that many terminals can share a common, wide-band channel without any need of over-all traffic control. Therefore, Aloha gives fast dynamic allocation of channel capacity to the different connections. A typical application for an Aloha packet channel is for the first call from a terminal to the base, when setting up an inward call.

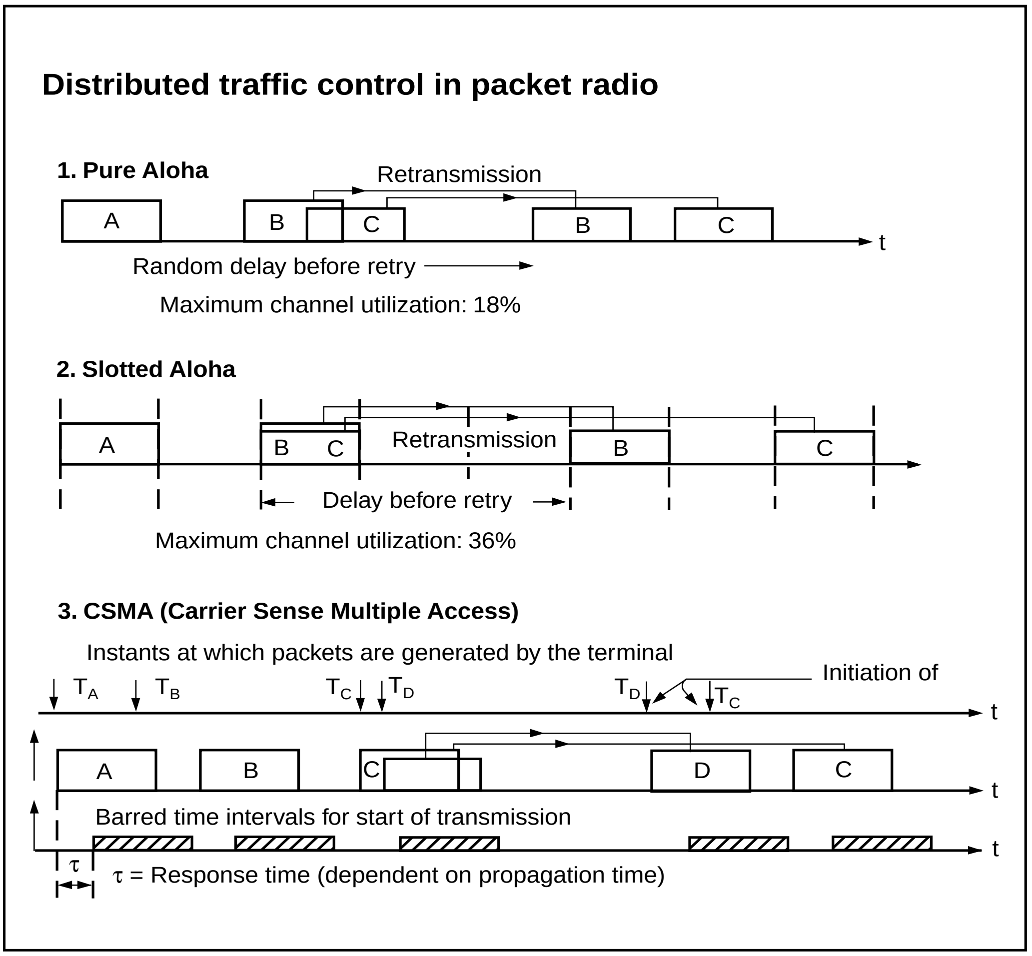


Figure 5.19

The risk of collision can be reduced, and thus the throughput increased compared to the previous case (pure Aloha) by means of **Slotted Aloha.** An example of the use of Slotted Aloha is the inward paging channel at GSM. Slotted Aloha means that the terminals use a common time slot structure (the time slots in the TDMA frames). The packets can still be transmitted without any mutual coordination, but each packet must be fully placed within a time slot. The maximum channel utilization with pure Aloha is about 18% and with slotted Aloha 36%. These are theoretical bounds, which cannot be reached in practical applications.

The risk of collision can also be decreased by using CSMA (Carrier Sense Multiple Access). In this cases, a terminal that has something to transmit, listens on the common channel to check that it is free, before a packet is transmitted. However, it still remains a certain collision risk due to ”hidden terminals” and propagation times. More information about Aloha and CSMA can be found in module S5.

Contention arrangements, with risk for collisions and therefore need for an ARQ with retransmission of lost packets, are difficult to use for two-way speech due to the additional, random transmission delay. As the absolute delays due to retransmissions are reduced, if the system data rate is increased (i.e. reduced packet duration), packet transmission over wide band channels can be used also for speech connections (i.e. using IP-technology also for telephony).

Reservation protocols

An alternative protocol, suitable especially for long messages (i.e. file transfers) and for delay sensitive applications is based on reservation of channel resources. As there are no retransmissions due to colliding packets, higher throughput is obtained and there is no extra transmission delays due to collisions. However, reservation procedures give an additional signalling overhead and it takes time to set up a traffic channel.

The principle of a reservation arrangement is shown in figure 5.20 for the inward direction (multiple access). A similar frame structure as with TDMA can be utilized, but a frame contains several classes of time slots. One part of the time slots can be allocated for data traffic according to Slotted Aloha (i.e. short data messages, each of which can be contained in one time slot), other time slots are also of the Aloha type, but must only be utilized to request the reservation of a number of time slots for speech or long data messages. To bring down the risk for collision between the reservation packets, is the necessary that the reservation Aloha channel is designed for low traffic.

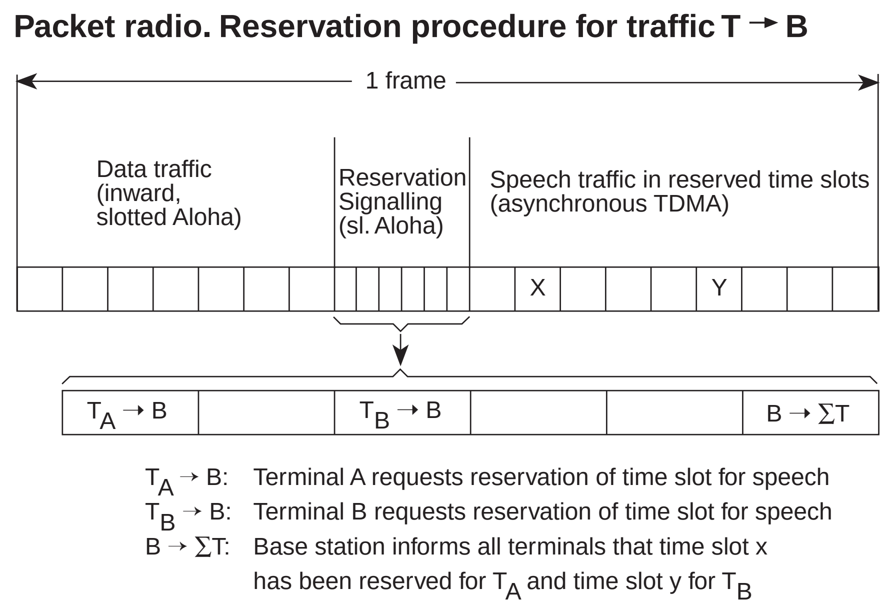


Figure 5.20

The reservation arrangements is suitable for speech connections, but the possibilities to release the channel for other users during speech pauses might be limited by so long reservation time that the first part of a talk spurt must be dropped. When the speaker at a terminal starts a new talk spurt, the terminal requests reservation of a number time slots with a suitable mutual separation, e.g. one time slot per data burst from the speech coder. If the reservation is accepted, the terminal is given access to a number of suitably spaced time slots until the next pause, when the reservation is cancelled. The combination of an Aloha channel for reservation requests from terminals and reserved time slots for traffic is utilized at the ATDMA system (appendix 4). It is also used at the GPRS of GSM appendix 1 of module DM1).

Need for packet headers

The random or dynamic allocation of time slots, as described above (statistic multiplex or demand assignment), puts additional requirements on the signalling to make certain that the packets (time slots) reach the intended receiver. A simple arrangement is to add an identification code (header) to each transmitted packet. This code can consist of transmitter and receiver ID:s and also time information (time stamp) or sequence number.

This means that each packet carries all the required information necessary to route the packet through the network and on the receive side to put together the packets with the right mutual order or timing, so that the original message can be put together. In the general case, different packets can be routed through the network by different paths with different transmission delays. A packet can even be routed through several paths in parallel in order to improve the probability of successful transmission (i.e. soft handover). The packet arrangement works even if the time order of the received packets is changed or if duplicate packets reach the final destination.

There is a certain risk that packets get lost in the network, due to sudden breakdown of node computers and links. To recover lost packages, an end-to-end ARQ arrangement is often introduced. This is called a virtual connection or circuit. (Sometimes another definition is used: a virtual circuit means that at the start of the transmission of a message, the first packet establishes a route through the network, which is followed by all the following traffic packages. This quaranties that packets are delivered in the right order to the receive end).

Store-and-forward

A commonly used feature of packet radio is store-and-forward, see figure 5.21. This means that a packet, which is received by a network node, is stored in a buffer memory. The packet remains in the memory until the next node (i.e. a terminal) sends back a confirmation of successful reception. If no confirmation is received by the transmit side within a resonable time, the packet is retransmitted. Parity bits for error detection must be added to each packet, to make it possible for the receive node to verify the correctness of the packet.

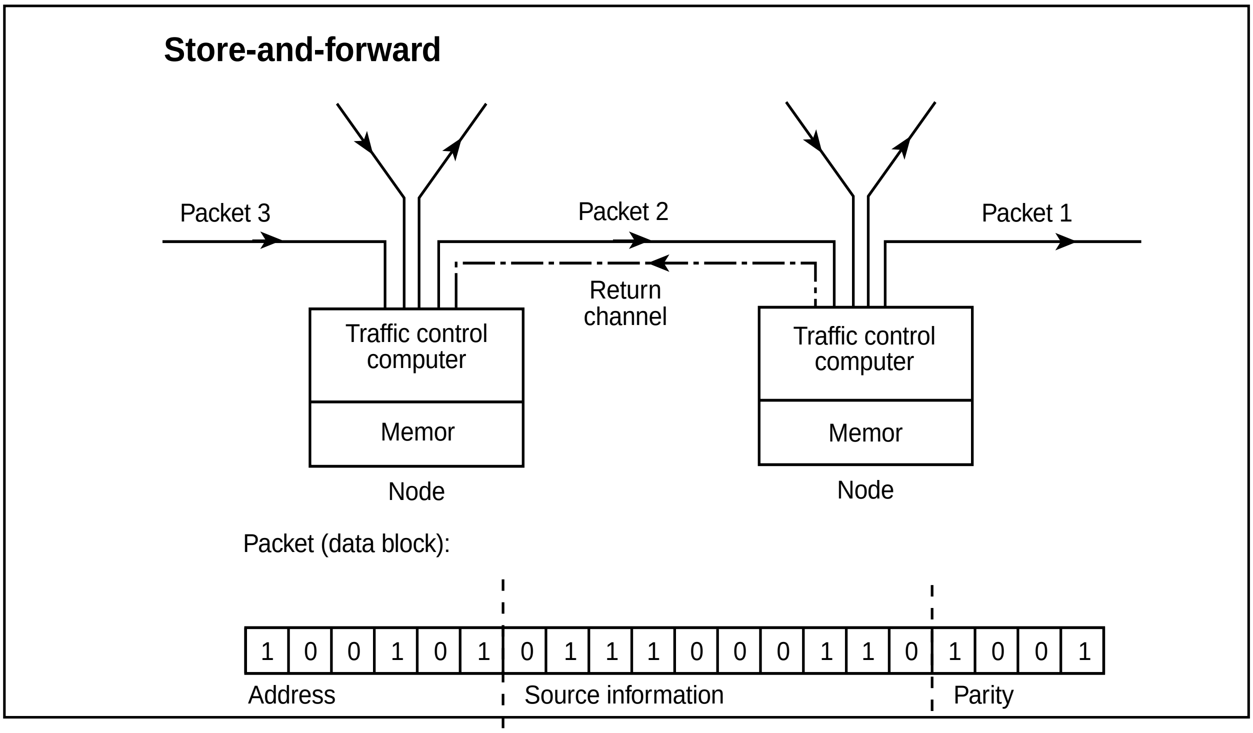


Figure 5.21

The large flexibility in transmission arrangements and system architectures through packet radio can, however, result in reduced frequency economy. As mentioned above, random transmissions of the Aloha type gives low channel utilization - of the order of 30 % in the best case. This is one of the reasons to use reservation arrangements for messages with a length corresponding to several packets. (However reservation signalling in connection with the reservation protocol takes up channel capacity).

The possibility to route packets over several parallel paths is useful in connectionwith macro diversity with soft handover. However,macro diversity requires additional transmission resources both in the radio network and in the fixed network. The principle is shown in figure 5.22, and also discussed in section 3.5.

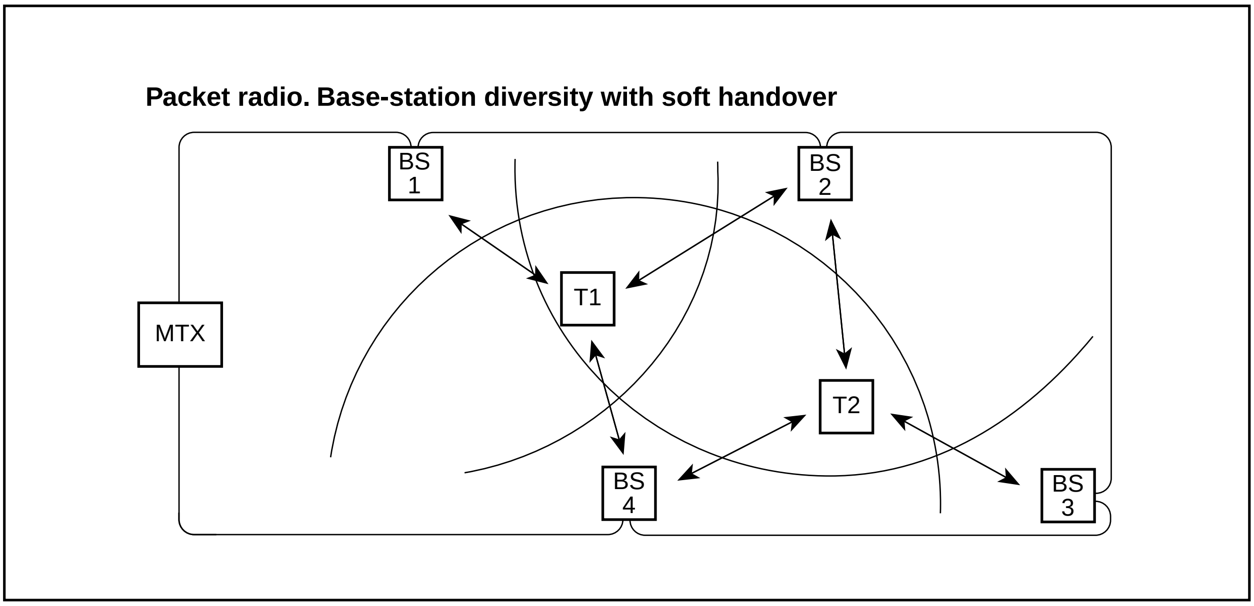


Figure 5.22

Between a terminal and the MTX to which it belongs are set up two or more parallel connections via suitable base stations. Macro diversity with soft handover is possible, if the terminal is situated in the common coverage area of two or more cells. All base stations could be connected to the MTX via a loop, over which the packets circulate.

In the inward direction, a packet from a terminal can be received by more than one base station and sent over to the MTX. Diversity gain is obtained through suitable combination of the information in the received packets, which correspond to the same transmitted package (with the same sequence number).

In the outward direction, each packet intended for a certain terminal, is transmitted from two or more base stations, in whose nominal coverage areas the terminal is situated. Suitable base stations to use for the connection are determined by the network control logic, based on measurements of the transmission quality of the different propagation paths to the terminal (from involved base stations).

Packet transmission is further discussed in the next section in connection with overviews of ATM and IP technology. However, an important difference is that ATM and IP networks have originally been optimized for fixed links between nodes. This means that the error rates over the links are extremely low ($< 10^{-10}$) and also that we have a single-access situation.

5.5 Data transmission principles

5.5.1 Introduction

The requirements on the radio system differs in several respects between speech transmission and data transmission.

a. Typical data services require very low error rates. Over a fading radio channel, this can only be obtained by means of ARQ, based on error detection coding (backward error control) or acknowledgement protocols. This means that a large number of retransmissions will be necessary, if the quality of the propagation channel is marginal. The result is large and variable transmission delays, which generally would not be acceptable for two-way speech connections. The situation can be somewhat improved by adding FEC.

b. Data transmission between terminals or host processors must be controlled by exactly defined protocols. Often the protocols are structured according to an universal hierachical framework called the ISO/OSI model. Concepts from this model have also been used for GSM and DECT.

c. Due to the high burstiness of many types of data signals, the utilization of transmission channels are muched improved by demand assignment of channel capacity. This can be achieved through packet transmission.

d. The telephone networks, until recently mainly used for speech signals, have been defined and developed within the centralized framework controlled by ITU and telecom organizations. The goal has been to develop detailed, universal standards, which is a very time consuming process. Therefore commercial systems have been introduced before the standardization process was completed. The existing infrastructure is optimized for circuit switching. Packet oriented structures, such as ATM, are an overlay on the existing circuit-switched network.

The wide-area data communication networks have been developed through decentralized extensions of different types local area networks used at universities and other research organizations in the U.S. Protocol standards have gradually evolved from experiences and comparations between alternatives, developed by different research groups. The main users have on the same time been data network researchers. A loose control came from ARPA (Advanced Research Project Administration), which supplied a large part of the research funds to develop the ARPANET. The ad-hoc standards were an outcome of this experimentally based process, and these standards have been gradually improved and extended.

When ARPANET growed into the commercially important INTERNET, more stable standards were necessary. This stifled the continuous improvement process. Therefore, the research community in the U.S. are presently engaged in two new research programs: IP2 (INTERNET 2) and NGI (Next Generation Internet). The very high-speed transport network for these programs will be vBNS(very-high-performance Backbone Network Service). This research is given considerable federal support. Similar research goes on in Europe (GHz network).

Research topics, which will studied at the IP2 and NGI programs are:

100 -1000 higher speed than the simple INTERNET connections of today

Quality-of Service guaranties with different traffic priorities

Multicast

Use of a new IP protocol (version 6)

5.5.2 The ISO/OSI reference model

As the complete protocol package needed for the transmission over large data networks becomes extremely complicated, a suitable layered structure is generally used. The objective is to make the internal structure of the different layers decoupled from the other layers. The different layers are connected through well defined interfaces.

Each layer establishes communication with the corresponding layers on the other side of a network connection (peer-to-peer communication), even if the real physical communication between the two sides relies on the lowest layer.

The advantages of structuring the protocol in layers with exactly defined interfaces is that each layer can be optimized separately. In principle, the physical layer can be designed for different transmission media, without any influence on the higher layers. The same applies to the transport layer.

On the telecommunication side, a large effort has been spent for many years on the development of a universal reference model suitable for interconnection of many types of applications. The activity has been supported by ITU and is organized through ISO (International Standards Organization). It is called the OSI (Open Systems Interconnection) reference model. Even if the detailed OSI model has not been directly used in practical applications, the principle of a layered protocol structure has found widespread acceptance.

On the data communication side, structured transmission protocols have also been developed based on a gradual development and geographic extension of the local area networks (Ethernets), used by differerent research organizations mainly in the U.S. The ARPANET gradually developed into the INTERNET of today. INTERNET has not been directly influenced by the OSI model, even if there are considerable similarities.

The complete ISO model consists of 7 layers, see figure 5.23. Of these only layers 1 to 4 are directly involved in the data transfer in the network. The background to the model is the fixed telephone network.

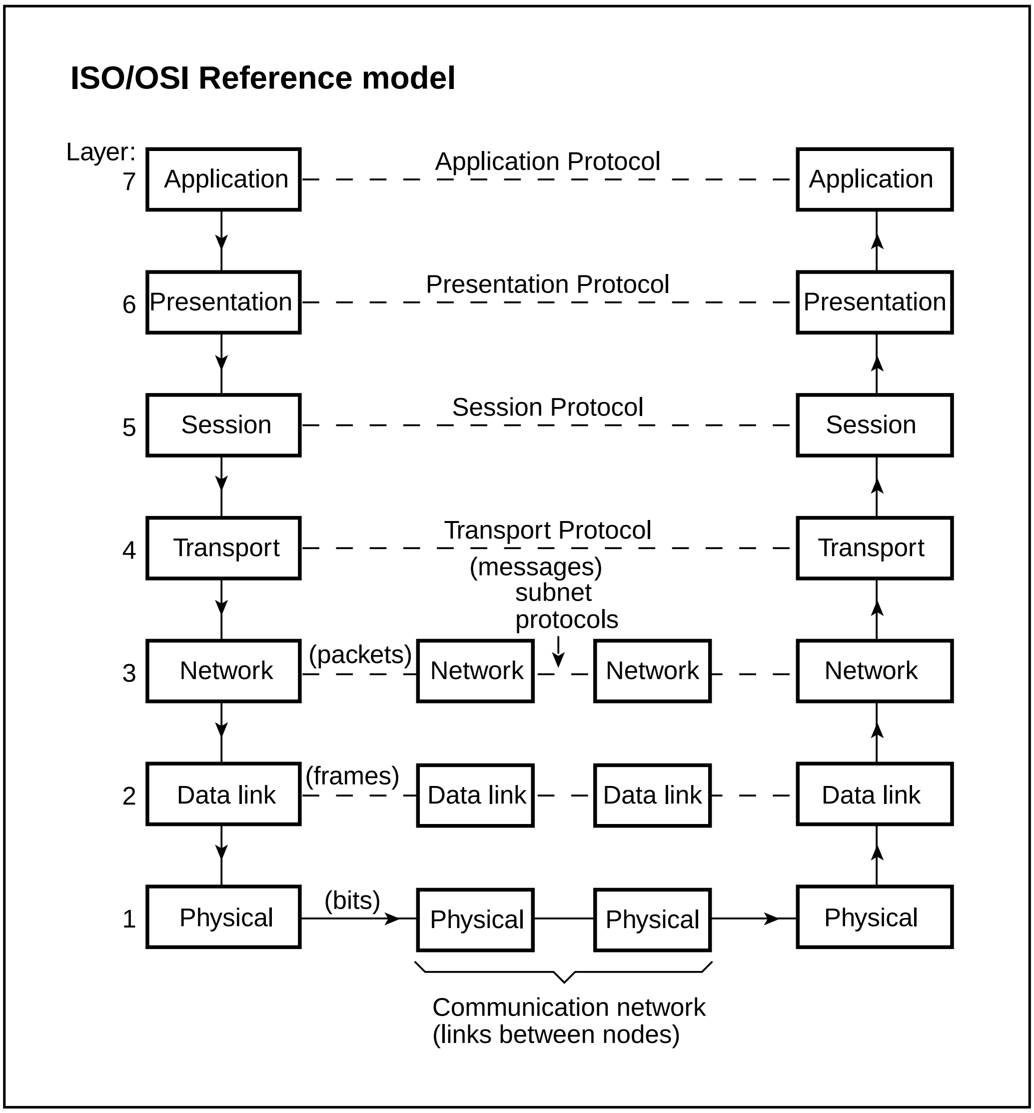


Figure 5.23

The main functions of the different layers is summarized in figure 5.24.

Application layer
e-mail, file transfer, remote login, www
Future: telephony, TV, video

Presentation layer
standard protocols to handle applications
data compression, encryption
format conversion

Session layer
establish sessions: session protocols, authentication
recover from break-down in transport layer

Transport layer
segmentation of large messages, i.e. to datagrams
establish and close down of end - to - end connections
end - to - end error control (virtual circuit performance)
flow control to avoid congestion

Network layer
fragmentation of datagrams to packets and the reverse
routing and flow-control within network
handling of billing information

Data link layer
handling of transmission errors: backward E.C.
acknowledgements
placement of packets in frames
transmission rate control (to avoid buffer overflow)

Physical layer
modulation, detection, synchronization
multiplexing FEC
physical links, incl. line equipment

Figure 5.24

The lowest layer is the **physical layer,** i.e. the transmissin of data over a physical link between two nodes. Important parameters are modulation-detection, synchronization and multiplexing.

The **data link layer** handles primarily the ARQ function for the connection between two network nodes. Buffering of the information is therefore necessary (logical channels).

The **network layer** split up messages coming from the next higher layer in suitable packets for transmission over the network. It handles routing of the packets in the network and access or congestion control to avoid overload of the transmissions links and the buffers in the node computers (packet switches or routers).

The **transport layer** handles the end-to-end transmission of messages (source computer to destination computer). Too long messages from the next higher layer are split up into shorter messages, i.e. datagrams. It also combines information from the differerent terminals, which are attached to the host or router via different ports. If a virtual circuit shall be set up, end-to-end acknowledgement and retransmission arrangements are included to make certain that no packets are lost or recombined in the wrong order.

The **session layer** establishes sessions, agrees with the session layer at the other end about session protocols. If the communication is broken temporarily, it reestablish the connection in an orderly way.

The **presentation layer** is a library of special procedures such as data compression, encryption, format transformations.

The different **application layers** are mostly defined by the user groups.

The communication process is based on contacts and dialogues between the corresponding layers on both side of the connection. These contacts are established by adding on a succession of headers to the packets, see figure 5.25. Also error control might be needed. This has been indicated in the figure by a trailer with parity bits for CRC (Cyclic Redundancy Check) in connection with the data link layer (the CRC or check sum could also have been included in the header).

Information flow in the ISO Reference Model (OSI)

(layers 1 to 5)

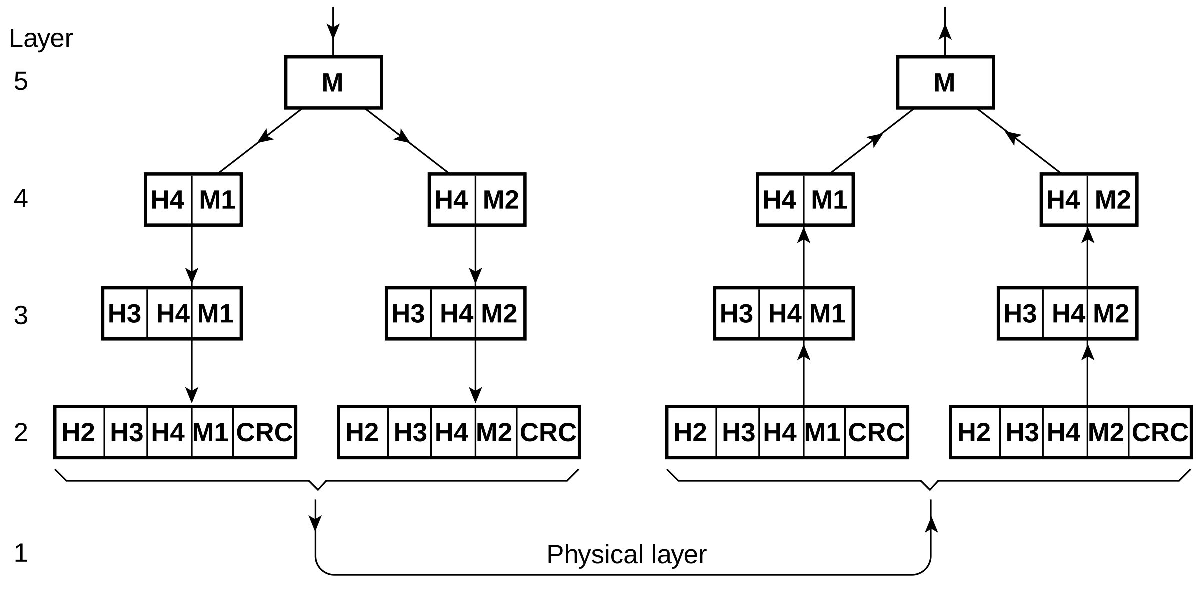


Figure 5.25

5.5.3 INTERNET

A much simplified network structure for INTERNET is shown in figure 5.26. The terminals are generally connected to local host computers via ports, but can also be directly connected to the IMP computer, which are their interface to the data network. (IMP – Interface Message Processor – is the ARPANET name for the routers or packet switches, which form the nodes of the network.)

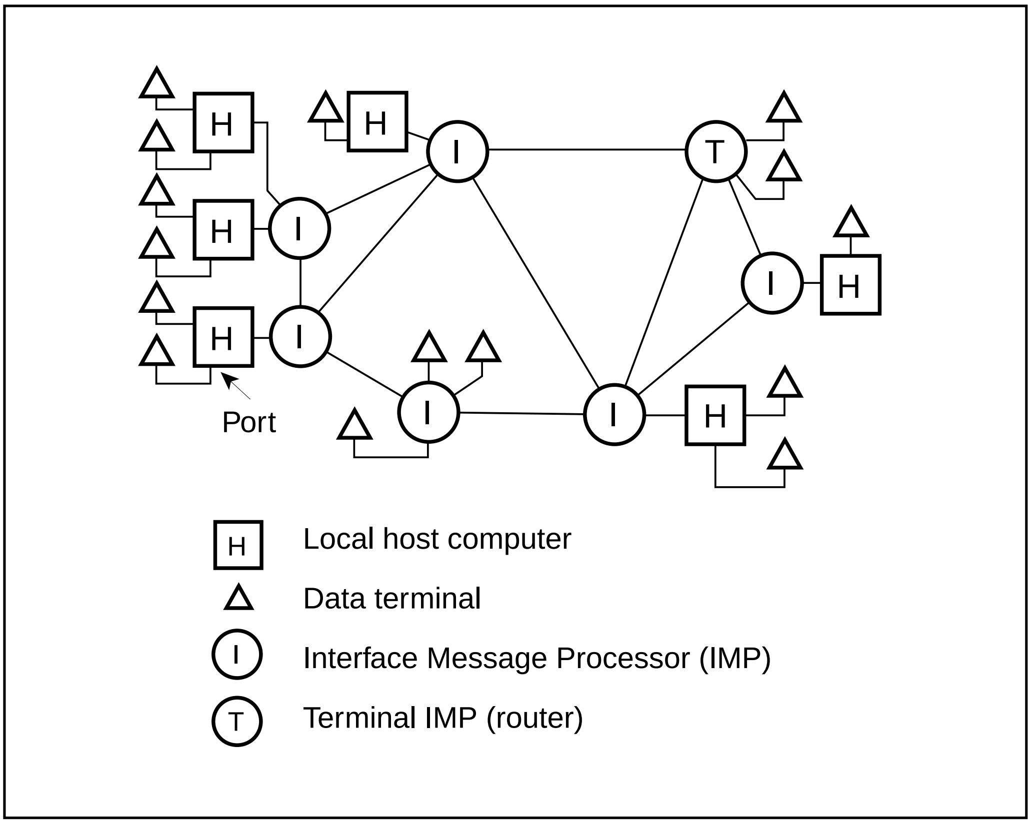


Figure 5.26

The layered protocol structure is shown in figure 5.27. There is no exact correspondence between this structure and the OSI reference model.

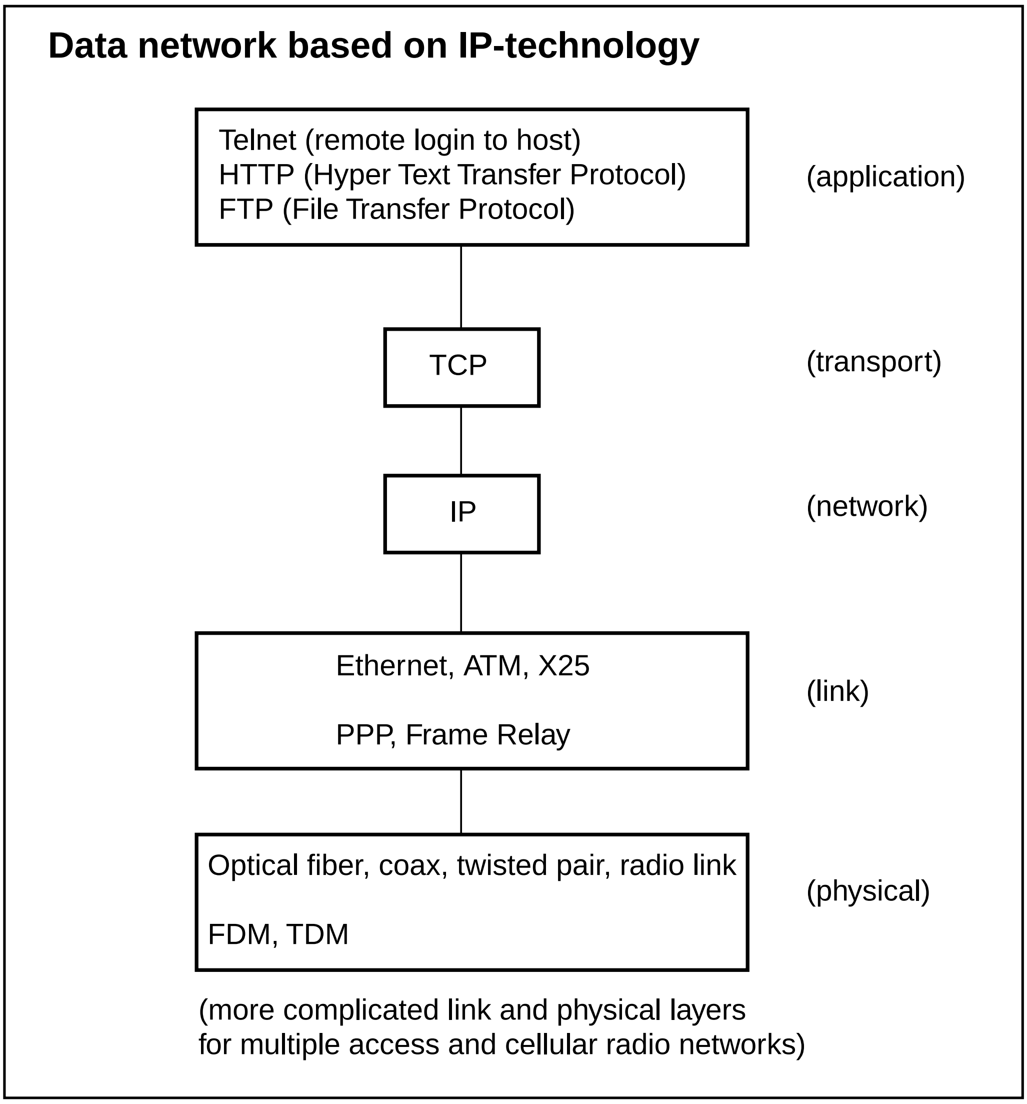


Figure 5.27

Several applications are handled by INTERNET, such as mail, remote login to hosts, file transfer, WWW. Future applications under study or trial are telephony, TV and video. However, TV and high-quality video require much higher bandwidth than what is generally available in the present INTERNET networks.

The key functions are **TCP (Transport Control Protocol) and IP (Internet Protocol)**. The corresponding TCP and IP layers exchange information through headers attached to the packets. Besides the traffic packets , different types of signalling and control packets are exchanged between the node computers, i.e. to support routing and congestion (access) control.

The TCP breaks up the messages into datagrams and reassembles them at the other end. It includes end-to-end control that all datagrams are transferred successfully. The IP is responsible for routing the individual datagrams (or packets) through the network.

The datagrams, that are generated during a session, are handled as completely separete entities by the network. They can be routed different ways through the network, which means that they may arrive in the wrong order to the other end. However, the TCP protocol feeds out the datagram to each user port in the right order and retransmit datagrams that have not been transferred successfully. That means that the connection for the next higher layer is exactly equivalent to a perfect virtual circuit connection with no lost packages.

The layer below the IP layer could be implemeted in many different ways. For the traditional fixed applications, ETHERNET or ATM can be used. If Ethernet is used, it would be necessary to add an additional header to the packets. In principle, the packets transmitted in the network would therefore have a structure according to figure 5.28a. The TCP and IP headers are quite complicated, see figure 5.28 b and c. The Ethernet header is shown in figure d.

In the original Arpanet, also the link layer protocal was included. A corresponding header was attached to the packets sent between the routers, see figure 5.28e. The maximun packet or frame length was about 1000 bits and the maximum length of a datagram was 8 packets, i.e. about 1000 bytes.

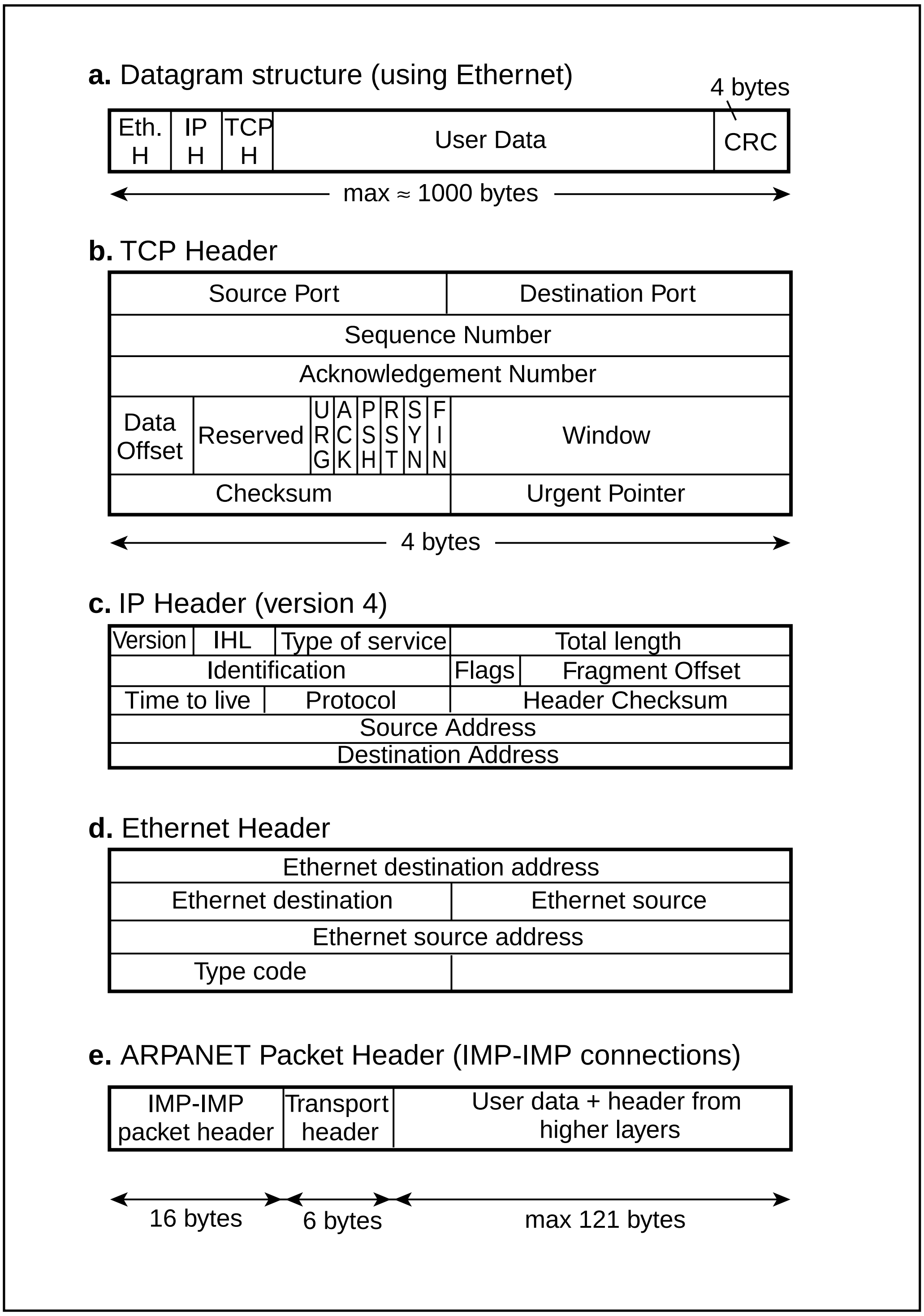


Figure 5.28

Further development of IP-technology goes on continuously. Presently, there are discussions about introduction of a new IP header (version 6). Version 6 will use adress fields of 128 bits instead of the 32 bits fields used presently. Also an Extension Header might be added, which could be used for autentication, encryption and in connection with mobile applications.

The adding on of several headers (and trailers) gives very large flexibility, but also a large overhead, which means increased data rate over the links. This is one of the problems, when INTERNET shall be extended to cellular radio networks with high demands on frequency economy. Also, the basic INTERNET is designed for a network, where the nodes are connected via fixed connection lines (single access) with extremely low errror rates. Changes and additions to the INTERNET protocols are therefore necessary to handle roaming terminals and transmission over cellular networks, with problems with fading, multiple access and handover. See figure 5.29.

Use of IP with cellular systems

Problems:

IP optimized for
- fixed terminals
- single access
- physical connections with high bandwidth and low b.e.r.
- lost packages only due to congestion in network

Cellular systems characterized by:
- roaming terminals, handover
- multiple access
- fading gives high b.e.r. with burts errors
- requirements on high spectrum utilization

INTERNET with cellular systems gives low throwput:
- slow action of TCP flow control
- inefficient backward E.C. at high b.e.r.
- long and several headers give high overhead

Need for changes:

Improved TCP protocol

Improved Link-layer protocol
- ARQ and FEC optimized for fading channels

Header compression (to improve frequency economy)

New protocol features
- to avoid package losses at handover
- to handle roaming terminals
- to handle multiple access (MAC: Medium Access Control)

Figur 5.29

One possibility to combine the fixed IP-structure with cellular systems is to use a separate mobile structure, see figure 5.30. In the wireless part of the network, functions are included to handle fading radio channels and multiple access (MAC: Media Access Control), and also to improve the frequency economy, see figure 5.30. In addition, mobility functions are needed to handle handover, roamers, registration of roamers at visited nodes and authentication. For the mobility function, it might be suitable to add an additional layer (with header), which translates the normal home address of a terminal (host + port) to a temporary mobile adress, with gives routing information to the visited node.

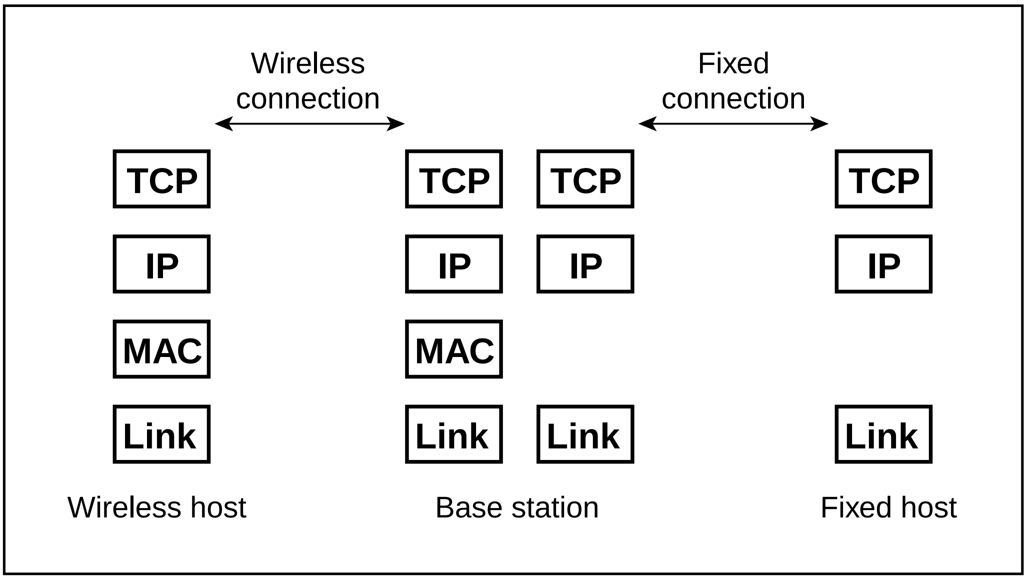


Figure 5.30

5.5.4 ATM

Structure

The basic ATM network is optimized for transmission in the fixed network. According to the OSI model, the main ATM function roughly corresponds to the data link layer. However, the main function of the data link layer – backward error control (ARQ) is not included. The reason is that the optical links, which are the typical physical transmission medium, give extremely low errror rate. Therefore, it is enough with end-to-end error control of a whole PDU (Protocol Data Unit) at a higher layer. (CRC is included in AAL 5, see below)

A drawback with this arrangement is that if a packet is lost in transit, it is generally nesessary to retransmit the whole PDU. A PDU often split up in many packages. Therefore, the capacity of the network could be seriously degraded if many packets are lost due to poor connections. To improve the situation, it is necessary to introduce ARQ (and possibly also FEC) in the data link layer. This is especially the case in cellular networks with fading radio connections.

Within ATM, the packets transmitted in the network are called cells. A basic ATM cell consists of an header of 5 bytes, followed by 48 information bytes. During time intervals with no traffic over a connection, idle cells are introduced, so that the connections between routers carry a continuous flow of symbols (synchronous trans-mission).

Above the ATM layer there is the AAL (ATM Adaption Layer), which controls how ATM is used to transmit data with different characteristics and quality requirements, see figure 5.31a. The signalling within the network is controlled by a separate function, called Control Plane. Four different AAL alternativs have been standardized. Two of these need to add additional control information to each cell, which means somewhat reduced payload per cell.

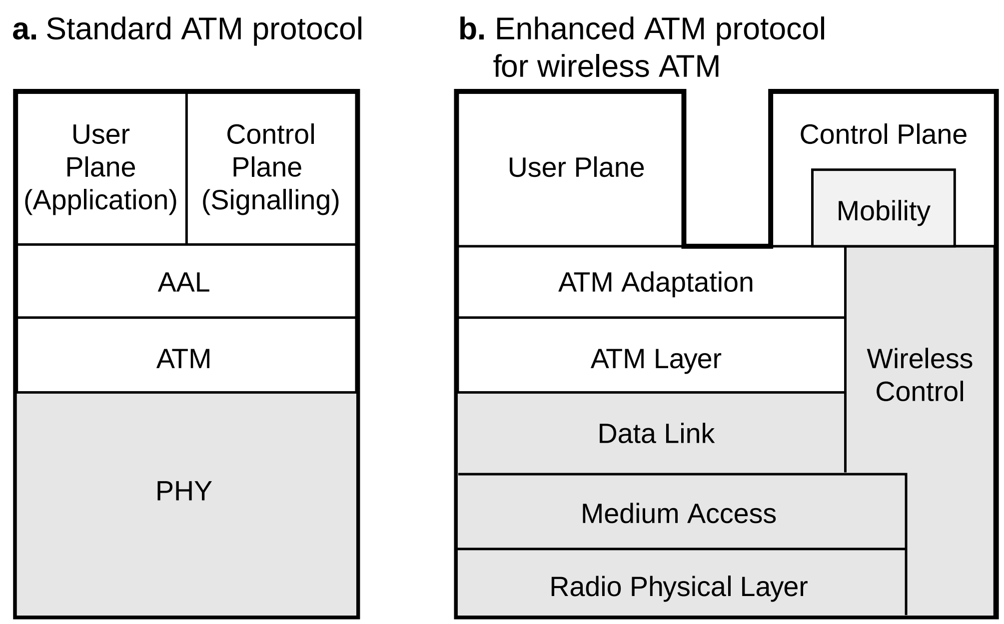


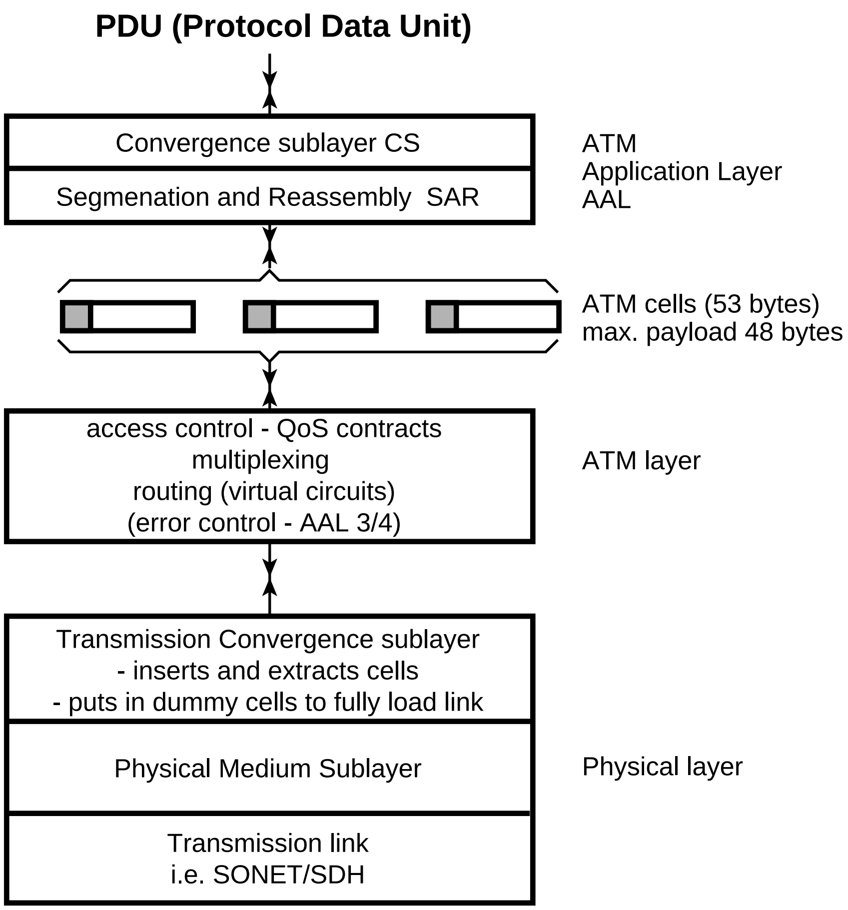
Figure 5.31

Protocol or Payload Data Units (PDU) are sent to the AAL layer from the application layer. Different applications generate information with different rate and statistical characteristics and have different requirements on the transmission network with respect to ber and transmission delay. See figure 5.32 and 5.33.

Application	Type of ATM service	Delay	Required b.e.r.	Bit rate
Voice / Audio	CBR	Bounded	medium	8-128 kb/s
Digital data	ABR/UBR	Unbounded	low/medium	0.1-10 Mb/s
Video telephony	CBR	Bounded	low	3.87 Mb/s
Motion video MPEG 1/2	CBR/VBR	Bounded	low	1.5-6 Mb/s
File transfer	ABR	Unbounded	low/medium	1-10 Mb/s

CBR and VBR combined with QoS
ABR only minimum data rate guaranteed
UBR on "best effort" basis with low priority

Figure 5.32



AAL 1	cell payload 47 bytes (one control byte for timing information) For CBR, real-time signals (voice, video)
AAL 3/4	cell payload 44 bytes (control bytes for multiplexing identification + CRC) For VBR, delay tolerant data traffic
AAL 5	"light weight" cell payload 48 bytes For VBR, delay tolerant data traffic requiring minimal sequencing or error detection support

Figure 5.33

Some applications cannot accept time jitter in the output signal. That means that delay variations, caused by the packet transmission over the network, must be eliminated.

The traffic characteristics of different applications are described with the following terms:

- Constant bit rate, CBR
- Variable bit rate, VBR (i.e. specified by peak and average rate)
- Unspecified bit rate, UBR (the user cannot give a statistical traffic description)

  Available bit rate, ABR (the bit rate is continuously adjusted by the source not to exceed what the network can cope with)

Examples of AAL protocols are AAL1 for CBR, optimized for 64 kb/s PCM speech (bytes generated 8000 times per second) and AAL5, optimized for transfer of large files and CBR. A new protocol, AAL2, has been standarized especially for the use of ATM in connection with cellular networks. This is discussed below.

Real-time applications often cannot accept varying transmission delays. That means that delay variations, caused by the packet transmission over the network, must be eliminated in an output buffer. Also the time stamps must be added to the cells. The maximum delay must be fairly low it it is a two-way connection. To give acceptable service to this type of applications, the ATM network must give a guarantee that the maximum delay will not exceed a certain value (bounded delay).

Access control, contracts about Quality of Service (QoS)

A drawback with the present IP-technology is that a certain QoS cannot be guaranteed. The offered performance is ”best effort”. The access control corresponds roughly to the ABR case above. Improvements are needed, if IP-technology shall be widely used for telephony, TV and video.

A key aspect of ATM is the concept of a **contract between a user or application and the network**. The user requests a connection with a certain traffic profile **(Source Traffic Description)** and **QoS.** The **Connection Admission Control** checks if the request can be accepted or must be rejected. If it is accepted, a contract is set up according to which the user agrees to adhere to the traffic description and the network agrees to give the requested QoS. If only small delays are acceptable, reservation of network resources must be made, but that limits the possibilities of sharing the resources between many bursty users (statistical multiplexing).

Another network function is the **Usage Parameter Control** which checks that the user does not violate the traffic decription. (If he does it is a risk that other users suffer due to congestion somewhere in the network). If violations occur and the network is heavily loaded, the bits exceeding the contract are directly discarded or packed in special cells, marked with low priority. At congestion, these cells are discarded in favour of cells with better priority.

The most important QoS requirements are the reliability (Cell Loss Ratio) and the transmission delay characteristics.

Real-time signals, such as two way speech and video phone, tolerate only small transmission delays and delay variations, but might tolerate fairly large cell loss ratio. As ARQ arrangements with retransmissions give large variations in transmission delays, this arrangement might be unsuitable. Another problem in connection with low data rate signals might be the time needed to fill a cell completely, before it is transmitted. This could give unacceptable transmission delay or result in only partly filled cells.

On the other hand, extremely high reliability is generally required for file transfers, but fairly large and varying transmission delays can be tolerated. ARQ arrangements are therefore suitable.

If it is impossible or unnecessary to define a certain QoS (i.e. UBR), the transmission is based ”best effort”, i.e. no guarantees on delay and lost cell ratio. The information is packed in low-priority cells.

5.5.5 Modified model for wireless access

Extension of ATM to Wireless ATM (W-ATM) is studied within study groups attached to ATM Forum and the ETSI BRAN (Broadband Radio Access Network) program. W-ATM would have similarities to UMTS/IMT-2000, but will offer user data rates up to 20 Mb/s.W-ATM will have similar characteristics to high-speed W-LAN, but the objective is that W-ATM shall also cover outdoor micro cells, serving portable terminals (limited mobility).

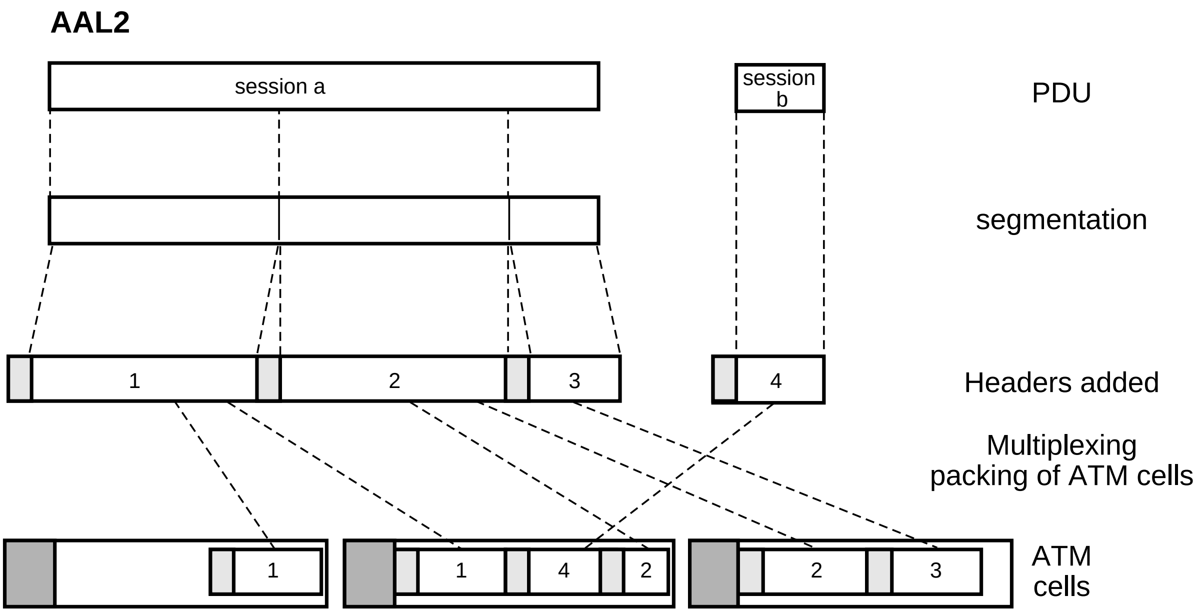
The ATM was developed for data transmission in the fixed network, which is characterized by single access and extremely reliable links. As dedicated links are used between nodes, unused capacity cannot be utilized by other purposes. Certain modifications are therefore necessary in connection with fading mobile links in cellular networks, which are based on multiple access and need to comply with high requirements on frequency economy :

a. Additional functions should be included in the ATM protocol, see figure 5.31b,
   An extra layer (MAC: Media Access Control) is needed to control the access to the common transmission media (transmission and reception of packets).
   An additional Data Link Layer with advanced error control
   A mobility control function for handover, location of roamers, registration at visited nodes and authentication

b. The frequency economy is improved through:
   A new AAL protocol (AAL2 "minicell" transmission protocol)
   Only active packets (cells) shall be transmitted over radio connections.

AAL2

The new AAL2 protocol gives increased flexibility in connection with PDU:s of different length, including variations during a session. Several short segments can be multiplexed into one ATM cell. The main advantage of AAL2 is that the cells can be fully loaded with user information. Also cells are generated only when there is user information to send. Se figure 5.34.



Advantages:

- Optimum packing of ATM cells
- Several sessions can be multiplexed on one ATM connection
- Low delays

Figure 5.34

The first usage of AAL2 is for the fixed links between the MSC and the base stations, allowing several times improvement in the effective link capacity through statistical multiplexing. ATM is used for these links as it permits the tight time synchronisation needed for soft handover in a DS-CDMA based system. AAL2 is also suitable to transmit speech channels in mobile systems with low rate coders, i.e. data blocks 50 times a second and reduced date rate during speech pauses for the transmission of comfort noise.

Radio transmission

At a normal ATM connection, continuous transmission is obtained by sending idle packets without user information. This is evidently not compatible with good frequency economy. Therefore, it is suggested to eliminate these idle cells before going over a radio connection. On the other side of the radio connection the right time relations are obtained again by putting in the eliminated idle cells. (Time marks are added to the transmitted cells.) See figure 5.35.

Also the packets are protected with better channel coding arrangements, combining error correction and detection, especially of the header. Also interleaving must be introduced to handle error bursts. The detailed specification is not yet determined.

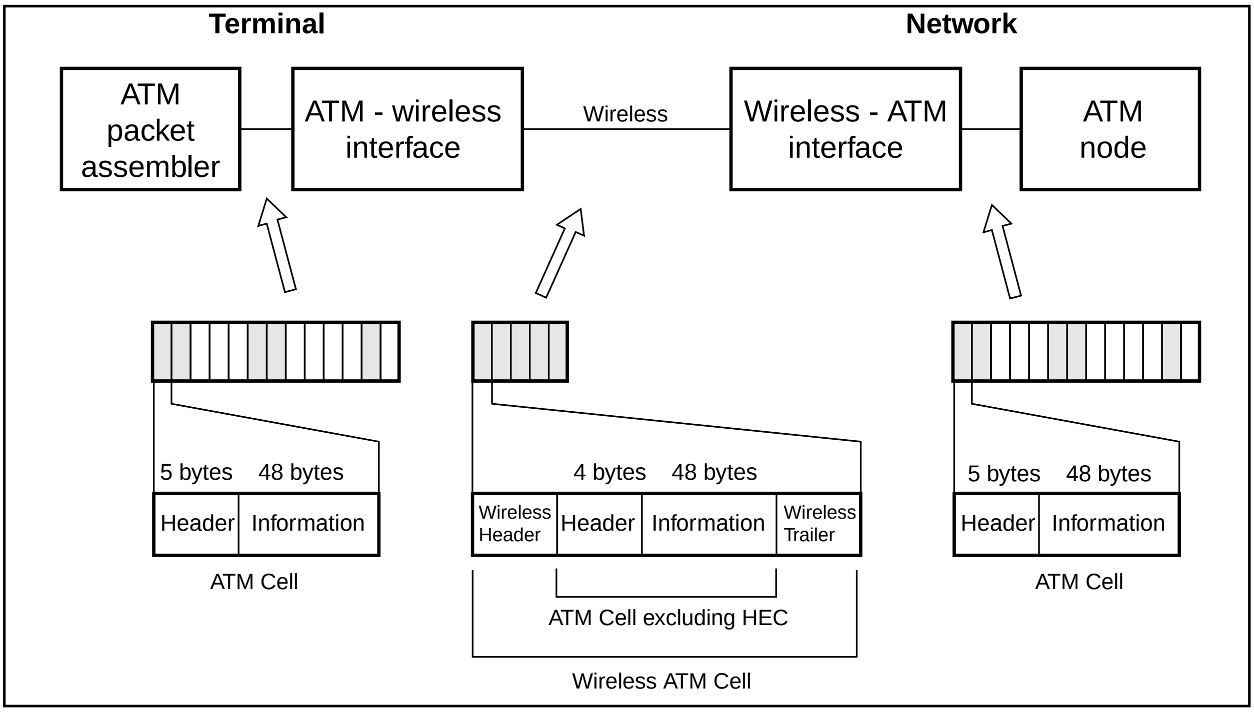


Figure 5.35

5.5.6 Adaption to different type of messages, MAC

Dependent on the characteristics of the data messages to be transmitted through the radio network and the required QoS, different packet transmission arrangements can be used.

Reservation procedures result in considerable signalling overhead and delay when used for short random messages. Therefore, a contention protocol is often preferred, i.e. slotted Aloha. For long messages, i.e. file transfers, reservation arrangements are suitable, as this gives maximum throughput.

For messages, which are given reservation, a further choice is if they shall be allocated a dedicated radio channel or share a channel with other users. An advantage with a dedicated channel is that it might be easier to dynamically adapt the channel capacity to varying user rates during the connection.

The optimum transmission arrangement also depends on the acceptable delay. If only very short delays are acceptable, the possibilities to apply ARQ are limited and only FEC might be possible (transparent mode). An example is packet transmission of speech. On the other hand, if considerable delay variations are acceptable, ARQ makes it possible to reach very low error rates.

Besides the random delays, the retransmission of blocks with errors also gives a reduction of the throughput.

In connection with handover, a new virtual circuit connection must be set up. Special arrangements must be introduced to avoid cell losses, which might temporarily reduce the troughput. Also the new virtual circuit might be suffer from more congestion than the previous one, which might make it impossible to keep the QoS agreement.

This means that the normal QoS levels at fixed ATM networks probably cannot be guarantied, when ATM is extended to cellular networks. This must be taken into account at the negotiation of the QoS between the user application and the ATM network. The users might have to accept a lower QoS during bad radio conditions and during handover.The quality degradaion might be higher transmission delays, lower transmission rate (i.e. ABR) or higher ber, giving worse speech or video quality.

6. Interference-limited systems (CDMA)

6.1 Characteristics of interference-limited systems

At an interference-limited system, many simultanious connections share intimately a common transmission channel. In principle, every connection shall utilize the full bandwidth for each transmitted symbol. See figure 6.1. This is a clear distinction to FDMA (each connection occupies one frequency slot) and TDMA (each connection occupies one time slot). Both FDMA and TDMA permit very high isolation between the connections (orthogonality). On the other hand, the number of available channels are limited by the width of the allocated band. FDMA and TDMA are channel-limited systems.

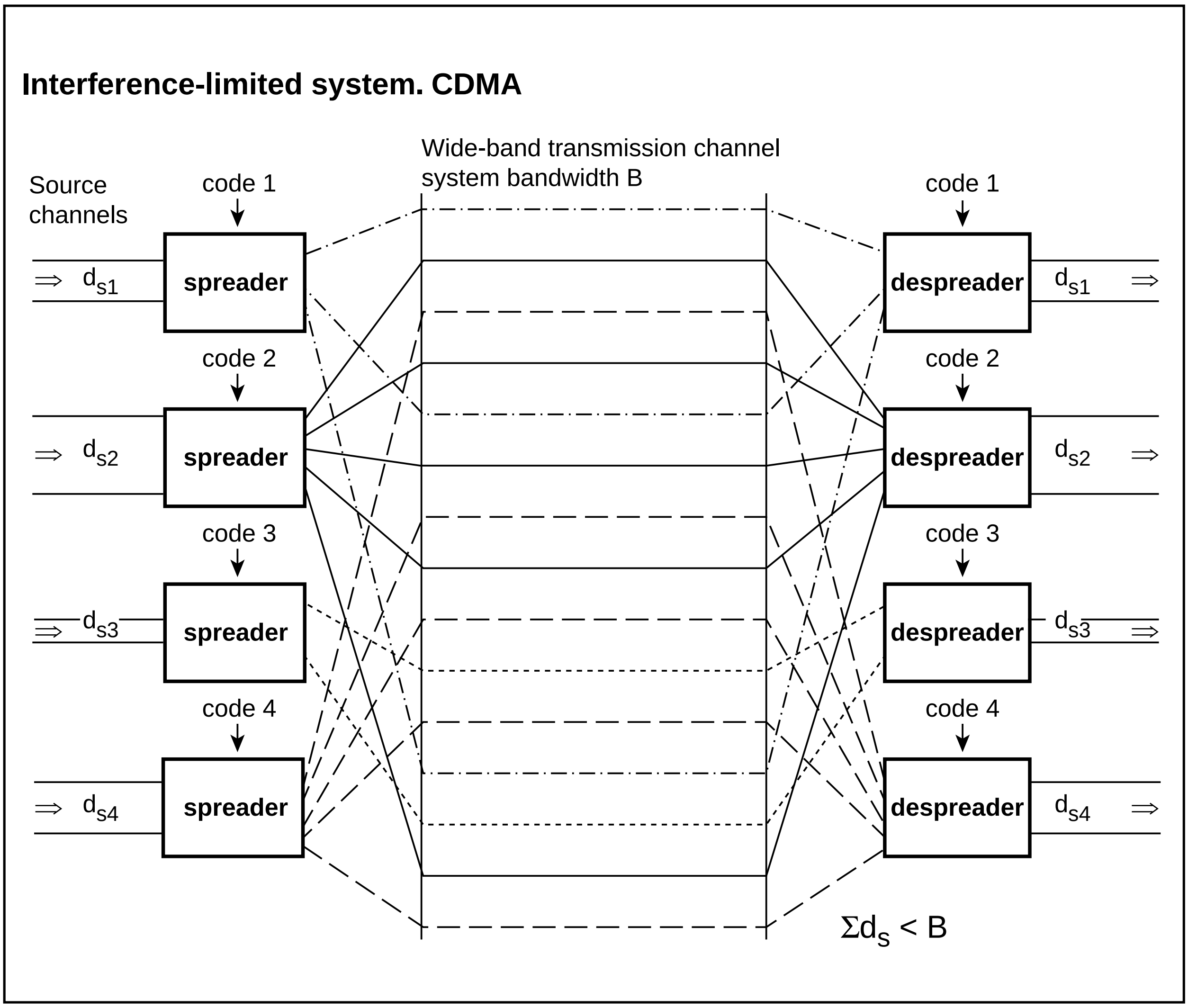


Figure 6.1

The method to isolate the connections from each other in an **interference-limited** system is to assign a specific code to each connection. Knowledge of this code permit the receivers to discriminate between the wanted signal C and the interference I from all the other signals. Therefore, a system of this type is usually called a CDMA-system (Code Division Multiple Access).

In the basic CDMA configuration, the possible discrimination between connections is limited. This means that when the total interference level is gradually increased, a point is reached when the C/I becomes too low and thus the transmission quality unsufficient. To describe this characteristic, the term **anti-jamming margin (AJ)** is introduced. The AJ is the maximum acceptable value of I/C for adequate transmission quality. Therefore, the system capacity is interference-limited. On the other hand, the number of available codes can be much larger – the capacity of the system is not limited by the number of available codes.

To get an adequate isolation between the connections, considerably larger system bandwidth would be necessary than for a channel-limited system with the corresponding capacity. Increased **bandwidth expansion** gives increased AJ. A related parameter is the **processing gain**, which is also related to the bandwidth expansion. The processing gain determines how much an optimum receiver can suppress the interference relative to the wanted signal. These parameters are discussed in detail in connection with DS-CDMA (section 6.4).

As the system capacity is limited by the total amount of interference (received signals from all the other simultanious connections), all means that reduce the level of interference will directly improve the total system capacity. One example is that the transmitters should be active only when there is useful information to send. On the other hand, it is very important with accurate power regulation so that all signals have the same level at the receiver input. Otherwise high level interference will drastically reduce the system capacity (near-far problem).

Advantages of interference-limited systems

It follows from the general discussion above that a perfect interference-limited system has the following characteristics:

a. Highly effective **dynamic allocation of channel capacity**, which accurately follows the varying needs of the different connections (bandwidth on demand). As all connections fully use the available channel bandwidth, the spreading ratio and thus the processing gain are larger for connections with low source data rates. This allows the corresponding decrease of transmitter power, and thus reduced contribution to the total interference level. This can be considered as a further development of the DSI and DTX arrangements, discussed above.

b**. Soft capacity limit** (graceful degradation). During extreme traffic peaks, a modest increase of the total interference level might be acceptable. This will increase the peak capacity, if it is acceptable that the transmission quality of all connections are reduced during short overload intervals.

c. The required cluster size (given by the normalized reuse distance) is determined by the **average of the cochannel interference**, instead of the worst case as for systems based on narrow band multiple access such as FDMA and TDMA (without frequency hopping). However, a prerequisite is that a very large number of connections share the same radio channel, so that the statistical variations of the total interference level around the average value become small.

d. In order to get maximum gain from **frequency diversity** the utilized bandwidth must be considerably larger than the coherence bandwidth of the propagation channel. As all connections share a wide common frequency band, the possibility for diversity gain is improved.

e. The combination of c and d, together with sufficient processing gain through band spreading, makes it possible to reduce the effective interference level so much that **cluster size one** can be utilized, i.e. the same wideband radio channel can be used by all cells.

This eliminates the need for frequency-cell planning, which otherwise would become extremely complicated for systems using micro and pico cells. (Another possibility to eliminate the tedious frequency-cell planning and on the same time improve the frequency economy is to use C/I-controlled dynamic channel allocation of the type which is used at DECT).

Need for fast dynamic power control and soft handover

Power control to minimize the interference level at the base station receiver is only effective for terminals assigned to this base station. A remaining problem is therefore interference from terminals, whose output power is controlled by nearby cells.

To minimize this interference, it is necessary to use soft handover and careful selection of which base stations shall be involved in the macro diversity. Instead of the problem with cell-frequency planning, it is a need for accurately planning and controlling the power levels within the network. This is discussed in connection with the detailed discussion of DS-CDMA in section 6.4.

6.2 Overview of DS-CDMA, FH-CDMA and OFDM

DS-CDMA

The name Direct Sequence CDMA refers to the fact that a code sequence is directly used to mark the transmitted symbols so that bandwidth expansion is obtained. (Another alternative is Frequency-Hop CDMA (FH-CDMA) in which case the code sequence is fed to a series-parallel converter, which determines the frequency slot, to be used for the next hop interval.)

The key feature of DS-CDMA is that each transmitted radio symbol is coded with a random sequence or code. Each information bit is thus represented by a sequence of ”chips”. Simultanious radio connections in the same geographic area use different code sequences. The receiver must use the same code as the corresponding transmitter. The name Code Division Multiple Access refers to the fact that the processing gain through the coding gives enough isolation between simultanious connections to make multiple access possible.

The coding of each information bit with a chip sequence gives a considerable bandwidth expansion, as the chip time is much shorter than the time per source bit. (The modulation bandwidth is roughly the inverse of the length of the source bit and the chip bit respectively). Another name for a system of this typ is therefore a **band spreading system**. See figure 6.2. A simplification in the figure is that antipodal signals are used. The relations between the source and chip data rates and the corresponding bandwidths are somewhat different, if high level modulation (i.e. 4 QAM or 16 QAM) is used. It is the ratio of the bandwidths after and before the spreading that gives the processing gain.

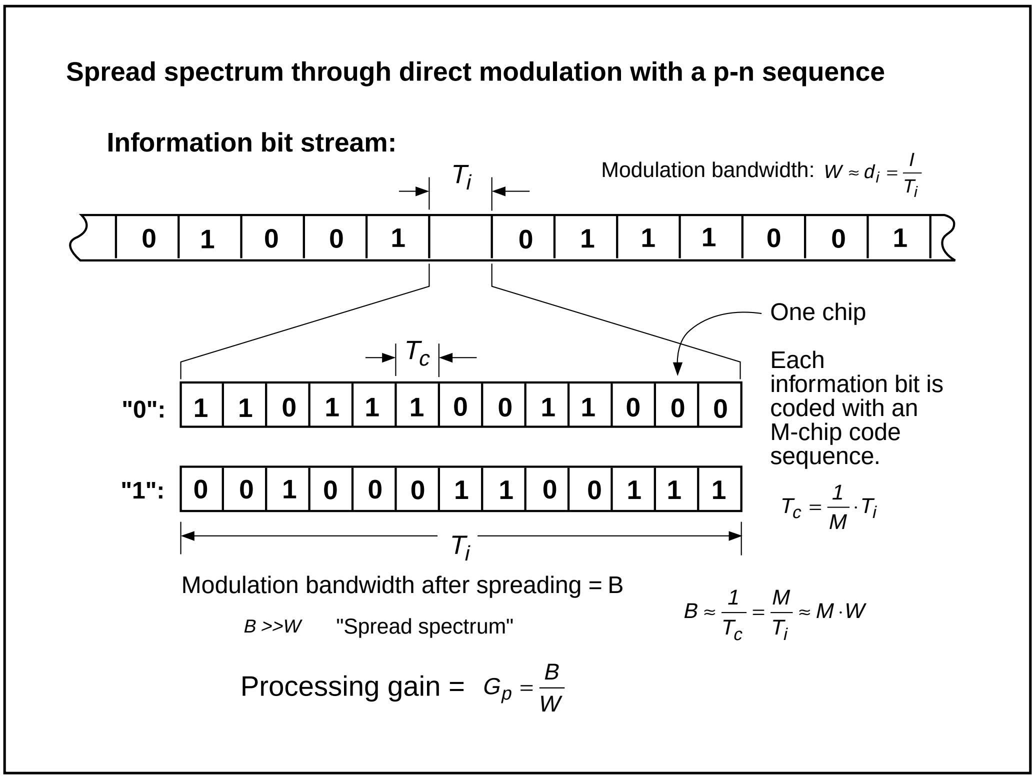


Figure 6.2

With increased number of chips per bit, the processing gain is increased, but at the expense of larger modulation bandwidth. There is a direct relation between the bandwidth expansion (spreading ratio) and the processing gain. An increase in the processing gain gives the corresponding increase in the anti-jamming ratio and thus a higher number of simultanious connections can be accepted.

FH-CDMA

At FH-CDMA, the bandspreading and CDMA function is obtained through random frequency hopping. If the basic system is TDMA, the carrier frequency is usually changed for each frame. See figure 6.3. This is called slow FH-CDMA, and is the FH-CDMA arrangement mainly proposed for cellular systems. As shown in the figure, another advantage of frequency hopping is the possibility to obtain frequency diversity, if the system bandwidth exceeds the correlation bandwidth. GSM with the frequency hopping option (FH-TDMA) can be operated in a FH-CDMA mode.

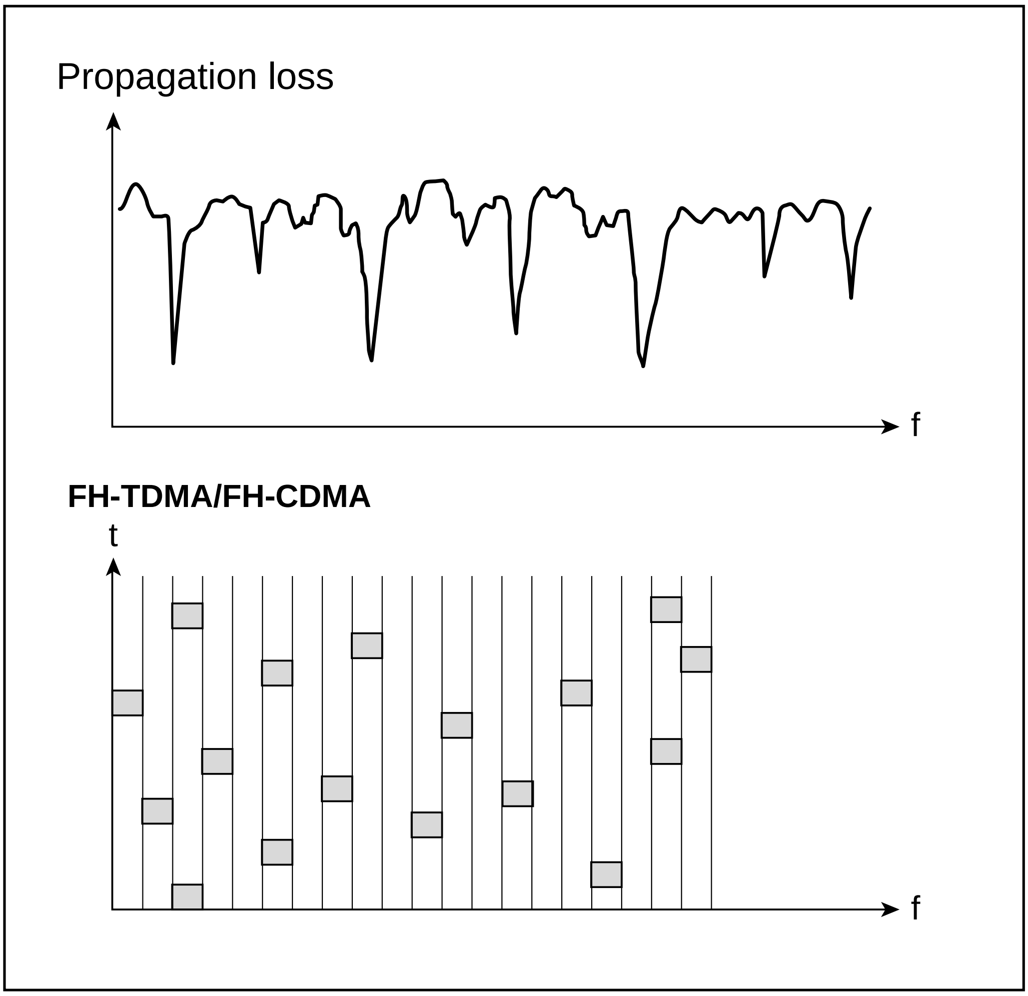


Figure 6.3

As mentioned above, the full processing gain is only obtained if each transmitted symbol is spread out over the system bandwidth. This does not seem to be the case, as the radio signal occupies a narrowband frequency slot during each hop interval, during which several symbols are sent. FEC in combination with interleaving must be added to obtain processing gain (and frequency diversity). This is discussed in section 6.5.

OFDM

The basic OFDM (Orthogonal Frequency Division Multiplex) divides up the available system bandwidth in a very large number of narrow frequency slots. Each connection can be allocated several slots, more or less randomly distributed over the system bandwidth. A user, needing large transmission capacity, can be given a large number of slots. See figure 6.4. As with FH-CDMA, FEC combined with interleaving (over frequency slots) is necessary in order to get processing gain and frequency diversity. (To obtain frequency diversity it is also necessary that the system bandwidth exceeds the correlation bandwidth of the propagation channel.)

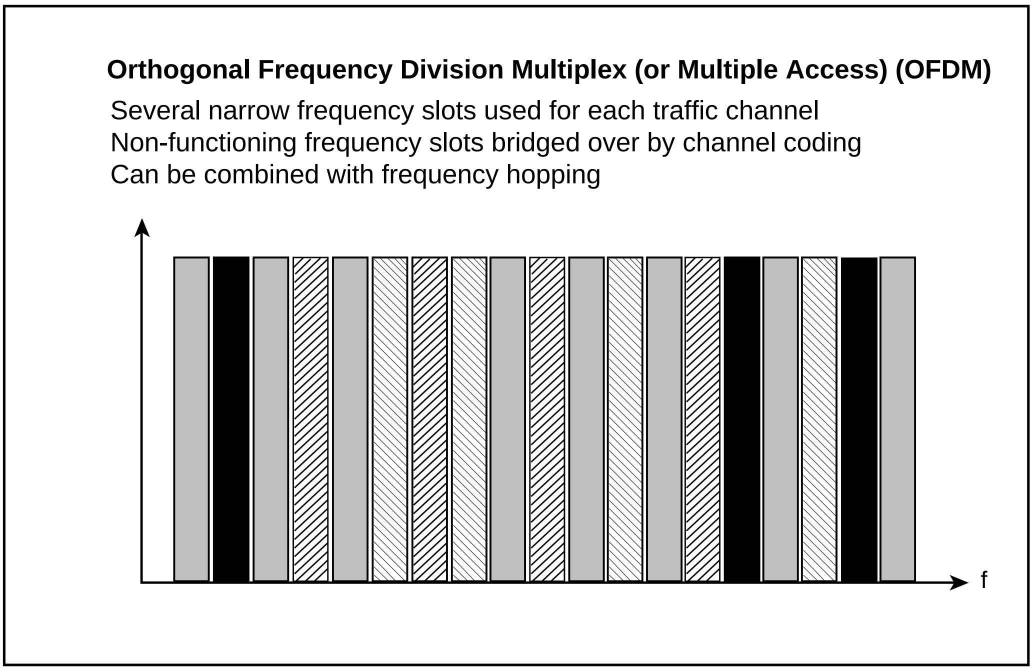


Figure 6.4

This basic system can be transferred to an interference-limited system by introducing a time slot structure and randomly assign a new set of frequency slots to each user for each time slot. The selection of frequency slots for each time slot will be determined by a random code, specific for each user. OFDM is further discussed in section 6.6.

6.3 Orthogonal and quasi-orthogonal connections

An important concept in relation to different types of Multiple Access is the mutual orthogonality between the connections. The ideal situation would be that the connections are completely isolated from each other, thus completely avoiding mutual interference. Examples of orthogonal MA arrangements are FDMA (if wide enough frequency guard slots are used to suppress adjacent channel interference) and TDMA (if wide enough time guard slots are introduced, to eliminate time overlap between data bursts in adjacent time slots).

Another example, taken from GSM, is cyclic, coordinated frequency hopping used for the connections belonging to each base station. In this way, collisions between the connections within each cell are avoided. The hopping arrangement retains both the full traffic capacity (no bandwidth expansion) and the full orthogonality between the connections.

It is also possible to make connections orthogonal by taking advantage of the mutual orthogonality of Walsh functions (see appendix 1). However, there is an important limitation. Orthogonality is only obtained with perfect mutual time syncronization between the Walsh functions. This syncronization is simple to implement, if the orthogonal channels are generated by the same transmitter (i.e. a single-access situation). On the other hand it is hardly possible to generate orthogonal channels at the input of the base receiver, if the signals come from terminals with different propagation delays to the base. Even in the outward direction, the isolation is decreased if the time dispersion gives deviations from the optimum timing at the terminal receivers.

The maximum number of orthogonal connections or traffic channels, which can be contained within the allocated frequency band for the system, is determined by the required bandwidth per traffic channel. At a traditional cellular system, based on FDMA or TDMA, a certain number of channels are available per system. These channels must be distributed between the cells within a cell cluster. A compromise must be made between a high number of traffic channels per cell and too low transmission quality due to cochannel interference, if the cluster size is unsufficient.

This limitation of the number of available connections per cell can be avoided by accepting that the connections are not perfectly orthogonal - they are only **quasi-orthogonal**. As has been discussed above, this leads to the concept of an interference-limited system. Exampels of MA arrangements with quasi-orthogonal connections are the basic DS-CDMA with random spreading codes and FH-CDMA with random hopping patterns, which result in collisions.

System alternatives with improved isolation

An important consideration in connection with interference-limited systems is if some of the advantages of orthogonality can be retained.

A possible arrangement might be to have the connections within each cell mutually orthogonal, but to accept that the connections, belonging to different cells, are quasi-orthogonal. This can be achieved with **FH-CDMA** by using coordinated, cyclic frequency hopping within each cell, but randomness between the frequency hop structures of connections belonging to different cells (this arrangement is used at GSM).

At **DS-CDMA**, the utilization of Walsh-functions (at channels with relatively little time dispersion) can give full orthogonality between outgoing connections within the same base station. This is used at **IS-95**. However Walsh functions are not used at IS-95 to improve the orthogonality in the inward directions, as the necessary time syncronization cannot be achieved.

Recent developments of DS-CDMA make it possible to increase the mutal isolation betweem connections by **interference cancellation**. The concept is included in the two system alternatives for UMTS. The simplest (but still extremely complicated) arrangement is that the receiver first detects the stongest interference and substract this contribution from the total interference. In the next step, the next largest interferer is handled in the same way - and so on. A necessary condition for this procedure is that the receiver knows the spreading codes of the interferers to be suppressed. A more advanced concept for interference suppression is **joint detection** of all signals.

6.4 DS-CDMA

The principle of DS-CDMA has been described above in connection with figure 6.2. This section gives a general coverage of the main characteristics of DS-CDMA. A specific system, the US standard IS-95, is discussed in appendix 1.

Processing gain, anti-jamming margin

The discrimination between the wanted signal, marked with the right sequence, and the interfering signals, marked with other code sequences, depends on the operation of the receiver mixer. See figure 6.5. If the LO-signal is modulated with the same code sequence as the received wanted signal inclusive the same timing, the receiver mixer performs the invers operation to the transmitter mixer (**despreading**). This means that the bandwidth of the original input signal to the transmitter mixer is restored.

Therefore, the wanted signal from the mixer can pass the receiver filter without any attenuation. The filter is matched to the original narrowband radio signal, which was fed to the transmitter mixer. This means that the noise-limited sensitivity, corresponding to $(C/N)_{min}$ or $(E_s/N_0)_{min}$, is the same as for the basic transmission arrangement without band spreading (if the interference is neglected). In a noise-limited situation, no processing gain is obtained.

Other signals, which either have been coded with other chip sequences or have the wrong timing of the code sequence, have at least a bandwidth corresponding to spread bandwidth B after having passed the receiver mixer. Only a minor part of their spectra can therefore pass the matched receiver filter with bandwidth $W \approx d_s$. The suppression of these other signals, i.e. the processing gain, is given by the **spreading factor** $B/W = d_{ch}/d_s$ i.e. the ratio between the signal bandwidths before and after the the transmitter mixer. (Antipodal waveforms are assumed).

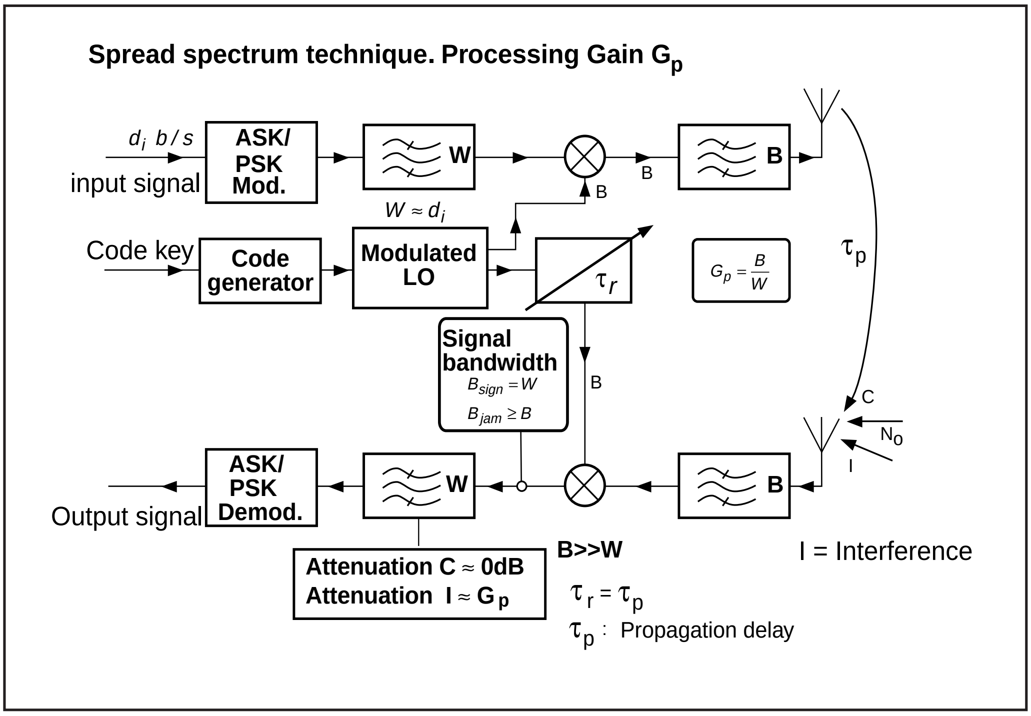


Figure 6.5

If a large spreading factor is used, all wideband signals from the receiver mixer (noise and interference) have roughly the same structure as gaussian noise. The required C/I into the detector for a certain bit error rate is nearly the same as the corresponding E_s/N_o, if the interference density ($I_o = I/B$) is much larger than the noise density N_o at the receiver input. This means that the minimum acceptable value of C/I into the receiver is:

$$(C/I)_{min} = (E_s/N_0)_{min} - G_p \text{ dB},$$

where G_p is the processing gain.

This value of C/I must be negative to make CDMA possible. Therefore, the concept anti jamming margin is generally used instead:

$$AJ = -(C/I)_{min} = G_p - (E_s/N_0)_{min} \text{ dB}.$$

Traffic capacity

In the ideal case with all signals into the receiver (wanted signal C and interfering signals I) having exactly the same level, the AJ value determines how many signals that can share the common frequency band. See figure 6.6. (An additional assumption is that the interference level after the matched filter is much higher than the thermal noise.)

As example, for AJ = 13 dB, i.e. 20 times we can estimate the maximum number (N) of simultanious connections. The number of interfering signals is (N-1). As C = I, the total interference becomes $I_{total} = (N-1)I = (N-1)C$. Therefore:

$$AJ = I_{total}/C = (N-1) = 20 \Rightarrow N = 21.$$

Increased band spreading gives increased process gain and thus increased AJ value, i.e. an increased number of simultaneous connections.

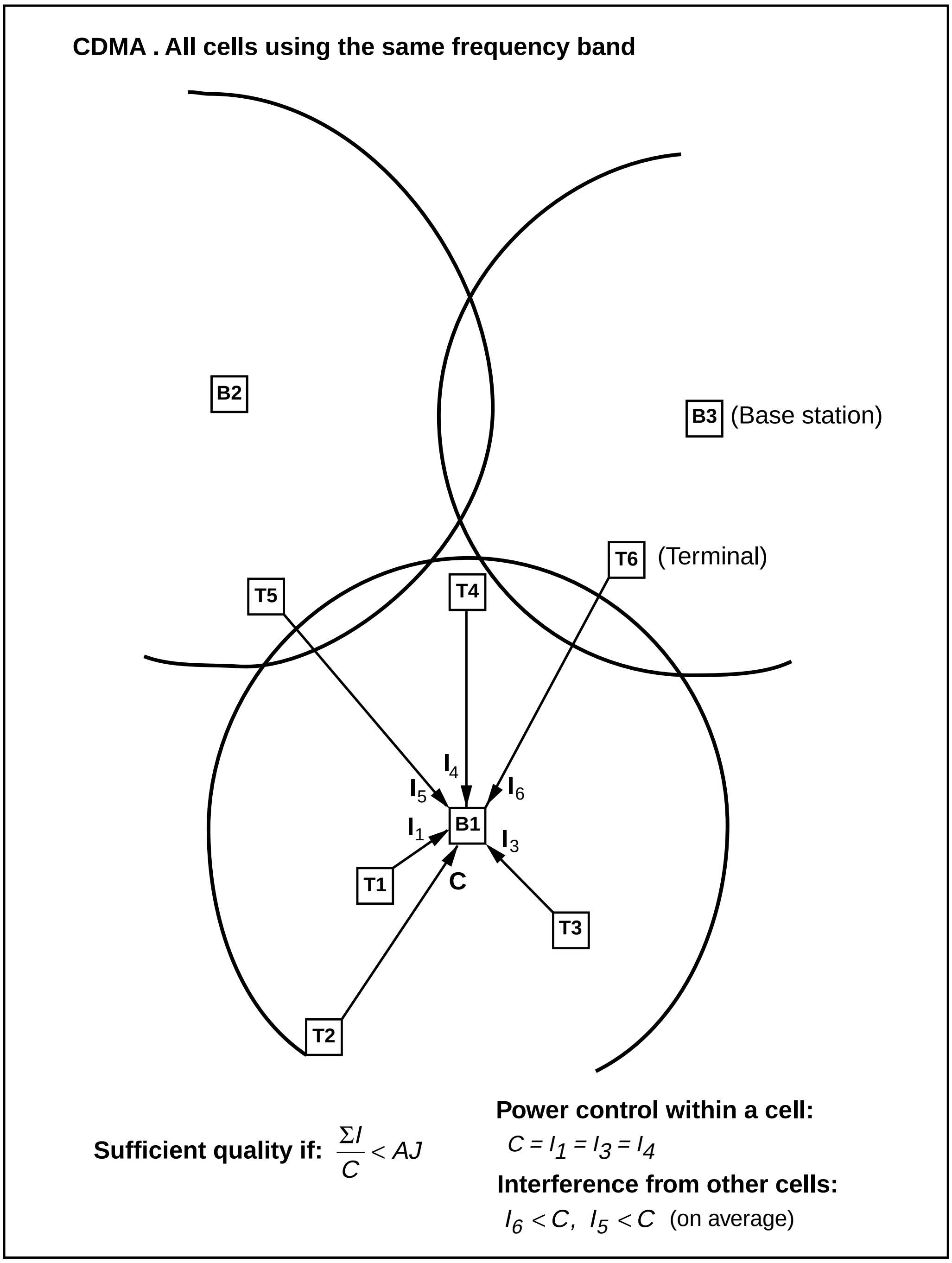


Figure 6.6

Degradation factors

If the level of the signals into the receiver are different, the stronger interfering signals will take up more than their share of the totally allowed interference. This results in a considerable reduction of the possible N value. One of the largest practical problems in connection with DS-CDMA is the need to continuously adjust the incoming signals to the base station receiver to their optimum level. Especially, fast control of the transmitter output power is necessary to compensate for the variations in propagation loss due to multipath fading.

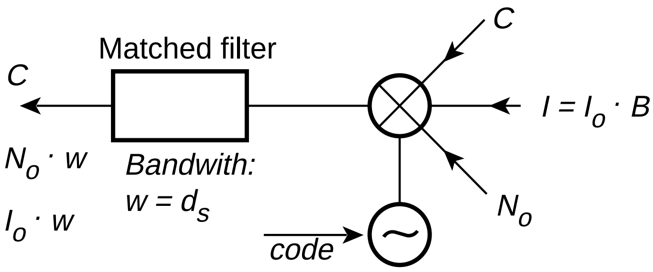
However, it is impossible to accomplish completely optimum level settings. Therefore, the practical multipel access capacity becomes somewhat lower than the ideal value corresponding to perfect power regulation. An example of how the power control can be accomplished can be found in appendix 1 (IS-95).

The traffic capacity per cell is further reduced due to interference from adjacent cells, if the same frequency band is used for all cells (cluster size one). This is the typical case when DS-CDMA is used. The interfering signals, coming from terminals registered at a nearby base station have in average a higher propagation loss, but on the other hand the power control to compensate for fading is controlled by their own base station. The fading variations are roughly uncorrelated over the propagation paths to the two base stations. The interference from terminals in other cells gives an additional reduction of about 30% of the traffic capacity, if close to optimum procedures are used for the soft handover.

The discussion above about the anti-jamming margin is somewhat simplified. We have assumed that the cochannel interference into the base station receiver is the only degradation factor that determines the detector characteristic, i.e. the normal thermal noise N is neglected. This is not quite realistic, as a very high value of Eb/No would be needed to obtain a sufficiently low No-value. This means seriously reduced sensitivity, i.e. a reduction of the noise-limited range. Also instability effects appear in the power control, when the system gets close to 100 % interference load. It seems that the interference load should not exceed 80% of the theoretical figure given by the AJ.

The case that both interference and receiver noise degrade the receiver performance is analyzed in figure 6.7.

Receiver performance degraded both by N_O and I_O



a. Normal, noise-limited case *(I = o)*
Required E_S determined by $(E_S/N_O)_{min}$

b. Both N_O and I at receiver input
Replace N_O by $(I_O + N_O)$ in case a

$$(E_s/N_o)_{min} = \begin{cases} E_s/N_o \cdot \dfrac{1}{1+\dfrac{I_o}{N_o}} & \text{Case 1. Noise limited sensitivity} \\[2ex] E_s/I_o \cdot \dfrac{1}{1+\dfrac{N_o}{I_o}} & \text{Case 2. AJ} \end{cases}$$

Case 1: $E_s/N_o = (E_s/N_o)_{min} \cdot \left(1+\dfrac{I_o}{N_o}\right) \Rightarrow E_s/N_o = (E_s/N_o)_{min} + \left(1+\dfrac{I_o}{N_o}\right) dB$

Case 2: $E_s/I_o = \dfrac{c/d_s}{I/B} = \dfrac{C}{I} \cdot G_p = (E_s/N_o)_{min} \cdot \left(1+\dfrac{N_o}{I_o}\right)$

$$AJ = \frac{I}{C} = G_p - (E_s/N_o)_{min} - \left(1+\frac{N_o}{I_o}\right) dB$$

Example 1: $N_o = I_o \Rightarrow \left(1+\dfrac{N_o}{I_o}\right) = \left(1+\dfrac{I_o}{N_o}\right) = 3dB$

Noise-limited sensitivity and AJ degraded 3 dB

Example 2: $I_o = 4N_o \Rightarrow \left(1+\dfrac{N_o}{I_o}\right) = 1dB \qquad \left(1+\dfrac{I_o}{N_o}\right) = 7dB$

AJ degraded 1 dB, noise-limited sensitivity degraded 7 dB

Figure 6.7

Bandwidth-on-demand, frequency diversity

The discussion about the system capacity above has assumed that all connections use the same source data rate. The situation is a little more complicated, if the source data rate d_s varies. Source signals with different bandwidth (data rate) are through the band spreading transformed to radio signals with a modulation bandwidth $B \approx d_{ch}$. This means that the spreading factor d_{ch}/d_s follows the variations in the source data rate ds.

The corresponding variation applies to the processing gain which is also d_{ch}/d_s , i.e. a narrow band source signal has a larger processing gain than a wide band signal. For a certain specified bit error rate and total interference level at the base receiver input, the necessary input level to the base receiver is therefore proportional to the bandwidth of the source signal. If the differences in propagation attenuation for signals from different terminals are disregarded, the optimum transmit power (normalized transmit power) is also directly proportional to the source data rate. See Fig. 6.8. The actual transmit power P_t deviates from the normalized transmit power P_t due to the need to also compensate for the variations in the propagation loss, (L_p).

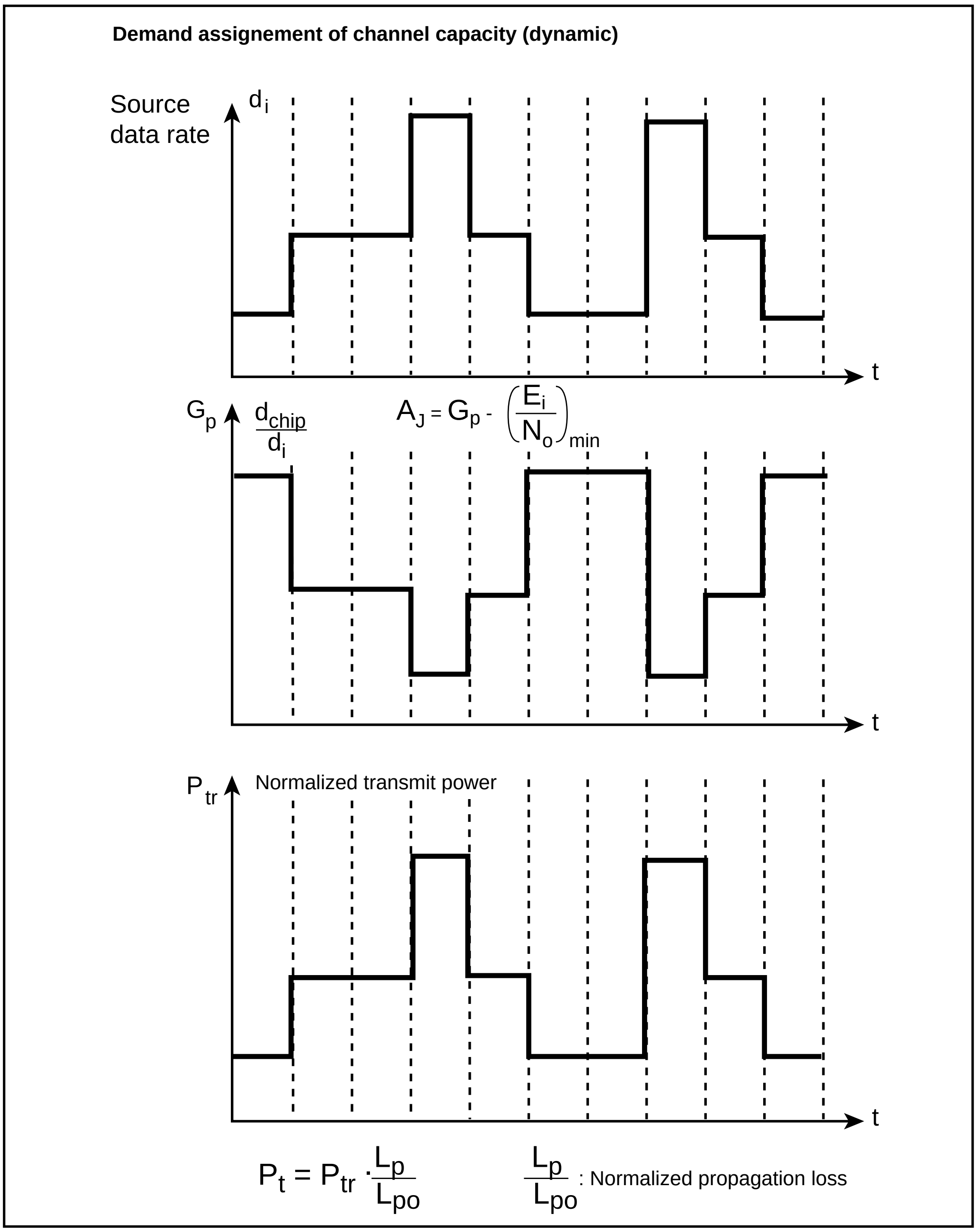


Figure 6.8

Due to the relation between source data rate and optimum transmit power, discussed above, (and the very efficient averaging of cochannel interference) the traffic capacity of the system is improved by DTX or adaption of the transmit power to the data rate of the source signals and by control of the transmit power to compensate for the variations in propagation attenuation.

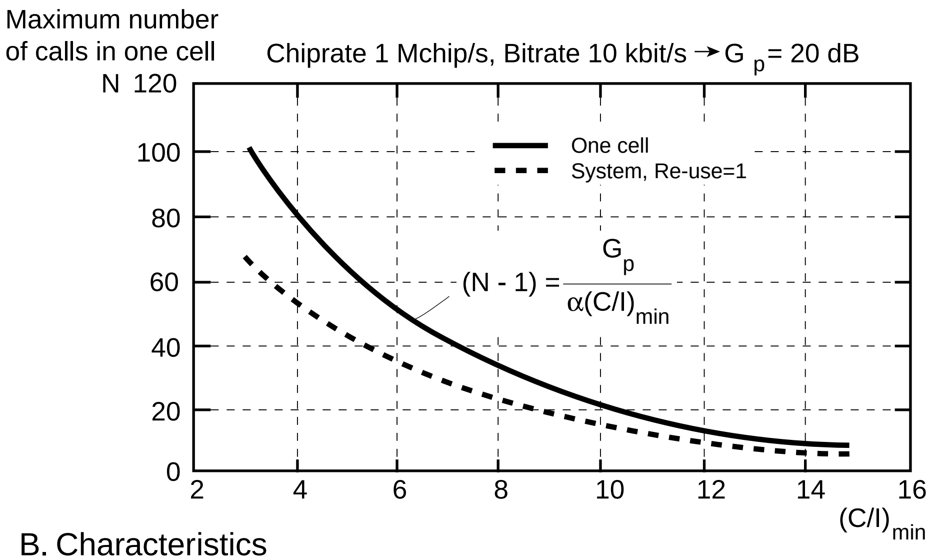
Another factor, which contributes to improved traffic capacity, is that the modulation bandwidth after band spreading in many cases becomes much larger than the coherence bandwidth of the radio channel. Therefore, frequency diversity could reduce the required local average of $(C/I)_{min}$.

Figure 6.9 gives the relation between the required protection ratio $(C/I)_{min}$

($C/I = E_s/N_0$) and the number of traffic channels per cell for a spreading factor of 100. Perfect power control is assumed and DTX with 50% duty cycle. The continuous curve (not relevant) is for an isolated cell, the broken curve takes into account the capacity reduction due to interference from nearby cells.

DS-CDMA

A. Traffic capacity



B. Characteristics

- DTX (α = 50%)
- Interference Averaging
- Graceful Degradation - Trade-off between Capacity and Quality
- Statistical Multiplex - Variable Data Rates

Figure 6.9

Rake receiver

The prerequisites for the despreading of the wanted signal in the receiver mixer are both the right marking (coding) of the wanted received signal and the right timing. The timing accuracy to obtain the full processing gain is approximately one chip time, i.e. the inverse of the channel bandwidth B. If the bandwidth after bandspeading is large, the width of the impulse response of the propagation channel is considerably larger than one chip time. This means that the basic receiver structure can only be adjusted to detect signals corresponding to a small part of the impulse response (i.e. the largest multipath component). The rest of the wanted received signal is suppressed. The result is reduced receiver sensitivity, as only part of the received signal energy can be used in the detector.

An optimum receiver therefore becomes more complicated. It contains several detection channels with different code delays, which are adjusted to match the major components of the impulse response. This arrangement is called a **rake receiver**. (The ”rake” collects together the contributions to the total signal energi from several multipath components). See Fig. 6.10. The rake arrangement can also be considered as an **equalizer**. It is necessary to measure continuously the impulse response of the propagation channel in order to set the delay and phase of the different rake branches. Thus the output from the channels can be added coherently, giving predetector diversity combining.

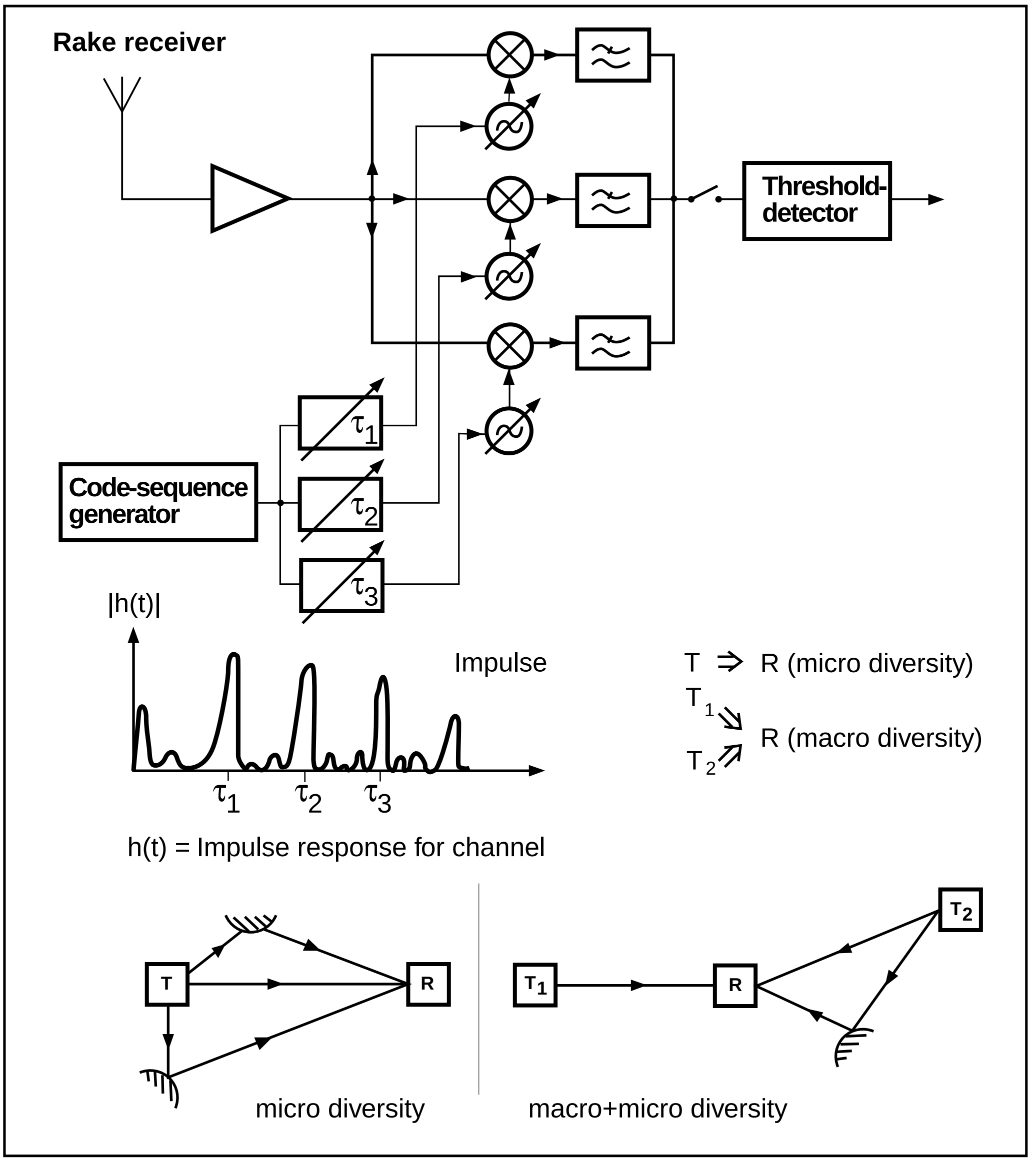


Figure 6.10

The rake arrangement can also give **macro diversity with soft handover** in the outward direction. The same source signal is transmitted from two base stations with overlapping coverage (in which the terminal is situated) and marked, using spreading codes, known to the terminal. The receiver also knows the impulse responses of the propagation channels. The code delays of the rake branches are adjusted for simultanious reception of multi-path components from both base stations.

Power regulation

As mentioned above, the power regulation must apply also to the fast fading, so that the variations in signal level due to fast fading are eliminated in the input of the base receiver. (In the outward direction, only rough power control is necessary, as all the signals from the base transmitter have the same propagation loss.) The basic DS-CDMA arrangement uses frequency duplex. This means that the fast fading is uncorrelated in the outward and the reverse directions. Therefore, a closed loop control is necessary with power commands from the base to the terminals. The speed of the control link is limited, however, which reduces the efficiency of the power control at high terminal speeds.

Only the output power of the terminals assigned to the own base station can be controlled. That means that the received interference from terminals, assigned to nearby base stations, would show strong variations due to fading. In order to limit this interference, it is necessary to use soft handover, in order to minimize the terminal output power, when it is in the handover region.

In addition, the relative power levels of different base stations must be set and controlled very accurately, as these power levels determines the cell boundaries. (A macro cell must use a much higher power level than a coexisting micro cell.)

On the other hand, a TDMA system must use hard handover. The difference in diversity gain between hard handover (in connection with GSM) and soft handover is only about 1 dB, see figure 6.11. Hard and soft handover has been discussed in section 5.2.3.

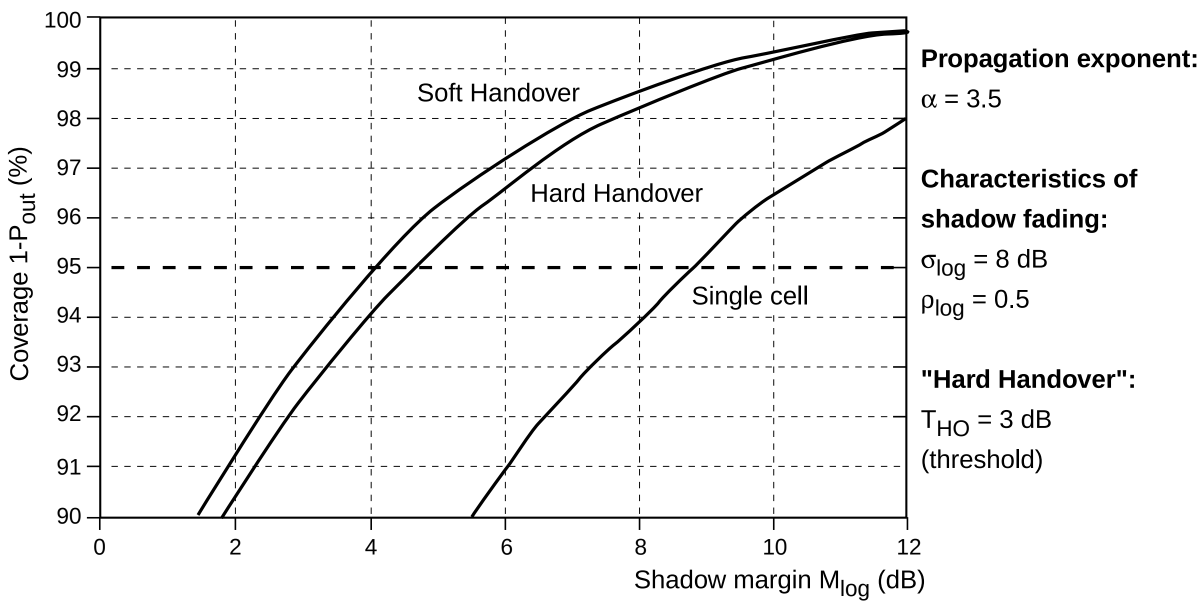


Figure 6.11

Advanced DS-CDMA systems include a time frame structure, which permits time division duplex (TDD) instead of FDD. If the time difference between the up and down slots are less than the correlation time of the propagation channel, the fast fading is highly correlated between an uplink and a downlink slot. That means that closed loop power control is no longer necessary, as the terminal obtains all information, needed for the optimum setting of the output power, from observation of the level of a pilot tone from its base station.

The main motivation for power control is that the interference level must not be too large. The amount of fast fading decreases with increasing bandwidth. See figure 6.12. This is the same effect as the frequency diversity, when the signal bandwidth becomes larger than the correlation bandwidth. The power regulation is therefore simplified if the spread bandwidth is much larger than the correlation bandwidth.

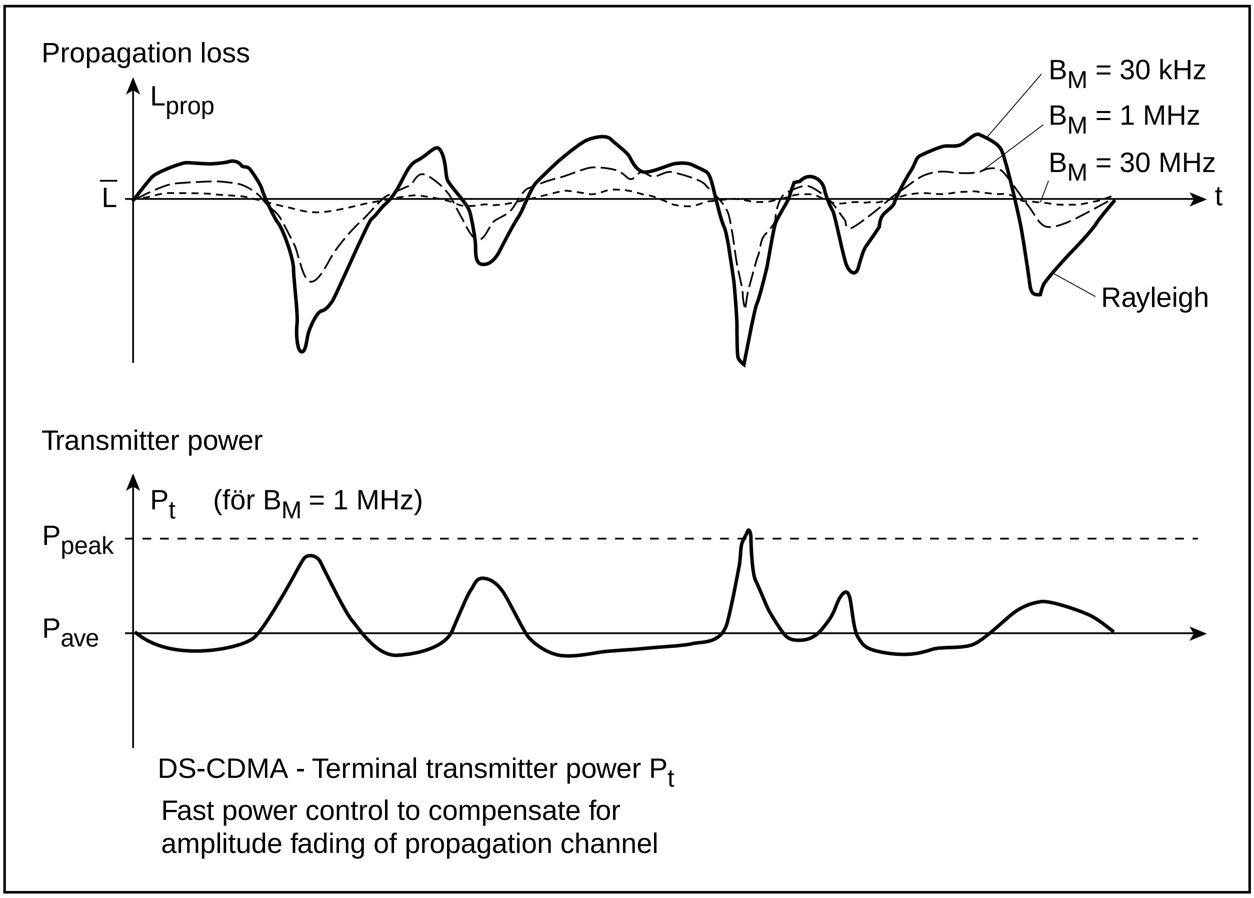


Figure 6.12

Another aspect of the power control is to keep up the level of the wanted signal during fading dips. However, it might be difficult to increase the output power sufficiently, when terminals are at the cell perifery. The problem is that the required peak power of the transmitters must be much higher that the average transmitter power, which is adjusted to the maximum level at the perifery of large macro cells.
If the signal level at the base station receiver drops during fading dips due to limitations on the terminal peak power, the effective signal level is reduced, which means reduced useful range.

Limitations

a. A practical limitation of DS-CDMA could be unsufficient available bandwidth for reasonable processing gain. The minimum spreading factor for reasonable multiple access performance seems to be about 10 (if interference cancellation is not used).

b. An implementation difficulty is the dynamic power control, especially for high speed terminals with rapid multipath fading.

c. Another problem is the rake detector, which becomes extremely complicated with many channels, if the width of the impulse response amounts to many chip times.

d. A limitation with a basic DS-CDMA system is that the terminal receiver must receive the wanted signal continuously. That makes MAHO impossible between hierarchical layers, which should use different carriers. The same problem applies with handover between cells with differerent numbers of carriers (hot spot problem) or when for other reasons cells must use differerent carrier frequencies.

This problem can be solved by introducing time frame stucture. At CODIT, parallell channels are established during handover. This makes it possible to simultaniously receive signals from the two cells, involved in the handover procedure.

6.5 FH-CDMA

The main alternative to DS-CDMA is FH-CDMA. In this case, frequency hopping is used to spread out each radio connection over the allocated frequency band. In the general case, the simultanious connections use **random hopping structures** without mutual coordination. See figure 6.13.

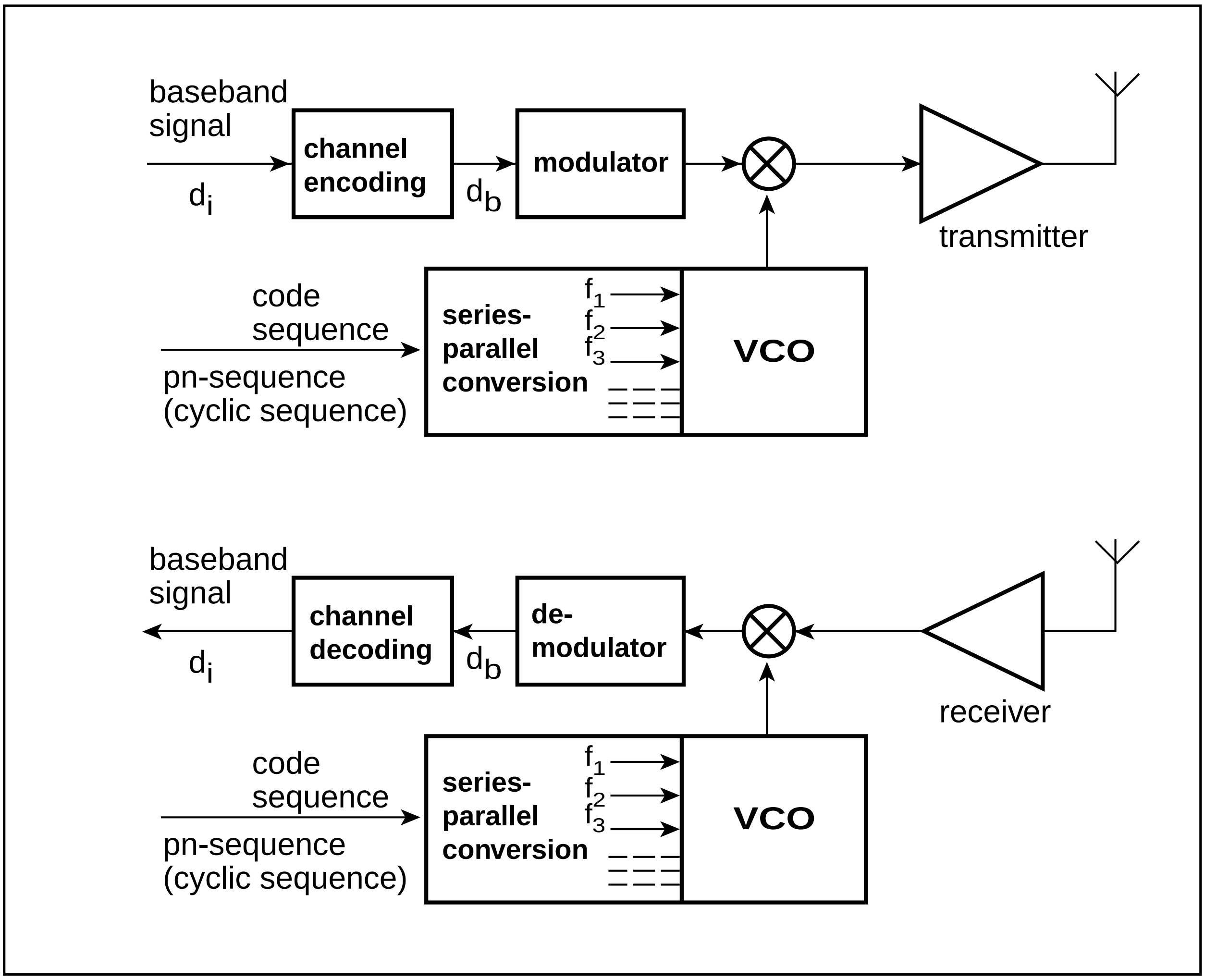


Figure 6.13

The system bandwidth should be larger than would have been necessary for a corresponding TDMA system without frequency hopping. This corresponds to a certain bandspreading.

To implement an interference-limited system with all the advantages mentioned above, it is necessary that many connections, even over a short time interval (effective symbol length), share a common wide band radio channel. An effective symbol length with an extension over several hops is obtained by using FEC and interleaving. This is necessary both for full processing gain and good frequency diversity.

As example, if the FEC works up to 10% ber into the channel decoder, the interleaving depth should be at least 5. (An error burst with 50% error rate due to a collision or deep fading dip will than be spread out over a 5 times longer interval, which reduces the ber to 10% over this interval.)

Due to the random frequency hopping, there is a certain collision probability, i.e. risk that more than one connection use the same frequency slot during the same hop interval. Generally, this results in so strong cochannel interference, that all useful information in the colliding data blocks is lost. However, the lost information can in most cases be recovered by the channel coding in combination with interleaving over several blocks. (Compare the corresponding GSM arrangements to bridge over fading dips due to multipath fading.)

The interference limitation is determined by the fact that with increasing traffic load, i.e. increased cochannel interference, the collision probability is increasing more and more. Finally the collision rate becomes so large that the channel coding in many cases cannot correct the errors due to collisions. Thus the transmission quality becomes too low. The collision rate can be decreased by band spreading. In principle, the relation between spreading ratio and processing gain described for DS-CDMA also applies to FH-CDMA.

The assumption that the average value of C/I should correspond to the required protection ratio is a certain simplification. Especially if only a relatively small number of connections share the common radio channel and also the required reliability is high, the worse-case performance is degraded due to statistical deviations from the average interference level. Therefore, it might be necessary to design the system (i.e. cluster size) so that the average value of C/I is several dB larger than the required protection ratio.

It is very complicated to make a mathematical assessment of the situation. It is often necessary to use computer simulations. These also take into account the effect of shadow fading, when estimating the service area reliability for different cluster sizes and protection ratios.

There are several arrangement of frequency hopping, see figure 6.14.

FH -TDMA

Channel coding supported by interleaving

Coordinated frequency hopping within each cell
orthogonality within each cell
frequency diversity

Channel-limited system

FH -CDMA

Channel coding supported by interleaving

Coordinated frequency hopping within each cell
orthogonality within each cell
frequency diversity

plus

Random frequency hopping between cells in an interference-limited system with:
averaging of I
demand assignement, soft capacity limit
DTX, dynamic power control
process gain through band spreading

Figure 6.14

We can discuss this in connection with GSM.

The connections within each cell use **coordinated or cyclic frequency hopping**. Different connections use different time shifts of the same hopping pattern. This gives the best frequency diversity and the connections are fully orthogonal as collisions are avoided.

Different cells use differerent hopping patterns, with a **random mutual relations**. This means that connections belonging to different cells are quasi-orthogonal, as random collisions occur. Mutually random hopping patterns give the best averaging of cochannel interference, even if the full performance of an interference-limited system is not obtained.

The same combined hopping arrangement can be used for FH-CDMA, retaining the advantage of orthogonality within each cell. In an interference-limited system, the system capacity can be improved by DTX. DTX gives reduced loading of the radio channels, which can be interpreted as a decrease of the average data rate per traffic channel and thus increased processing gain.

In connection with this arrangement are used the concepts **system load** or **fractional loading**, which means how large percentage of the available channels that are utilized, and **interference load**, i.e. the percentage of the channels which in average are active.

If the effect of imperfect statistical averaging is taken into account, the effective interference load for a speech connection is typically half of the system load.

The effect of fractional loading is that only some of the cochannel transmitters are active during each time slot in the frequency hop structure, which reduces the average level of cochannel interference. An alternative point of view is that fractional loading corresponds to band spreading, as more channels (larger system bandwidth) are used than for a normal, fully loaded TDMA system.

The frequency hopping arrangement also distributes the cochannel interference more randomly over time, which makes it easier for the channel decoder to bridge over the time slots, which are hit by cochannel interference. This gives a few dB of additional coding gain. Frequency hopping over a relatively large frequency band also gives an additional gain due to frequency diversity.

GSM in a FH-CDMA mode works with a cluster size of 3 instead of a cluster size of 9 which is usually used at the FH-TDMA version. This corresponds to 3 times better frequency economy. However, according to simulations fractional loading with 2/3 is necessary to reduce the cochannel interference to an acceptable level with 3 cell cluster. Thus, the net result is about 2 times improvement in frequency economy.

An interference limited system, such as FH-CDMA with fractonal loading, has no hard capacity limit. As discussed above, such a system has graceful degradation, that effectively gives a further improvement of the traffic capacity.

The development from NMT to traditional GSM and to FH-CDMA based on GSM is summarized in Fig. 6.15.

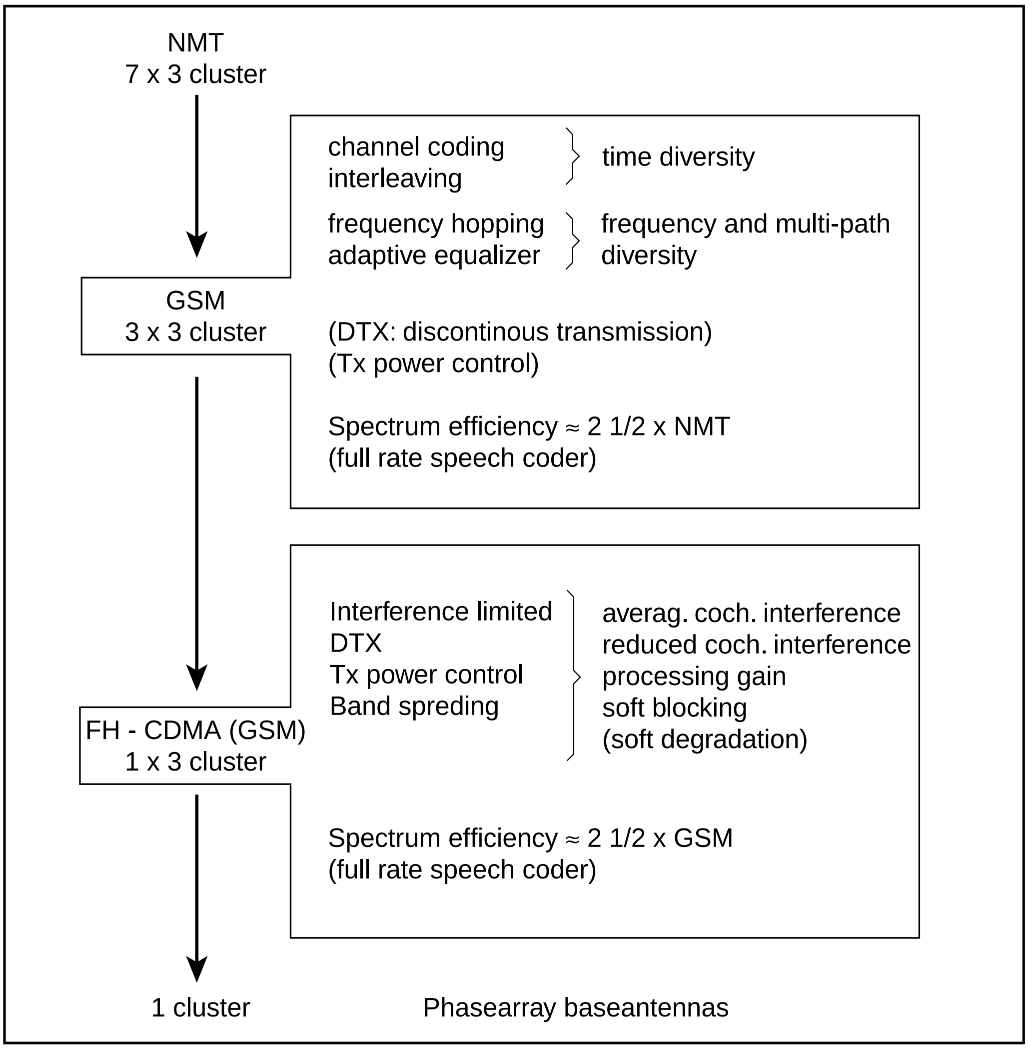


Figure 6.15

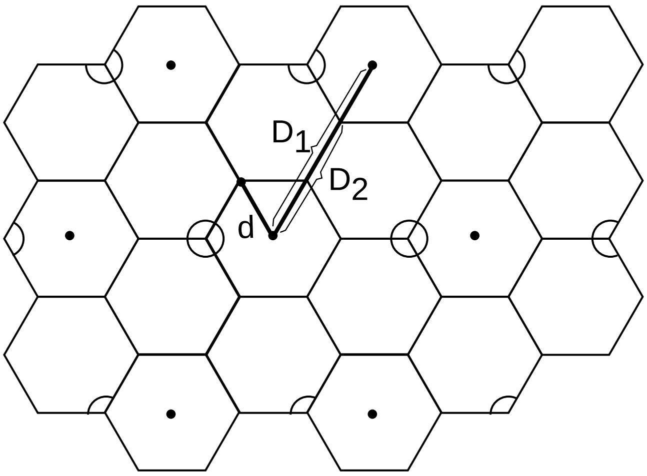
Example of interference averaging

As a concrete example of interference averaging, let us make an extremely rough analysis of the case discussed in appendix 2. Detailed simulations have shown that the basic GSM system can be arranged as a FH-CDMA system with cluster size 3, if an availability over a cell area of 90% is sufficient. To decrease the interference sufficiently, only 2/3 of the TDMA channels are used and in addition DTX is used with 50% duty cycle. That means that the probability that a cochannel transmitter is active becomes 1/3 (33% interference load).

Another improvement is antenna diversity at the base stations. Therefore, it is sufficient that the local average of the protection ratio is 5 dB. Also macro diversity is used, which reduces the required shadow fading margin to 5 dB.

Assuming that the system bandwidth is 6 MHz in either direction, the number of 200 kHz channels per cell becomes 6 x 5/3 = 10. The dominating cochannel interference comes from the 6 nearest cochannel cells. The total number of cochannel transmitters therefore becomes 60. Of these in average 1/3, i.e. 20, are active simultaniously.

Next we can study the cell stucture, see figure 6.16.



$$\frac{D_1}{d} = 3 \Rightarrow \left(\frac{D_1}{d}\right)^4 = 19dB$$

$$\frac{D_2}{d} = 2.1 \Rightarrow \left(\frac{D_2}{d}\right)^4 = 13dB$$

Figure 6.16

Distance D_1 is the average distance to a cochannel terminal transmitter, distance D_2 corresponds to the worst case (smallest distance). Assuming that the propagation exponent is 4, the C/I value corresponding to D_1 becomes 19 dB (global protection ratio). The C/I corresponding to D_2 becomes 6 dB less. Therefore the worst case I becomes 6 + 5 = 11 dB larger that the average (adding together the distance factor and the shadow margin). To obtain efficient interference averaging, the worst case I (from one cochannel transmitter) should be less than the average cochannel interference. As 20 (13 dB) cochannel transmitters are active each instant, the worst case I is roughly 13 - 11 = 2 dB less than the average interference.

As the number of active transmitters are fairly small, we do not get perfect averaging and the peak interference is somewhat more than the average value. Without averaging, the difference between the worst case I and the average is 11 dB = 13 times (see above). Averaging reduces this difference with roughly the square root of the number of interferers (20), i.e. 4.5 times. The margin to be added due to imperfect averaging becomes 13/4.5 = 2.5 times = 4 dB.

Another type of imperfect averaging is random traffic variations (number of connections, DTX function). At least 1 dB additional margin should be added.

With the parameters indicated above, we can make a very rough estimate of the required global average of the protection ratio:

Local average of protection ratio	5 dB
Shadow margin	5
Number of cochannel interferers/channel 20/10 =2	3
Margin for imperfect averaging	4
Margin for random traffic variations	1
Required global protection ratio	≈18 dB

This corresponds to the figure 19 dB, estimated above for the 3-cluster structure.

6.6 OFDM (OFDMA)

A third type of wide band multiple access is called **Orthogonal Frequency Division Multiplex** (OFDM). It was originally introduced for digital TV, i.e. only for the outward direction from fixed transmitters to fixed and mobile broadcast receivers (single access). OFDM is therefore, strictly speaking, a modulation method combined with a multiplex arrangement and not a multiple access arrangement - which explains the name. The results from the broadcast application cannot directly be taken over to mobile radio applications, as the inward direction (corresponding to multiple access) generally is more complicated to implement than the outward direction. The basic characteristics of OFDM can however be extended to a wide band multiple access method for mobile radio. A more appropriate name for this application would be OFDMA.

OFDM(A) can be interpreted as an extreme version of FDMA. The main difference is that each connection has access to several frequency slots, spread out over the shared frequency band. The band is divided up in a very large number of frequency slots, which are packed as tight as possible under the condition of full orthogonality. See figure 6.4 above. This means that the channel spacing is the same as the baud rate per channel. A complication is that for full orthogonality it is necessary that the data signals in differerent frequency slots are mutually synchronized. This might be difficult to arrange in the inward direction.

The total data rate (including channel coding) for a connection is this baud rate multiplied by the number of slots allocated to the connection. A simplified block diagram is scetched in figure 6.17.

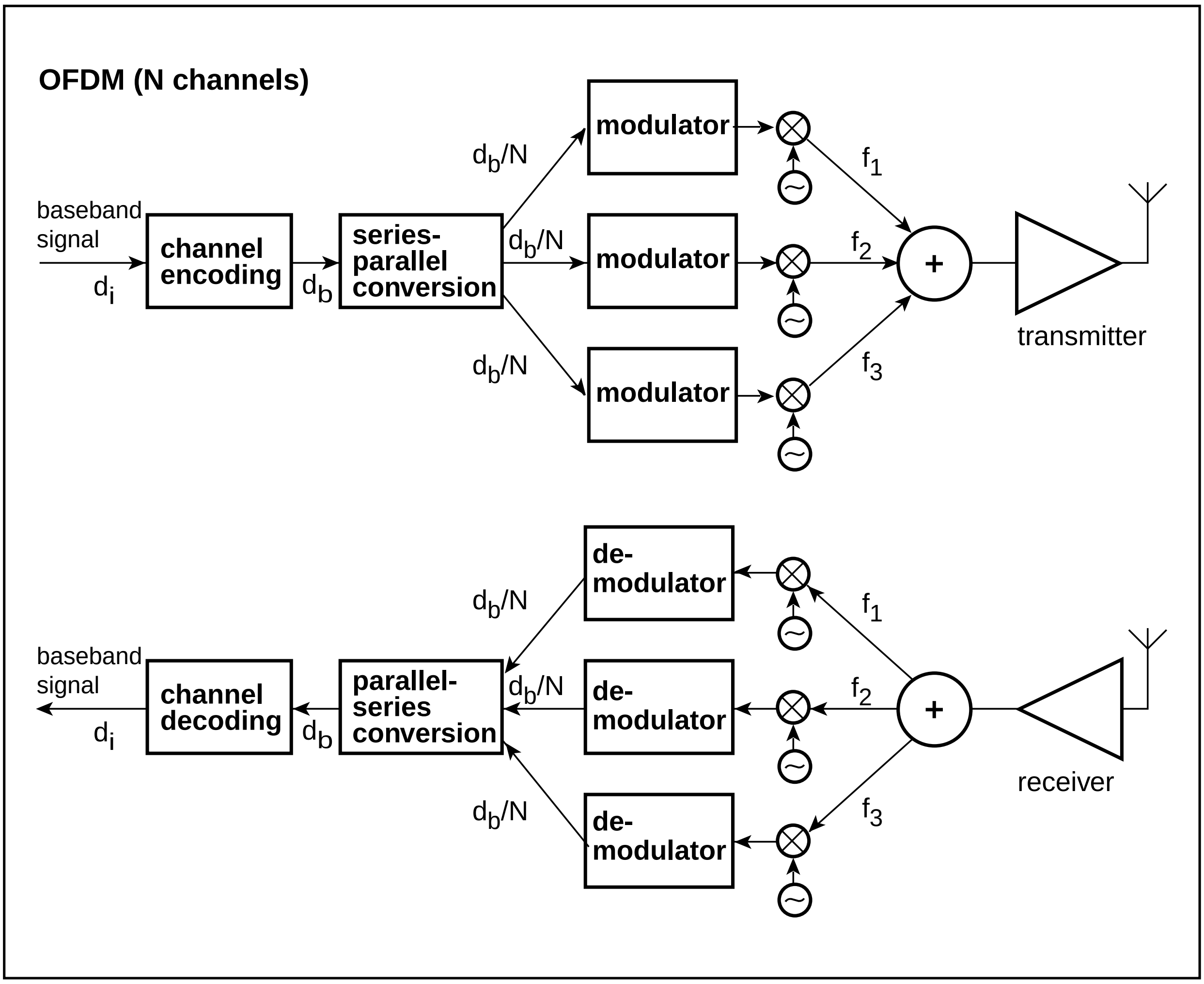


Figure 6.17

In order to get good frequency diversity performance, the arrangement above must be combined with channel coding supported by interleaving over the frequency slots allocated to a connection. The channel coding takes care of transmission errors in frequency slots, hit by outage due to selective fading. The basic OFDM system has the following characteristics:

a. Flexibel allocation of capacity to connections with different peak data rates. By means of a reservation procedure, wideband services are allocated a larger number of frequency slots than narrow band services. A drawback is that reservation procedures are fairly slow. The dynamic channel allocation is not fast enough to handle fast variations in source data rates.

b. Frequency diversity with good performance, if the total bandwidth is considerably larger than the correlation bandwidth of the propagation channel. Even if a small portion of the allocated frequency slots are hit by frequency selective fading, error free transmission is possible with the help of FEC + interleaving.

c. Low symbol data rate per frequency slot, i.e. symbols with long duration are used. This eliminates or at least considerably eases the channel equalization. (On the other hand equalization in the frequency-domain might be necessary to obtain enough isolation between frequency slots).

In the same way as for TDMA, a time slot structure with frequency hopping can be introduced. See figure 6.18. This could give several advantages. Cyclic frequency hop (with channel coding and interleaving) can increase the gain through frequency diversity, especially for narrowband connections, which only have been allocated a few frequency slots. The use of a random hopping structures gives averaging of cochannel interference in the same way as for FH-CDMA. The system can also be further developed to an interference-limited system through fractional loading, with the advantages discussed above in connection with FH-CDMA. Overall there are considerable similarities between OFDM and FH-CDMA.

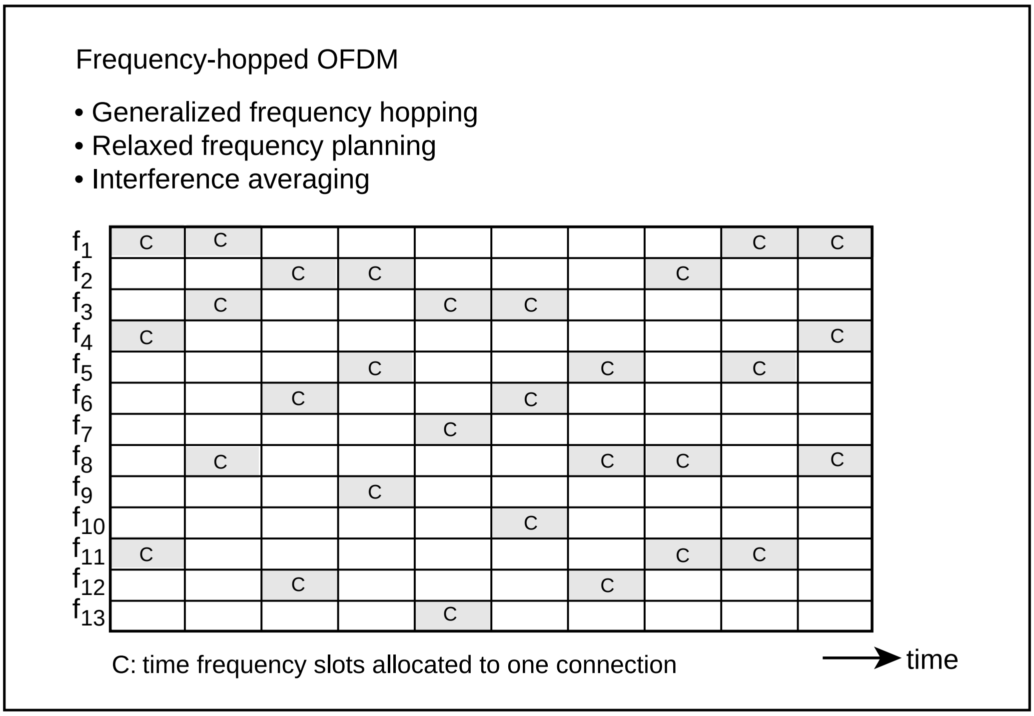


Figure 6.18

In connection with OFDM there are a few implementation problems:

a. High peak factor of the combined radio signal into the transmitter output stage. The usual assumption, when combining many carriers with random mutual phases, is that the sum signal will have a Rayleigh-distributed envelope, which will have a fairly high peak factor (depending on the relative time the signal is allowed to spend above the specified peak level). Simulations of OFDM signals indicate even higher peak factors. One reason is that the phases are not random, another is that rounded symbol waveforms must be used to keep down the out-of-band radiation.

b. Complicated digital signal processing is needed for the multiplexing to generate the OFDM signal and the corresponding demultiplexing at the receiver. (An often used procedure is IFFT on the transmitter side and FFT on the receiver side.)

c. Complicated symbol synchronisation and channel estimation.

6.7 Comparation between DS-CDMA and FH-CDMA

In this section is made a rough, elementary comparation between a FH-CDMA system based on GSM (see appendix 2) and a DS-CDMA-system, which corresponds to IS-95 (see appendix 1). To get a more or less fair comparation, it is assumed the same maximum data rate – 13 kb/s – from the speech coder. For the FH-CDMA system, the mean source data rate is reduced to 50% of the peak rate by DTX, for IS-95 is used a variable rate speech coder, whose mean rate is 6.5 kb/s.

At the comparation it is assumed that both the systems are allocated the same frequency band, i.e. 2 x 1.25 MHz. A considerable uncertainty is the required E_i/N_o ratio for IS-95. The value – 7dB – which is usually assumed, is probably too optimistic.

FH-CDMA

Cluster size: 3

System load: 2/3

Bandwidth per two way speech channel: 2 x 25 kHz

A 2 x 1.25 MHz wide radio band has room for 50/3 x 2/3 = 11 traffic channels per cell.

IS-95

The inward direction determine the frequency economy (non-coherent detector and degradation due to non-perfect power control). The bandspreading, including channel coding, gives a processing gain (information data rate 6.5 kb/s, bandwidth after spreading 1250 kHz) of:

$$G_p = 1250/6.5 = 192 \approx 23 \text{ dB}$$

The required signal-to-noise ratio (average over the multipath fading) into the data demodulator (taking into account the gain through diversity and FEC):

$$C/I = E_i/N_o = 7 \text{ dB}$$

This gives an Anti-Jamming margin of:

$$AJ = 23 - 7 = 16 \text{ dB}$$

From this basic value must be subtracted several degradations factors. The most important ones are:

Interference from nearby cells:	2 dB
Non-perfect interference averageing	2 dB
Non-perfect power control	<u>2 dB</u>
	6 dB

This gives a net AJ of 10 dB = 10 times, i.e. each cell has a capacity of 11 speech channels.

Results

According to this rough estimate DS-CDMA according to IS-95 and FH-CDMA based on GSM has the same frequency economy. A more detailed analysis gives about the same results, possible with a certain advantage for FH-CDMA, as all degradations factors connected with CDMA have not been included above.

6.8. Summary

The main topic of section 5 and 6 has been different means to reduce the cluster size such as:

- reduction of the fading margin through diversity

- reduction of the effective interference level by using an interference-limited system.

These aspects are summarized in figures 6.19 and 6.20.

Systems aspects common for FDMA/TDMA and CDMA

- Cluster size
- Simplified cell-frequency planning through reduced cluster size
- Macro diversity (soft or hard handover)
- Micro diversity (more than two branches desirable)
- Time dispersion (channel equalization, path diversity)
- Frequency (micro) diversity improved with increased bandwith
- Hierarchical cell structures (wideband services only in the smallest cells)

Systems aspects of CDMA (interference limited)

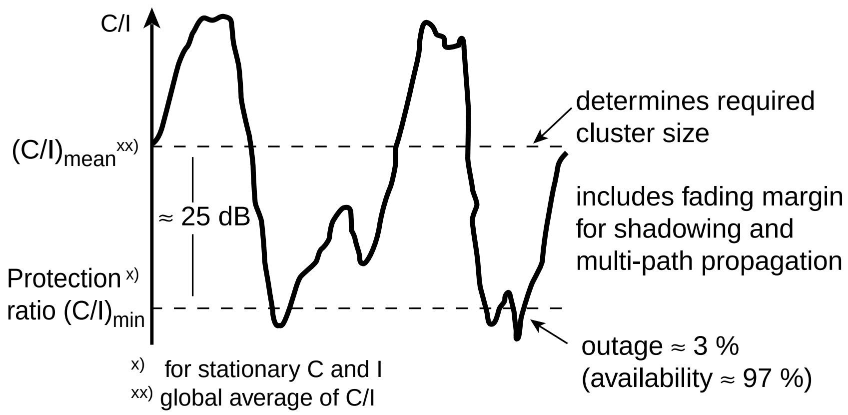
- Interference limited system
- Efficient averaging of cochannel interference $I_{max} \approx I_{mean}$
- Dynamic power regulation
- Fast, efficient demand assignment of channel capacity (DTX, variable source rate)
- Process gain through bandwidth spreading (to reduce cluster size)
- Quasi-orthogonal Multiple Access (basic system)
- Handover between layers in hierarchical cell structure (MAHO problem)

Figure 6.19

Figure 6.19 lists the factors which are the same for a channel-limited system and an interference-limited system and which are different.

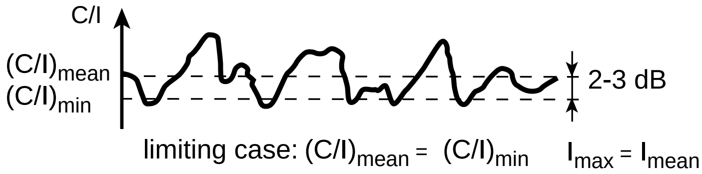
Figure 6.20 compares a basic cellular system without any signal processing to improve the situation with respect to fading and cochannel interference with an advanced system with several features to improve the situation both with respect to fading and cochannelinterference. The basic system roughly corresponds to NMT.

Basic system without diversity or averaging of I



Advanced interference limited system

with efficient averaging of a very large number of co-channel interferers, and combination of several diversity arrangements, including channel coding with interleaving



Here the following means are effective to reduce I_{mean} and thus I_{max} for a reduced reuse distance:

- DSI/DTX
- variable-rate speech coder
- dynamic power regulation
- band spreading

These measures give little improvement in the basic system above, as it is mainly the worst case combination of transmitter power and propagation loss that determines the reuse distance

Figure 6.20

The combination of different diversity arrangements reduce the variations of the wanted signal, so that the worst case of C (C_{min}) is only a few dB lower than the average C. With an interference-limited system with averaging of a very large number (of the order of 100) of cochannel interferers, the worst case of I (I_{max}) is only a few dB higher than the average I.

In a system without efficient averaging of cochannel interference, the worst case of I (i.e. corresponding to 97% availability) is hardly affected by power control or DTX, as the worst case corresponds to an active cochannel transmitter with maximum output power. The effect of power control and DTX is mainly to reduce the average I without much effect on the maximum I.

On the other hand with an ideal interference-limited system (extremely high number of interferers) the maximum value of the total interference is hardly effected by any single interferer. Any measure that reduces the total (average) interference level therefore gives about the same reduction of the worst case I (I_{max}) and the difference between I_{mean} and I_{max} can nearly be neglected.

7. Generation 3 of mobile telephone systems

7.1 Introduction

The probable scenario for the situation around year 2005 is a mass market for personal terminals, which will replace a large part of the wire-connected phones. They will also be used extensively for data services and multimedia with gradually increasing demands on data rates. The next generation of cellular systems (UMTS and IMT-2000) to be introduced around 2002, will allow user data rates up to 2 Mb/s indoors and to some extent in outdoor micro cells and up to 384 kb/s with wide area coverage. See section 7.3. In the U.S. there are similar plans to establish a wideband system based on IS-95 by increasing the system bandwidth with a factor of 3.

However, doubts have been expressed about a near-term mass market for data services as high as 2 Mb/s. For a long time, the dominating market for mobile telephone systems will be speech and moderate rate data services. The draw-back with UMTS/IMT-2000 is mainly the worse propagation conditions at 2 GHz which will make it expensive to extend the network to scarsely populated areas. For a long time, UMTS/IMT-2000 will be limited to local areas with high traffic density.

It will therefore be a need for combining GSM and UMTS into one integrated network. Users, that need both the advanced UMTS services in micro and pico cells and the wide-area coverage of GSM, will be equipped with double-mode terminals.

Therefore a parallel development to UMTS is the extension of the capabilities of the second generation of cellular systems such as GSM. The first step of the GSM evolution is to introduce HSCSD (High Speed Circuit Switched Data) and GPRS (General Packet Radio Serving) with data rates up to 115 kb/s. The next step ”evolved GSM” will increase the data rate to 384 kb/s by the use of linear high-level modulation (i.e. 0-16 QAM). See appendix 1 of module DM1 and figure 7.1.

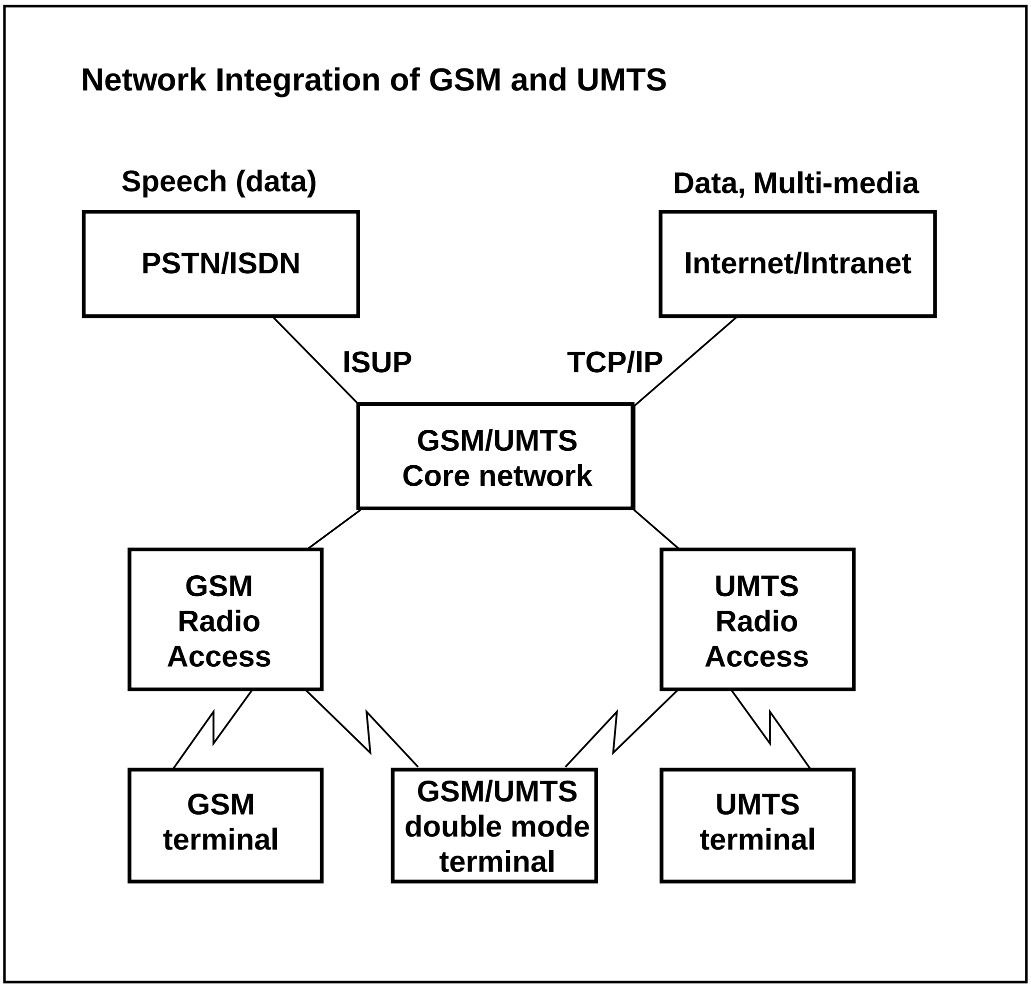


Figure 7.1

There is also a future need for local wireless networks with peak data rates around 20 Mb/s i.e. about ten times higher than the capabilities of UMTS. Different study groups are engaged in Wireless Lan and Wireless ATM, i.e. Hyperlan 2. This is discussed in section 8.

In Germany the equivalent to NUTEK is supporting a large research program called ATM-Mobile. The program is led by Bosch with participation by Ericsson EUROLAB and Aachen University. Another large EC research program related to W-ATM is Magic WAND. Nokia is one of the main participants in this program.

Wireless versions of LAN will mainly be used indoors for movable terminals, i.e. the application area is much more limited than for cellular systems. The attenuation in walls and floors will increase with frequency. At high microwave frequencies a natural cell size might be one room (”nano cell”). An alternative wireless technology is IR.

The trade-off between data rates and cell sizes for the systems mentioned above is indicated in figure 7.2.

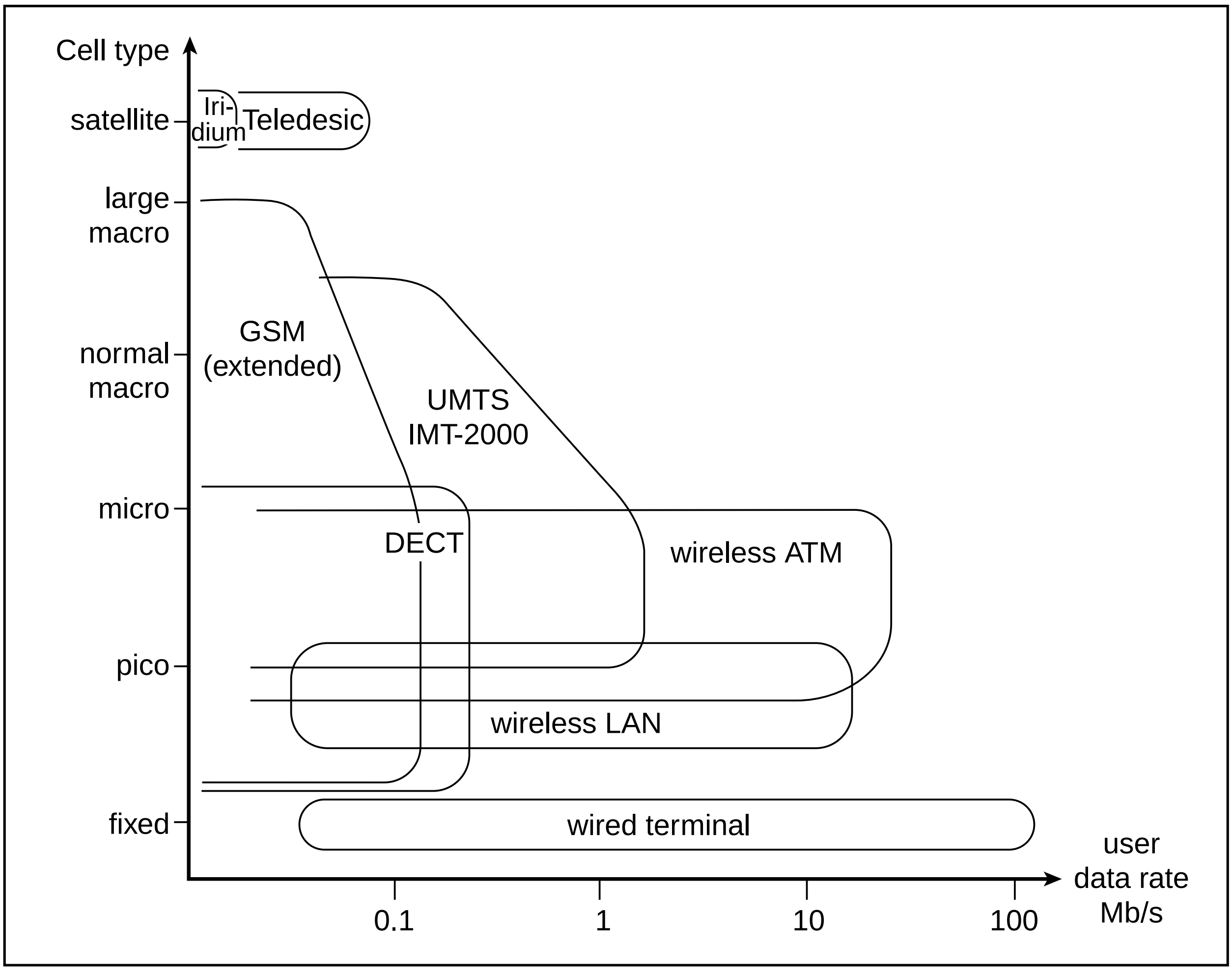


Figure 7.2

The different development steps from the first generation of digital mobile telephone to UMTS/IMT-2000 are summarized in figure 7.3

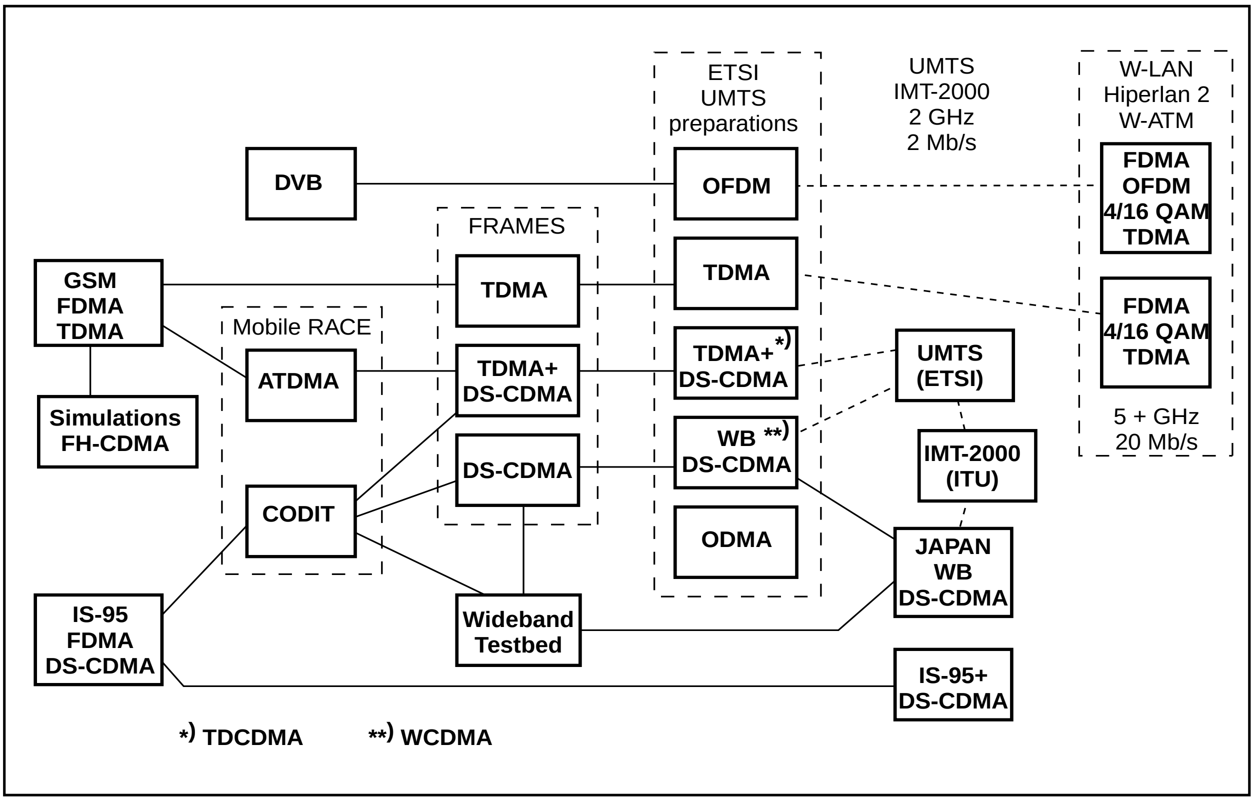


Figure 7.3

7.2. Developments in the US and Japan

USA

The market for IS-95 is still in the building-up phase and there is little user or operator interest for a system with improved capabilities. Therefore, it is not considered urgent to establish a new system of the IMT-2000 type. However, there are discussions about an evolved IS-95, which shall share the spectum with the present system. The basic module of the evolved IS-95 will use 3 frequency slots of 1.25 MHz. A system bandwidth of 3.75 MHz has been proposed (chip rate 3x1.2288 Mch/s) for the uplink. For the downlink it might be better to use 3 channels of 1.25 MHz in parallel, as this will retain the orthogonality through the Walsh functions between the present narrow-band system a new wide-band system.

Certain system improvements will also be introduced. Coherent detection will be used also for the uplink. Other means than pulsing will be used for the power control of the terminal transmitters.

Japan

It has already been decided to use DS-CDMA (WCDMA) for the third generation system in the IMT-2000 frequency band. There is an urgent need for a new system with better frequency economy, even if speech will be the dominating service for a long time. The objective is that a fully operational system shall be available to an important international sport event 2002.

In the first phase a chip rate of 4.096 Mchip/s will be used, but later the system bandwidth can be increased by using a multiple of this basic chip rate. User bit rates up to 384 kb/s will be used in wide area applications and up to 2 Mb/s indoors.

Through continuous contacts with ETSI with mutual adaption of the specifications, the air interface is of the Japanese system is now nearly exactly aligned with the corresponding WCDMA alternative of ETSI.

Ericsson participates in the development of experimental systems for two major operators. The system for NTT DoCoMo was ready for test late 1997 and will be tested during 1998. The system for Japan Telecom will be ready for test before the end of 98. The background for the Ericsson participation is the **wide-band testbed** based on the **CODIT** experiences. The main features of CODIT are summerized in appendix 3.

7.3 UMTS/IMT-2000

7.3.1 Background

The development towards personal mobile telecommunication in Europe is closely related to the concept of **UMTS (Universal Mobile Telecommunication Systems)**. Universal means two things: firstly the system shall be universal with respect to the adaption to many different services (multimedia, INTERNET), secondly the different UMTS systems shall work together to form an integrated network structure with more or less worldwide coverage and permitting unlimited roaming between IMT-2000 systems. This will be partly possible if IMT-2000 (see below) will be based on the WCDMA air interface used in Europe and Japan. However, it is uncertain if this IMT-2000 standard will be introduced in USA.

A related concept to UMTS is **FPLMTS (Future Public Land Mobile Telephone Systems)** which for many years has been studied by ITU study groups. As these systems will soon be implemented (ITU:s basic IMT-2000 decisions are expected in the end of 98), the ”future” in the name is no longer appropriate. The designation FPLMTS has therefore been changed to IMT-2000 (International Mobile Telephone for year 2000).

The 1992 World Administrative Radio Conference (WARC 92) allocated a total of 230 MHz of spectrum to be used on a global basis for FPLMTS. The bands 1885-2025 MHz and 2110-2200 MHz were assigned (see figure 4.1b). In Europe it is planned that UMTS, when operating in a frequency duplex mode, will use the bands 1920-1980 and 2110-2170 MHz (duplex spacing 190 MHz).

The allocated bands around 2 GHz might not be sufficient. It would therefore be a future need for around 200 MHz of additional spectrum. Frequencies about 3 GHz have been discussed. However, it is a problem that no common worldwide band could be made available. It also seems impossible to establish world-wide IMT-2000 coverage in the 2 GHz band, as a large part of the band has already been allocated for other services in the US (PCS).

UMTS is discussed in more detail in the next section. The universal service concept also means that UMTS might be extended to **dispatch applications**. (Examples of digital systems for dispatch applications are the already well established MOBITEX system and the trunked system TETRA, for which ETSI standards have recently been approved.)

Personal communication will use **personal telephone numbers** instead of the many different numbers (to work, home, summer home, telefax, mobile telephone etc.), which are needed today. The telephone number is coupled to a certain person instead of a certain fixed connection. Roaming is made possible through a **smart card** which will contain a personal ID and an associated subscriber number. When travelling abroad, the card is used to log-in to the public telephone network so that the subscriber will be accessible to incoming calls. It also gives access to telephone services for outgoing calls with bills sent home as usual for the calls they make. Another possibility is to buy phone cards, loaded with suitable number of traffic units.

Two new concepts have been introduced with respect to roaming: **personal mobility and equipment mobility.**

Personal mobility means the possibility to connect to many points in the public telephone network with the help of personal telephone numbers. The concept, which is developed within ITU-T, is **called UPT (Universal Personal Telecommunication).**

Equipment mobility means that fixed telephones are replaced by wireless terminals. Personal mobility does not necessitate the use of wireless pocket terminals, as it can be achieved by the future intelligent telephone network. It does, however, constitute an essential element of the future personal communication network.

One of the result from the **FPLMTS** study groups was a goal specification, including an estimate of the needed spectrum allocation. See figure 7.4.

FPLMTS: Overview taken from IWP 8:13 (T G8/1)

Characteristic		***Mobile***	***Personal Station (PS)***	
		M	*PS1*	*PS2*
Cell Plan		*Outdoor*	*Outdoor*	*Indoor*
Cell Area:		*(min) 0.5 - 1* km^2	*(typical) 16000* m^2	*(typical) 600* m^2
Base Station Antenna Height		*50 m*	*<10 m*	*<3 m*
Service area reliability		*90%*	*>90%*	*99%*
Base Station Installed indoor/outdoor		*No/Yes*	*Yes/Yes*	*Yes/Yes*
Voice traffic per station		*0.10 E*	*0.04 E*	*0.20 E*
Non-voice traffic per station		*0.05 E*		*0.11 E*
Voice traffic per km^2		*500 E*	*1500 E*	*20000 E*
Non-voice traffic per km^2		*80 E*	*150 E*	*5000 E*
Blocking		*2%*	*1%*	*<0.5%*
Station:	*Volume*	*Vehicle mounted*	*<approx. 200* cm^3	*<approx. 200* cm^3
	Weight	*or portable*	*<approx. 200 g*	*<approx. 200 g*
	Highest Power	*5 W*	*50 mW*	*10 mW*

Figure 7.4

The background to the estimate of the needed spectrum allocation was the assessment of the required traffic capacity for different services together with reasonable cell sizes. The needed allocation was also influenced by the increased requirements of service area reliability and low blocking. To make radio access really attractive in comparation with wire access, the service quality must be nearly as high as in present telephone network with fixed telephones.

The activities toward UMTS and FPLMTS are closely coordinated and the two concepts have similar overall requirements, see figure 7.5.

A. User requirements on UMTS

Wide range of services

- Multi media (speech of different quality, music, video, data), with possibilities to establish several simultanious services
- Packet type of services, i.e. INTERNET/INTRANET
- Maximum data rate 2 Mb/s
- Different requirements on Quality of Service (error rate and transmission delay)

Universal coverage with high service area and time availability

- Worldwide coverage of densely and scarsely populated areas
- Seamless (soft) handover

B. Operator requirements on UMTS

High traffic capacity

- Adaption to the capacity needs of different services
- Hierarchical cell structure

Cost effective

- Cost of radio equipment
- Transmission cost in the fixed telephone network
- Installation and maintenance costs
- Simple cell-frequency planning
- Gradual capacity increase and updating
- Compatible with GSM

Figure 7.5

To develop the needed new technology for UMTS, two large European research programs were established within the framework of **Mobile RACE** (RACE: Research for Advanced Communication in Europe). These research programs, which ended late 1995, studied two different radio alternatives. One of the projects, **CODIT** studied DS-CDMA. The other project, **ATDMA**, extended the TDMA technology, used for GSM, by increasing the maximum user rate, adding packet transmission and including adaption of the transmission parameters to different cell sizes (with different maximum user rates). The objective of both programs was to allow user data rates up to 2 Mb/s.

Overviews of the radio features of CODIT and ATDMA are given in appendix 3 and 4.

7.3.2.FRAMES

The CODIT and ATDMA programs were continued through a more applied research program, called FRAMES. The main participants were Siemens, Ericsson, Nokia, CNET. The directive of ETSI was that different MA alternatives should be compared, and one specific recommendation for UMTS should be given to ETSI at the end of the program. However no specific recommendation was made. Instead three alternatives were presented:

a. W-TDMA

This alternative is in principle a wideband GSM system, with 8 times larger system bandwidth (1.6 MHz channel spacing). Each frame of 4.615 ms contains 64 time slots. A multiframe of 13 basic TDMA frames is used. 12 of the frames are used for traffic. Multi-slot operation is possible, and 4 time slots can be combined into one channel. Several types of modulation could be used: GMSK, 4QAM, 16QAM (compare ATDMA in appendix 4).

The advantage of this alternative is good compability with GSM. Disadvantages are:
- high peak factor (64), which complicates the terminal output stage
- very complicated equalizer (32 taps), which will probably limit the wide-area capabilities.

b. TDMA/DS-CDMA hybrid ("TDCDMA")

As for GSM a basic TDMA frame of 4.613 ms with 8 time slots is used. The data rate per time slot is 16 kb/s (raw data rate 25 kb/s). Increased user rate is possible through multi-slot and multi-code. Each time slot is multiplexed between 8 users by means of DS-CDMA with spreading up to 1.6 MHz.

The spreading ratio is fairly low, which means that the normal processing gain is unsufficient to allow 8 simultanious connections. Therefore, interference cancellation (joint detection) must be used. The practical implementation is extremely complicated (10^8 multiplications per second). For the interference cancellation it is necessary that the spreading codes of all the interfering signals are known. That means that DS-CDMA can only be used for multiple access within each cell. TDMA is used to obtain isolation between connections in differerent cells.

With frequency hopping reuse 3 is possible, but this means that an operator must be allocated at least 9 frequency slots of 1.6 MHz (at least 3 frequencies must be available in each cell to make frequency hopping worthwhile). In many cases, this is more than what can be assigned to an operator. Without frequency hopping, a cluster size of 6 is necessary, i.e. an operator must be allocated 6 channels. However, this arrangement gives no improvement of the frequency economy compared to GSM.

The advantage with this alternative is fairly good compability with GSM. The disadvantage is very high receiver complexity, which especially will be a problem for small portable terminals.

An advantage (for Siemens, Alcatel, Motorola) or disadvantage (for Ericsson and Nokia) with this alternative is that it differs from the development outside of Europe – IMT-2000 will very probably be based on wideband DS-CDMA (preferred MA-alternative in USA, Japan and Korea). The result will probably be a market segmentation between Europe and the rest of the world.

c. Wideband DS-CDMA ("WCDMA")

This alternative is based on CODIT and the continued development of ERA of the Wideband Testbed. It quite similar to the final WCDMA ETSI alternative, see section 7.3.4.

7.3.3 ETSI standardization

Within ETSI, 5 study groups under SMG2 have worked have in parallell with the specification of 5 different air interfaces. The original objective was that the SMG group should select one of the alternatives at the end of 97. However there has been some delay. The first vote within SMG2 gave a majority for the α alternative (WCDMA), but not enough to be basis for an ETSI recommendation. Further discussions within SMG2 resulted in a compromise, see figure 7.8. Even if UMTS will mainly be based on WCDMA, a secondary mode of operation will be based on the δ-alternative (TD-CDMA). The detailed air interface specification for UMTS will be defined during 98.

Two alternatives remained as the main options late 97:

The α study group developed a proposal based on the FRAMES WCDMA alternative. Main participants in the group were ERA och Nokia. Also NEC, Fujitsi and Panasonic made contributions.

The δ study group put forward a proposal based on the FRAMES hybrid TDMA/CDMA alternative. Main participants were Siemens and Alcatel.

The other three study groups worked with alternatives, which were discarded before the final discussions at the end of 1997.

γ study group: FRAMES WTDMA

β study group. OFDM (Sony, previously also Telia)

ε study group: ODMA (“Opportunity Driven MA). The ODMA proposal contained no specific MA-alternative, but was a combination of different supporting technologies (i.e. efficient packet transmission) which could be combined with the proposals from the α and the δ groups.

The transport layer of the fixed radio network will be based on ATM, both for UMTS and GSM. The major differencies between an evolved GSM system and UMTS is therefore the air interface.

7.3.4 WCDMA alternative for the UMTS air interface

The proposal from the ETSI a group is practically the same as the preliminary air interface standard for the Japanese ITM-2000. However, the ETSI compromise (see figure 7.8) means only FDM will be used.

Time frame structure

A more advanced timeframe arrangement is used than in CODIT. The basic frame of 10 ms contains 16 time slots and can be used for TDD. In principle TDD can only be in micro and pico cells. TDD can handle asymmetric traffic. Another advantage is that the fast fading structure is identical in the inward and outward directions, if the corresponding time slots has small enough time separation. This gives a considerable improvement and simplification of the power control (open loop sufficient). FDD is used for large cells, as TDD would need excessive guard times to handle the large variations in propagation delay for different terminals (including terminals registered at adjoining cells).

If TDD is used, time syncronization is needed between near-by cells. This is not required when FDD is used.

The time slots structure can be used for TDM as an alternative to CDM, i.e. for multi-media. TDM gives a simpler terminal, especially the transmitter.

Band spreading, FEC

In the first phase, the system bandwidth is determined by a chip rate of 4.096 Mchips/s. The required channel spacing depends on the level of adjacent channel interference, i.e. a larger separation is needed between channels used by different layers in a hierarchical cell structure. The channel spacing can be chosen as suitable multiple of 200 kHz. Typical channel spacings are in the range 4.4 to 5.2 MHz.

In the **downlink** similar arrangement as for IS-95 is used. An improvement is that the Walch codes have been replaced with Othogonal Variable Spreading Factor (OVSF) codes, which can handle a wide range of user data rates. This is the first spreading process, used to isolate connections within each cell. The second spreading process is based on a 10 ms segment of a long Gold-code (2^{18}-1). Different 10 ms segments of the code are used for each cell.

In addition a short spreading sequence is used for the pilot.

QPSK is used for the data and BPSK for the spreading.

In the **uplink** the spreading is made by a 10 ms segment of a (2^{41}-1) Gold code. O-QPSK modulation is used.

Coherent modulation is used for both uplink and downlink. The reference frequency is derived from the pilot channel.

In a later phase, **interference cancellation** will be introduced to get a further improvement of the frequency economy.

In addition to the code spreading, FEC coding is used. For the traffic channels a convolutional code (R=1/3, K=9) is used and for the control channels a convolutional code with R=1/2, K=9. If extemely low error rates are required, an outer Reed-Solomon Code (36,32) can be applied.

Time slot structure

In the **downlink**, each time slot of 0.625 ms contains both a control and a data channel. The control channel has fields for pilot, PC (Power Control) and RI (Rate Information) The data channel can be multiplexed between several services. See figure 7.6a. In addition, also the data field includes pilot information. This is to prepare for the future use of phase array antenna with antenna lobes, which might move during one time slot.

Demand assignment of channel capacity is through DTX, which is acceptable at the base transmitter as several channels are combined. (The amplitude variations of the transmitted signal will therefore be fairly small).

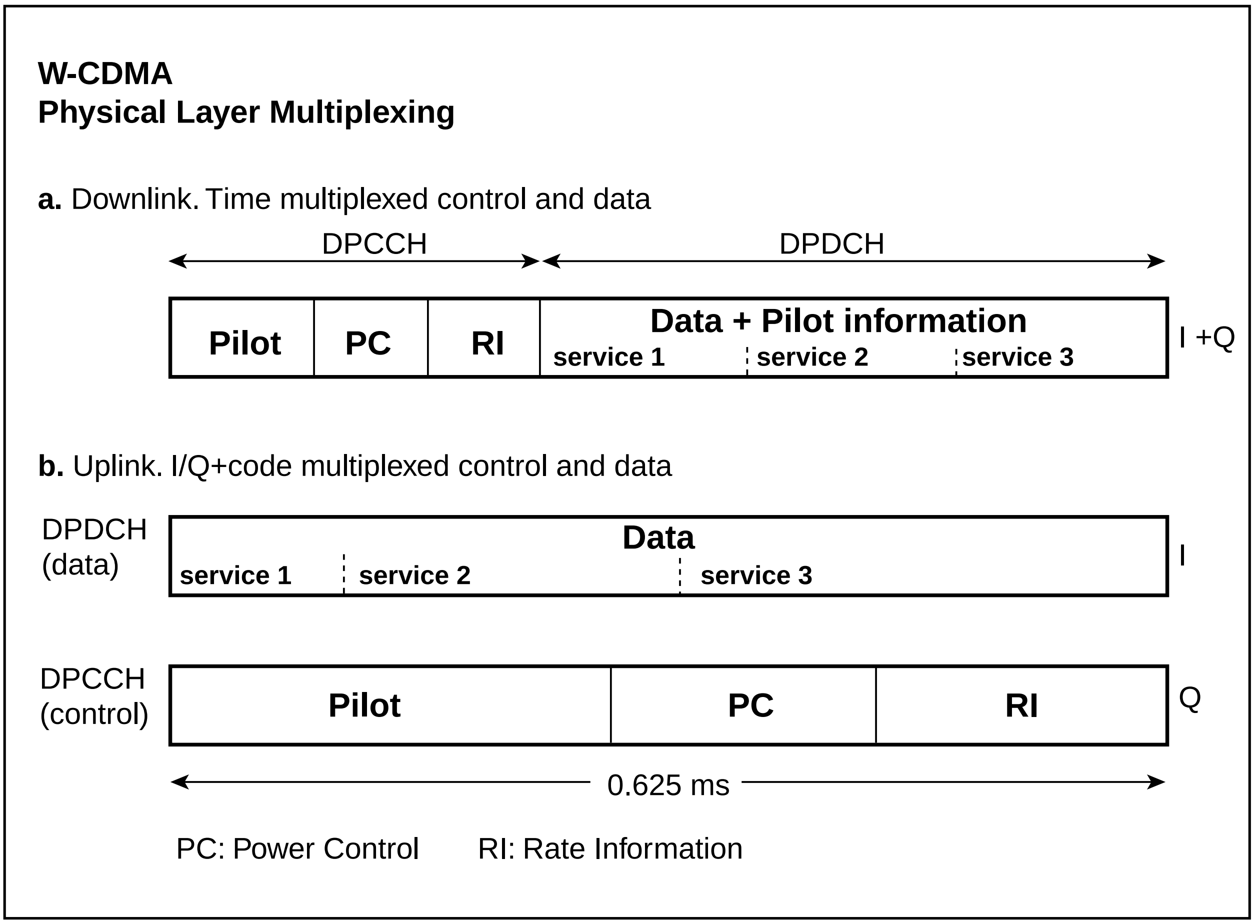


Figure 7.6

In the uplink, separate data and control channels are used, isolated by I/Q and code multiplex. See figure 7.6b. Constant power is used for the control channel, as it is used in connection with the power control at the base.

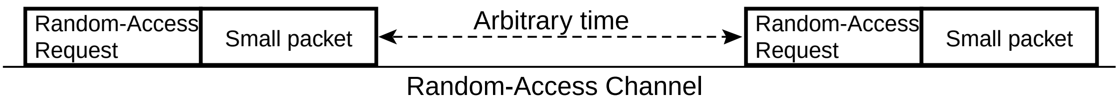
Packet access

For short message service, terminals can add on messages on the paging packages on the random access channel. For large or frequent messages, a dedicated channel will be reserved. See figure 7.7. The reservation might be delayed, if the system is already fully loaded.

Packet Access

Dual mode scheme with adaptive mode selection
based on packet-traffic characteristics

Small infrequent packets appended to Random-Access Request



Large or frequent packets transmitted on dedicated channel

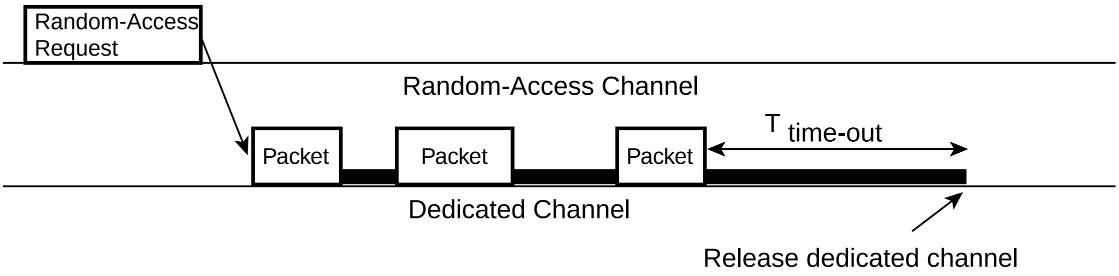


Figure 7.7

Interfrequency handover

The main use is in connection with hierachical cell structures with differerent frequencies for different layers. Interfrequency handover is also needed in connection with hot-spots, which are allocated more frequencies than near-by cells.

The same arrangement as in CODIT is used. Hot spots are handled by furnishing adjoining cells, using a smaller number of channels, with pilot transmitters for the not-used channels. This gives a terminal in a hot-spot cell information about channels available for handover, when it comes into the coverage area of another cell. It can than change channel to one that is used by that cell to prepare for a normal soft handover.

Title:

Consensus Decision on the UTRA Concept to Be Refined by ETSI SMG2

Consensus Agreement

This consensus agreement contains the key elements and advantages of WCDMA and TD/CDMA, and contains the following elements:

1. In the paired band, we adopt the radio access technique proposed by the ETSI Alpha group, that is WCDMA (Wideband Code Division Multiple Access), Tdoc SMG 903/97

2. In the unpaired band, we adopt the radio access technique proposed by the ETSI Delta group, that is TD/CDMA (Time Division/Code Division Multiple Access), Tdoc SMG 897/97

3. In the process of selecting the technical parameters the following shall be the objectives:
   Low cost terminal
   Harmonization with GSM
   FDD/TDD dual mode operation
   Fit into 2 * 5 MHz spectrum allocation

Note:

NTT DoCoMo participated as an observer in SMG#24bis and expressed support for ETSI SMG decision

Figure 7.8

7.4 Teledesic

The landmobile satellite systems discussed in section 4.4 cannot comply with the increased demand for wideband services, which can be handled by UMTS. Therefore several suggestions have been made for a new generation of satellite networks for wideband access. The most advanced of these is Teledesic, which already has been given frequency allocations in parts of the world. The plans are to start commercial service 2002. The main purpose of Teledesic is to service fixed terminals The service to portable terminals is secondary, and might not be economic viable (extremely complicated phase-array terminal antenna).

The general concept of Teledesic is somewhat similar to Iridium, but with much higher traffic capacity. There will be two mode of operations: to fixed terminals and to portable terminals. The original plan was to establish a network of 840 active satellites (+spares) placed in 21 orbits with a height of 705 km. The orbit period will be about the same as for Iridium. The number of satellites in the final proposal has been reduced to 288 using 12 orbit planes.

The links between satellites and terminals will use frequencies around 29 GHz and 19 GHz band. A 500 MHz spectrum has already been assigned in the US and the same allocation will probably be obtained in other regions. The Teledesic system will constitute a complete telecommunication network “in the sky“ comprising both the transport network with switches and the access network to fixed and mobile terminals. Most of the capacity will be used for the wide band fixed access, with terminals with high-gain antennas (35 dB).

The basic mobile service is based on a traffic module that will offer 16 kb/s speech (+2 kb/s control channel) and data services with similar bandwidth requirements. Packet transmission will be used. The net data rate will depend on how much channel coding that must be added. If higher capacity is needed several traffic modules can be used in parallell.

The antennas for the portable terminal must be phase-controlled, as a gain of 7dB is specified. Due to the high frequency, the size of such an antenna will be only about 5x5 cm. The difference between 29 dB and 7 dB is partly compensated for by 16 times (12 dB) lower system data rate (receiver bandwidth). The remaining difference of 10 dB means reduced propagation margin. The service quality with portable terminals will therefore be fairly marginal. Free sight towards the satellite would be necessary. A possibility is macro diversity, as with a network with around 300 satellites at least two satellites are well above the horizon at all times.

One 198 MHz radio channel for fixed service has a capacity of 1440 channels with 16 kb/s capacity. A 12.4 MHz channel for mobile service carries 90 channels.

The system parameters mentioned above will probably be modified during the final system design.

8. W-LAN, W-ATM

W-LAN

Local Area Networks (i.e Ethernet) are used to connect PC:s and computer-controlled equipment with central control or servers. In the future many new types of office and domestic equipment will be computer-controlled including a need to be connected to central computer systems. It will be a large demand for wireless connections either due to problems to arrange fixed wiring or the need to move equipment around. In a few years time, standards will cover systems with user rates up to 20 Mb/s. A similar application is to interconnect clusters of PC in the same room.

Hiperlan 2 (High-Performance Radio LAN)

The most interesting proposal, which is related both to W-LAN and W-ATM, is **HIPERLAN 2** which is under standardization by ETSI in cooperation with ATM forum (ATM with 25 Mb/s data rate will be used for the transport layer). A 100 MHz spectum around 5.2 GHz has already been reserved for this type of service nearly world-wide. (Also frequency bands at 17 GHz and above have been discussed.) The 5.2 GHz spectrum will be divided into four radio channels with 25 MHz width, which will permit about 25 Mb/s system data rate. 100 MHz is not enough for frequency reuse, as a cluster size larger than 4 would be needed. There are therefore plans to allocate an additional spectrum of 100 MHz. (Another possibility is to introduce time syncronization between near-by systems, so that they can be allocated different time slots in a TDMA structure.)

The range will be limited to about 50 m. The 5.2 GHz band might not be allowed for outdoor usage due to risk to interfere with satellite services, using this band.

TDMA/TDD with asymmetric connections will be used. Reservation protocols will be used to assign channel capacity. A promising modulation method (not for MA) seems to be OFDM, as this will give good frequency diversity and above all simplify the design of the channel equalizers. The other studied alternative is one-carrier 4 or 16QAM with about 25 MHz modulation bandwidth.

Wireless ATM

A parallell study program with somewhat wider scope is Wireless ATM. It is related to activities at ATM forum and ETSI (BRAN). W-ATM shall operate in connection with INTERNET/INTRANET. The objective is to offer up to 20 Mb/s user data rate as for Hiperlan 2, but also portable terminals indoors and in outdoor micro cells will be included. A total spectrum of about 500 MHz seems necessary when the system has been fully developed, but concrete proposals for spectrums assignments are not yet available.

A motivation for using the ATM protocols the whole way out to the terminals is that many users of the fixed ATM network might want to have simular QoS characteristics also when using wireless terminals.

9. Equipment technology

9.1. Wide band transmitters and receivers

The rapid development in the areas of digital VLSI and digital signal processing will make it more and more advantageous to implement most functions in transmitters and receivers with digital signal processing. This will have a considerable impact on the design of future systems.

Block diagram based on direct conversion

A suitable starting point for a discussion of these possibilities is the block diagram in figure 9.1. On the transmitter side, the modulator generates the radio symbols as I and Q baseband signals (complex baseband I_b+jQ_b) through digital signal processing followed by D/A-converters. These baseband signals are then directly mixed up to the final transmitter frequency. Independent upper and lower side band can be generated by the I-Q arrangement, which in principle can generate all possible signal waveforms.

The receiver can be a homodyne, i.e. the wanted input signal is directly mixed down to base band I and Q signals, which after suitable amplification and anti-aliasing filtering are fed to the A/D converters. The remaining receiver functions are performed by digital signal processing.

The main advantage of using direct conversion between the base band and the radio band is much reduced spurious problems and therefore no need for high-stability, high-Q circuits. If it would be possible to implement perfect I-Q mixers for the frequency conversion, the upper and lower sidebands would be completely orthogonal.

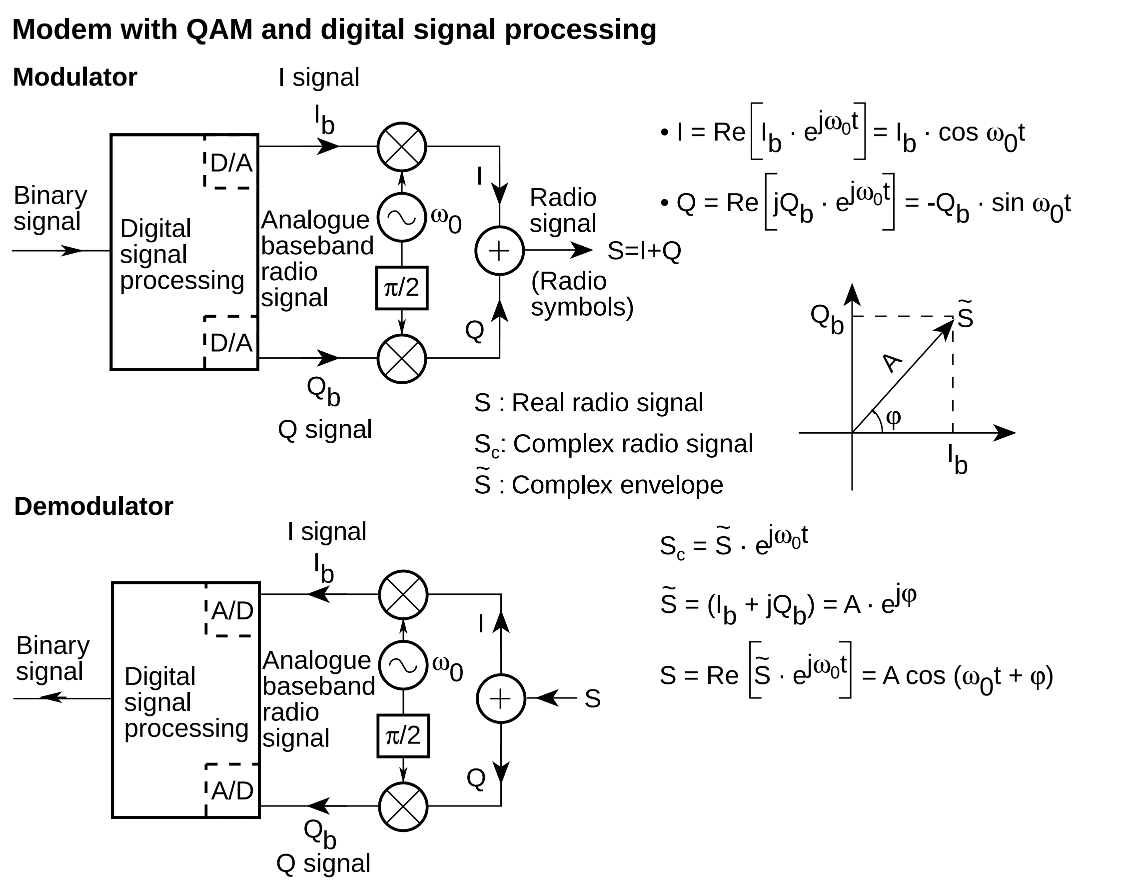


Figure 9.1

Block diagram with IF stages

However, due to practical implementation problems of analog I-Q mixers, it is impossible to obtain perfect isolation between the upper and lower sidebands. Therefore, a strong interfering signal on one of the sidebands could seriously degrade the reception of a weak signal on the opposite site of the carrier (LO frequency). On the transmitter side, a spurious would be generated on the opposite side of the LO frequency.

Another problem is that analog mixers cannot be perfectly balanced. Therefore, the transmitter I-Q mixer generates carrier leakage and the receiver mixer a DC voltage in the output.

These limitations make it impossible to use direct conversion for application with very high requirements on system selectivity. Instead, only one of the sidebands is used. For instance, on the receive side the input signal to the A/D converter is a low IF signal, comprising only one sideband. The unwanted sideband is stopped in the anti-aliasing filter before the A/D converter. On the transmitter side, the output from the D/A converter is an IF signal.

Wide-band transmitters and receivers

A further development of these transmitter and receiver block diagrams is to introduce common wideband stages for several modulated carriers (radio channels). The major new feature is that **wideband digital signal processing** is used for the multiplexing and demultiplexing of these carriers. In principle, the whole duplex band, allocated to an operator, is handled by one transmitter and one receiver structure.

On the transmitter side, a digital base band signal (I+Q) is generated, which consists of several modulated carriers. The whole package of analog carriers is generated in the D/A-converter and after upconversion amplified in a highly linear wideband output amplifier. As discussed above, either direct conversion of a multi-carrier signal comprising two sidebands (complex baseband) is used or alternatively a single-sideband multi-carrier signal at IF is taken out from the D/A converter.

The receiver front end is also wide band and has enough linearity and dynamic range so that the whole packet of wanted carriers can be fed to the A/D converter without any distorsion. However, the linearity requirements on the amplifiers and the A/D converter are extreme, as even small non-linearities can seriously degrade the receiver selectivity (spurious and intermodulation). The demultiplexing into individual channels, the channel filtering and the detection of the carriers are implemented through digital signal processing. See figure 9.2. (Wideband transmitters and receivers of this type are discussed in module RT1E.)

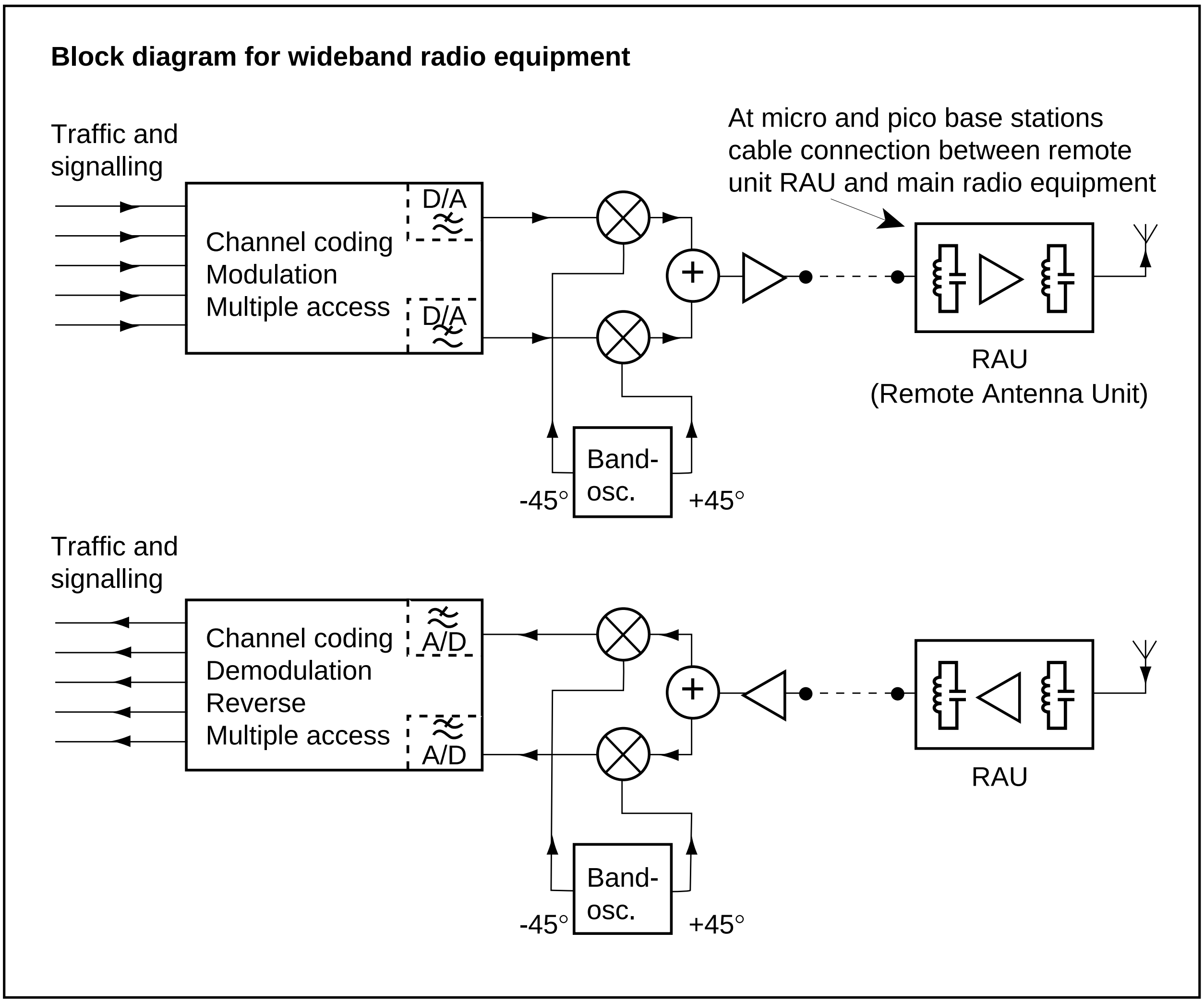


Figure 9.2

Alternatively, the A/D and D/A interface is on a very low IF, as has been discussed above. See figure 9.3.

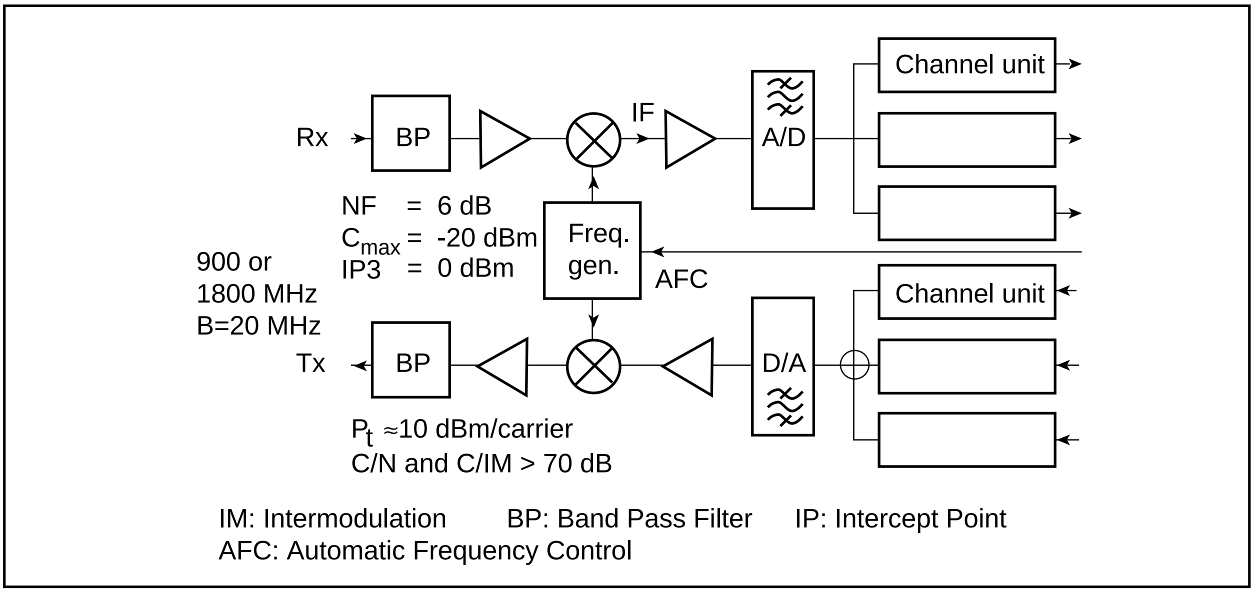


Figure 9.3

Implementation problems

The major design problems are related to the **extreme requirements on linearity and dynamic range** especially in connection with systems that are based on narrowband multiple-access arrangements (i.e. GSM and D-AMPS).

On the **transmitter** side, the most difficult functions to implement are the D/A converter and the wide band power amplifier. Different linearizing arrangements have been tried in connection with the power amplifier to obtain both sufficient linearity and reasonable efficiency.

On the **receiver** side, no channel filters can be introduced before the A/D converter. Therefore, the receiver stages from the input to the A/D-converter must have sufficient dynamic range and linearity to amplify simultaniously weak and strong carriers, without any generation of spurious and intermodulation products, which would degrade the receiver selectivity.

The advantages and the implementation problems in connection with wide band transmitters and receivers are summarized in Fig. 9.4.

Problems with wideband transmitters and receivers
(for handling of several narrow-band signals)

1. Transmitter noise and intermodulation

    High peak factor of composite transmitted signal

    No help of filter selectivity to attenuate output spectrum

Therefore:

Difficult tradeoff between linearity and efficiency

2. Receiver blocking and intermodulation performance

    No help of AGC
    No help of filter selectivity

Therefore:

    Extreme requirements on dynamic range up to and including A/D converter

Advantages with wide-band transmitters and receivers

- Wide-band transmitter amplifiers and receiver frontends are transparent, i.e. can be used with any type ot Multipel Access and modulation
- Common modem and channelizing unit implemented by DSP with software control
- Major part ot transmitter and receiver equipment common to many traffic channels (radio connections)
- Improved flexibility through:
    adaptivity
    gradual generation changes

Figure 9.4

New transmitter and receiver systems

As indicated in figure 9.2, the transmitter and receiver functions close to the antenna can be connected to the rest of the transmitter and receiver over a **cable connection**. This might be an attractive solution in connection with micro and pico cells, where a cluster of those cells can be fed from a common base station site (i.e. co-located with a macro cell base station) by cables of reasonable lengths. See figure 9.5. The signals over the cables can be on a suitable IF or consist of optical signals. Another possibility is to place the A/D and D/A interfaces at the antenna and transmit digital signals over the cable.

A similar pico cell arrangement is shown in figure 9.6. In this case, a system with cluster size one is assumed. Exactly the same radio signals are transmitted from the base stations, which have been selected to establish a diversity connection with a certain terminal. The signals are added in receiver input in the same way as over different propagation paths from one base station. The sum signal is typically Rayleigh distributed. If the modulation bandwidth is wide enough (compared to the correlation bandwidth) macro and micro diversity through the equalizer is possible (multi-path diversity).

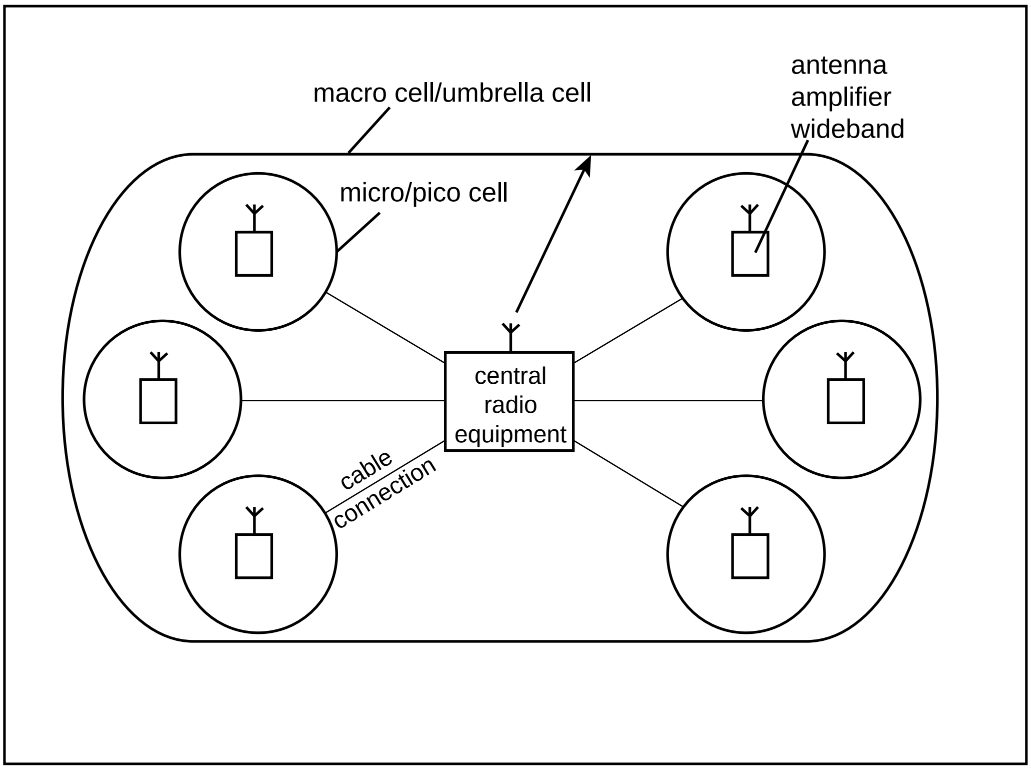


Figure 9.5

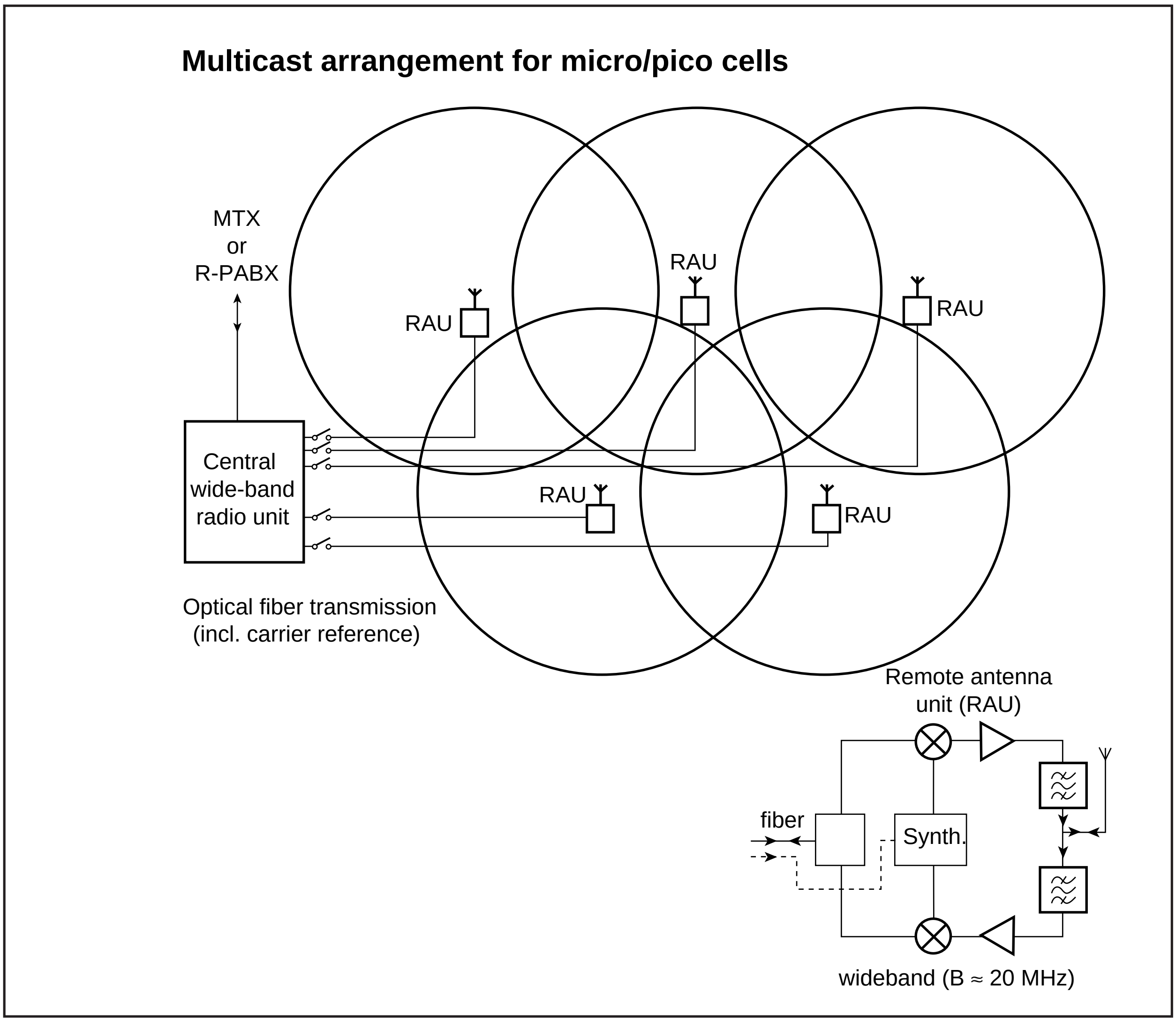


Figure 9.6

The wideband transmitter and receiver structures are also suitable in connection with phase-controlled base station antennas. See next section.

9.2 Advanced base station antennas

Increased number of lobes, diversity

The traditional base station antenna arrangement consists of three **120º sector antennas**, which cover one cell each (three cells share a common site). The maximum antenna gain is determined by the smallest usable vertical lobe width. This width cannot be too small, limitations are the swinging of the mast at strong winds and also that a certain vertical angle shall be covered. The maximum antenna gain for such sector antennas is around 20 dB.

Sometimes, the vertical lobe maximum is pointed below the horizon line (**tilt-down**) in order to better match the wanted vertical coverage. An additional advantage by pointing the antenna below the horizon might be added suppression of cochannel interferences from other cells and of multipath components (time dispersion) origina-ting from strong distant reflections.

Antenna diversity comprising two receive antennas with 5-10 meters horisontal separation is sometimes used. This helps to decrease unbalance in the link budgets for the two directions (the terminal transmitter power is typically much smaller than the power to the base station antenna). **Diversity with 4 channels** can be obtained by designing each antenna with outputs for horisontal and vertical polarization. (Even if only one polarization is transmitted, the radio wave contains about the same power in the horisontal and vertical polarization after a few reflections. Also, the two polarizations fade roughly independent of each other.)

The next development step is to introduce an arrangement with many fixed antenna lobes for the receiver antenna, but to keep the traditional sector antenna for the transmitter. The receiver antenna gain is typically increased from 20dB up to 24 dB. This reduces further the unbalance between the inward and outward directions.

Phase-controlled antennas

The next step, which still is in the R&D phase, is to introduce advanced phase-controlled base station antennas which could give much higher gain. The antenna structure comprises several elements, which are fed with adjustable mutual phase. By suitable setting of the phases, it is possible to change the lobe direction nearly instantaniously, or even to form several lobes. The phase shifters used to be analog and placed close to the antenna elements. In the future the phases will be controlled by digital signal processing. The phase control can also set a suitable direction of the main lobe and in addition a number of lobe minima (nulls) in selected directions.

The most suitable antenna lobe pattern is a compromise between three desirable features (see figure 9.7):

a. The largest possible antenna gain. This reduces the requirements on transmit power, and helps to reduce the average level of cochannel interference.

b. The antenna diagram ought to have lobe minima in the direction towards the stongest cochannel interferers. This can give a considerable reduction of the typical C/I, which helps to reduce the average cluster size and thus improve the frequency economy, especially at interference-limited systems.

c. The antenna diagram ought to have lobe minima in the direction towards strong multi-path reflections with excessive delay (that falls outside of the equalizer window).

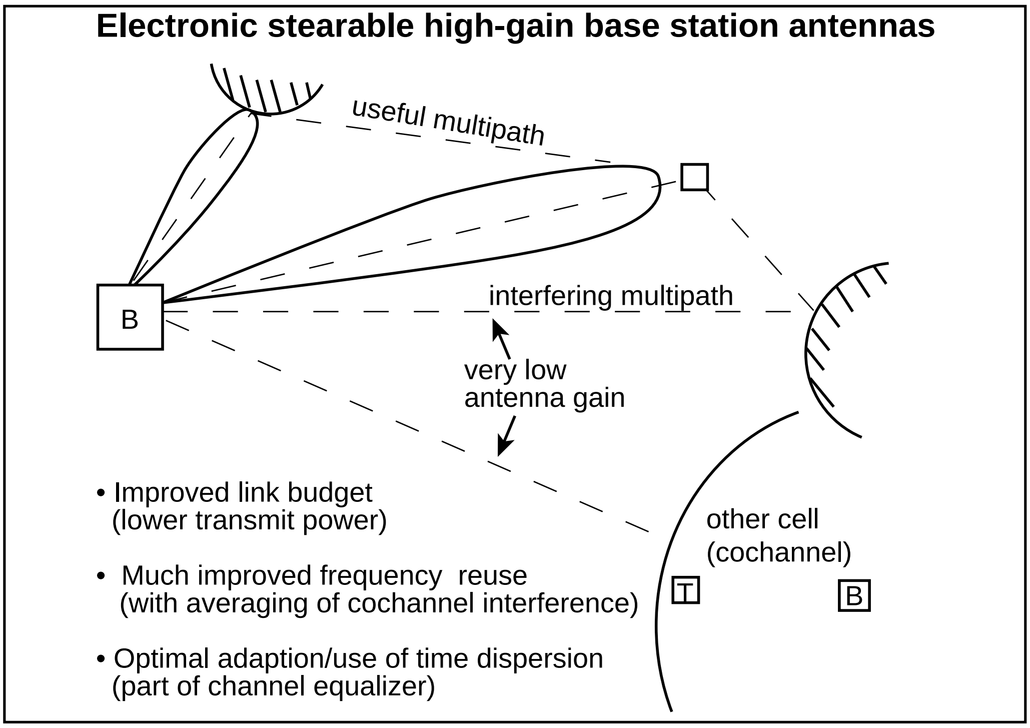


Figure 9.7

The antenna arrangement can be considered as an important part of the total digital signal processing to adapt the receiver in an optimum way to the total radio conditions (wanted signal, noise and interference). The system is jointly optimized with respect to the spatial and time-dispersion characteristics of the wanted and the interfering signals. See figure 9.8. The function of the analog phase shifters can be transferred to the DSP subsystem, by introducing a multichannel arrangement connecting each antenna element to the DSP subsystem with A/D and D/A functions.

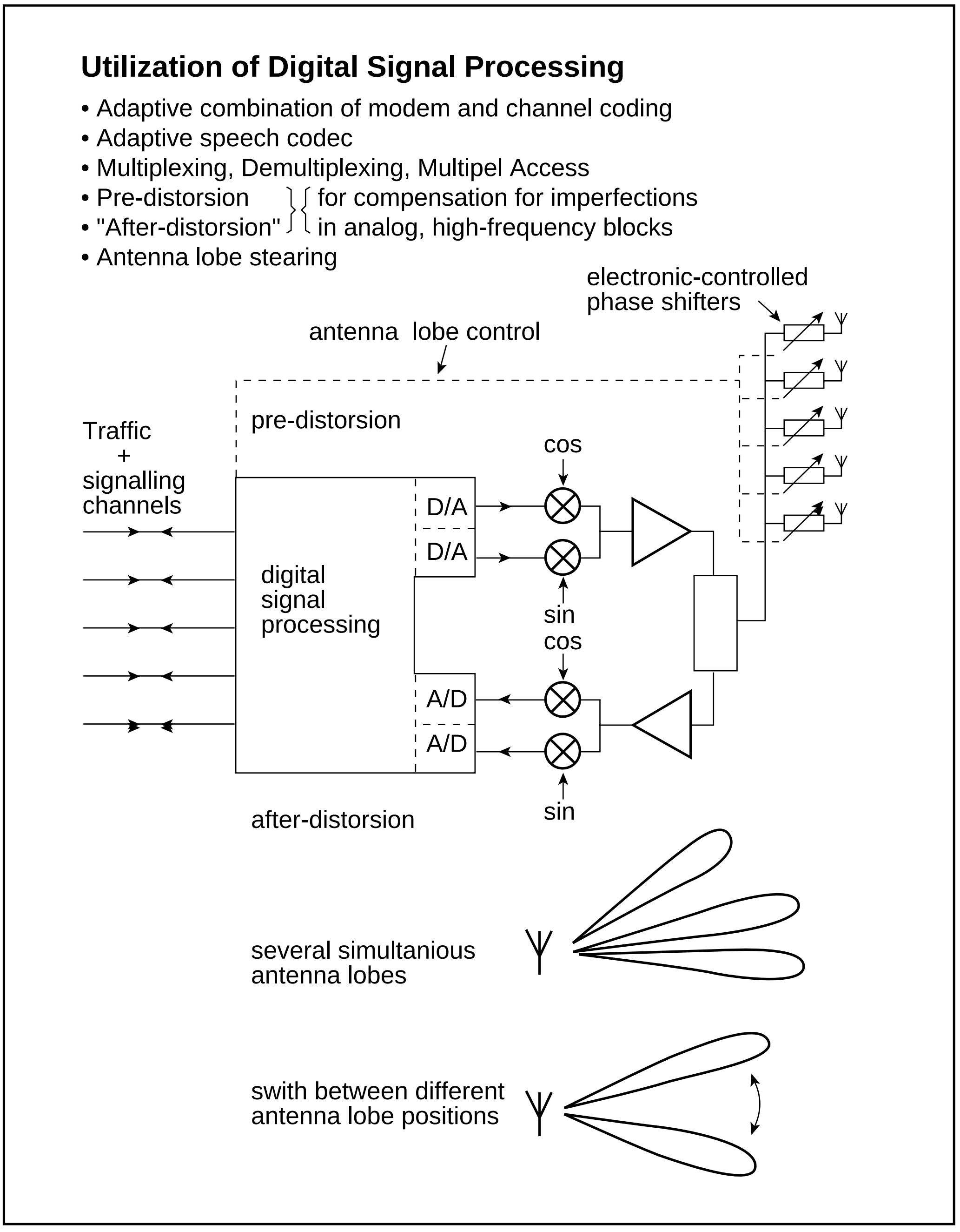


Figure 9.8

If the antenna consists of N elements in the horizontal direction, it is said to have N-1 degrees of freedom. One degree can be used to set the directions of the main lob and the other degrees are used to set N-2 nulls.

Measurements indicate that in a typical propagation case with no direct path the multipath situation generally consists of reflections from houses and other obstacles in the direct vicinity of the terminal Seen from the base station, the incoming signal has an angle spread of a few degrees.

The receiver signal processing analyzes the spatial characteristic of the received signal and optimizes the antenna pattern for reception. The signal processing must be able to discriminate between the wanted signal and the interference. One possibility is to use the code marking of the signals at DS-CDMA. Another possibility is to mark the signals with different pilot signals.

The adjustment of the transmitter antenna pattern must also be based on the analysis made by the receiver signal processing. An additional problem at systems using frequency duplex is that the multi-path fading is uncorrelated in the outward and reverse directions. However, the transmitter pattern can be optimized with respect to the local average of the received signals.

Appendix 1: Overview of IS-95 (DS-CDMA)

1. Introduction

A mobile telephone system based on DS-CDMA has been developed by Qualcom in USA. The result was the standard IS-95, which is an alternative to the TDMA based standard IS-54. FCC allocated an experimental frequency band of 2 x 1.25 MHz within the AMPS band for the original test system, and the commercial system is still based on this channel spacing.

The standardization activities started early 92 in the subcommittee TR45, formed by the Cellular and Common Carrier Radio Section of the Telecommunication Industries Association, TIA. Qualcom published a detailed description of the original system in connection with the standardization activities. This overview is based on this early information.

Features, which contribute to the good frequency economy are:

a. A speech coder with variable data rate in combination with low rate FEC

b. Advanced dynamic control of the terminal transmitter power

c. Macro diversity with soft handover and micro diversity
   Micro diversity is obtained by a combination of channel coding with interleaving and frequency diversity through the rake receiver. The gain by frequency diversity is however fairly small, as the 1.25 MHz channel bandwidth is of the same order as the correlation bandwidth of the propagation channel, especially in micro and pico cells.

d. Statistical averaging of interference from many cochannel transmitters.

a. Speech coding, channel coding

The rate of the speech coder can be switched between 1.2 kb/s, 2.4 kb/s, 4.8 kb/s and 9.6 kb/s. The lowest data rate is used in speech pauses (comfort noise). The output consists of 20 ms blocks, which are sent to a convolutional coder. After that interleaving is introduced, also over 20 ms. (The original speech coder gave too bad speech quality and has been replaced by an improved version with higher data rate).

In the **outward direction, channel coding with** the rate of 1/2 is used, i.e. with the highest rate from the speech coder 19.2 kb/s is fed to the transmitter modulator. It is suitable to feed a continuous signal with this data rate to the CDMA modem also in time intervals, when the data rate from the speech coder is below 9.6 kb/s. The blocks from the channel coder are therefore repeated a suitable number of times. (As a higher processing gain is obtained at the lower rates from the speech coder, the transmit power is also reduced correspondingly).

In the **inward direction** channel coding with rate 1/3 is used, i.e. 28.8 kb/s is fed from the channel coder at the highest data rate from the speech coder. In the same way as in the other direction, interleaving and repetitions are introduced. Thus a continuous 28.8 kb/s signal is fed to the CDMA modem. In this case, the matching of the transmitter power to the speech coder rate is made by pulsing the transmitter. A continuous signal is transmitted at the highest data rate from the speech coder, for the lowest data rate the duty factor is 1/8. (The motivation for this arrangement is that the efficiency of the power amplifier drops, if it is not operating at full output power. Also a transmitter peak power, which is independant of the varying source data rate simplifies the power control arrangements).

b. Control of the transmitter output power

Accurate, fast power control is above all necessary in inward direction, but a rough power control is used also in the outward direction. In the **inward direction,** the main purpose of the power control is to accomplish that signals from all terminals, connected to the base, at all times are received with nearly exactly the same level (if the power adjustments to account for the different source data rates are neglected). The power levels at the input to the base receiver must be kept within ± 1 dB of the nominal value, as otherwise the power unbalance would lead to a considerable reduction in the traffic capacity.

The shadow fading and the distance dependence of the global mean are completely correlated between the two transmission directions. Therefore the terminal can compensate for those variations in the propagation attenuation on the basis of measurements of the local mean of the received signal (actually a pilot tone transmitted with constant power from the base).

The fading around the local mean due to multi path propagation is uncorrelated between the inward and outward directions, which use different frequencies. Due to the band spreading to 1.25 MHz, the amplitude fading is typically half of what corresponds to narrow band Rayleigh fading. The remaining variations are however so large that they must be suppressed by very fast power control (closed loop). The compensation is made by power regulation commands from the base to the terminals every millisecond.

The adjustment of the terminal transmit power is thus based on a two step control. An open-loop control takes care of the distance dependence and the shadow fading. Data messages from the base inform the terminal of the transmitted pilot power, and the terminal measures the short term average of the received pilot level. From this information is determined a suitable basic value (short term average) of the transmitted power. This power control is fairly slow.

In addition a fine tuning of the output power is made to compensate for the multipath fading, through a fast control loop. A data link from the base to the terminal commands fine adjustment of the output power, so that the remaining signal fluctuations at the input to the base receiver are eliminated. The control information is transmitted as delta modulation, multiplexed in the convolutional channel coder with a signalling or traffic channel. Due to the time delay in the control loop, it cannot fully compensate for the multi-path fading from high-velocity terminals. On the other hand, channel coding supported by interleaving is more effective at high speed.

In the outward direction, the same type of open loop control is used as in the inward direction. The local mean of the propagation attenuation is estimated for the different terminals. From this information the base can distribute the available transmit power in an optimum way between the different connections.

c. Diversity and soft handover

Micro diversity and macro diversity with soft handover is utilized. It is mainly based on signal processing in the rake receiver arrangement.

Macro diversity in the **outward direction** is prepared by the terminals, which search for the strongest pilot channels from the nearby cells and sends information about their levels to the MTX. In this way a connectivity list can be prepared. When the signal quality of the current base-to-terminal link becomes marginal, the MTX establishes macro diversity by engaging also the base station, which is going to take over the call if a handover should become necessary. (Sometimes the same source signal is sent out from two additional base stations, establishing triple diversity.) Simultaniously the terminal is informed, so that branches in the rake arrangement can be allocated to receive signals also from the new base station, thus establishing soft handover. The corresponding arrangement is used for hand over between sectors serviced by the same base station (this is called softer handover in order to differ from handover between base stations, which is called soft handover).

In the base station receiver a similar rake receiver is used only for micro diversity. Macro diversity in the **inward direction** is arranged by having two or more base stations detect the signal from the terminal. The detected baseband signals are fed to the MTX. Selection diversity is implemented by selecting the version of the current 20 ms speech segment that has best quality. (An important limitation is that MAHO in combination with macrodiversity using soft handover is possible only within the radio channel beeing used - not to other radio channels that might be available in the sys-tem.)

2. The CDMA-modem

2.1. System components

Different multiple access and modulation arrangements are used in the inward and the outward direction. However common features are:

a. A long maximum-length shift register sequence

b. A short pn-sequence, which is close to maximum length

c. A Walsh-codec based on 64 orthogonal Walsh functions of 64 chips

d. A common system time reference for all base stations and thus also for connected terminals

e. CDMA-baserad isolation against interference from other cells

f. Rake-receiver arrangement.

a. The long pn-sequence has a period of $2^{42}-1$ chips and is generated with a chip data rate of 1.2288 megachips/s. Through decimation, the chipdata rate can be reduced, if needed. From this basic sequence individualized pn-sequences can be generated in two ways:

- The relative timing with respect to the common time reference according to item d below can be varied
- The pn-sequence can be modified by adding a sequence determined by a key.

b. A pair of short pn-sequences are generated by adding an extra 0-chip per period to two different maximum-length sequences having a length 2^{15}-1, i.e. the short pn-sequences have both a period of 2^{15} = 32768 chips. They are balanced, i.e. comprises equal numbers of 0 and 1 chips. The corresponding period will be 26.666 ms.
The same sequences pair is used in all the terminal units (but with different relative time positions in different cells). The pn-sequence pair is utilized to code the two local oscillators, used for the generation of the QAM or OQAM radio signals, whereby one sequence is used for the I channel and the other for the Q channel. The short pn sequence is used in the pilot signal transmitted from base station. The sequence is therefore also known as the pilot-channel sequence.

c. Walsh group. The Walsh functions are bipolar binary sequences with the characteristic that they are mutually orthogonal, if suitably time syncronized. In this case a 64-group of Walsh functions, with each function comprising 64 chips. Some of the functions are shown in Fig. A1-1.
The Walsh functions are used in the terminal modulator (**inward direction**) as an M = 64 symbol alphabet, i.e. each symbol carries 6 information bits. In the **outward direction** the Walsh functions are used for multiplexing, so that up to 64 mutually orthogonal channels can be transmitted from the base transmitter. These are used for pilot channel and synchronisation channel, paging channel and traffic channels.

d. Common system-time reference. This is definied through the use of the satellite based Global Positioning System (GPS). Every site is equipped with a GPS receiver. The specific system time is different for all nearby base stations but related to the common system time. Each base station transmits its specific system time to the terminals.

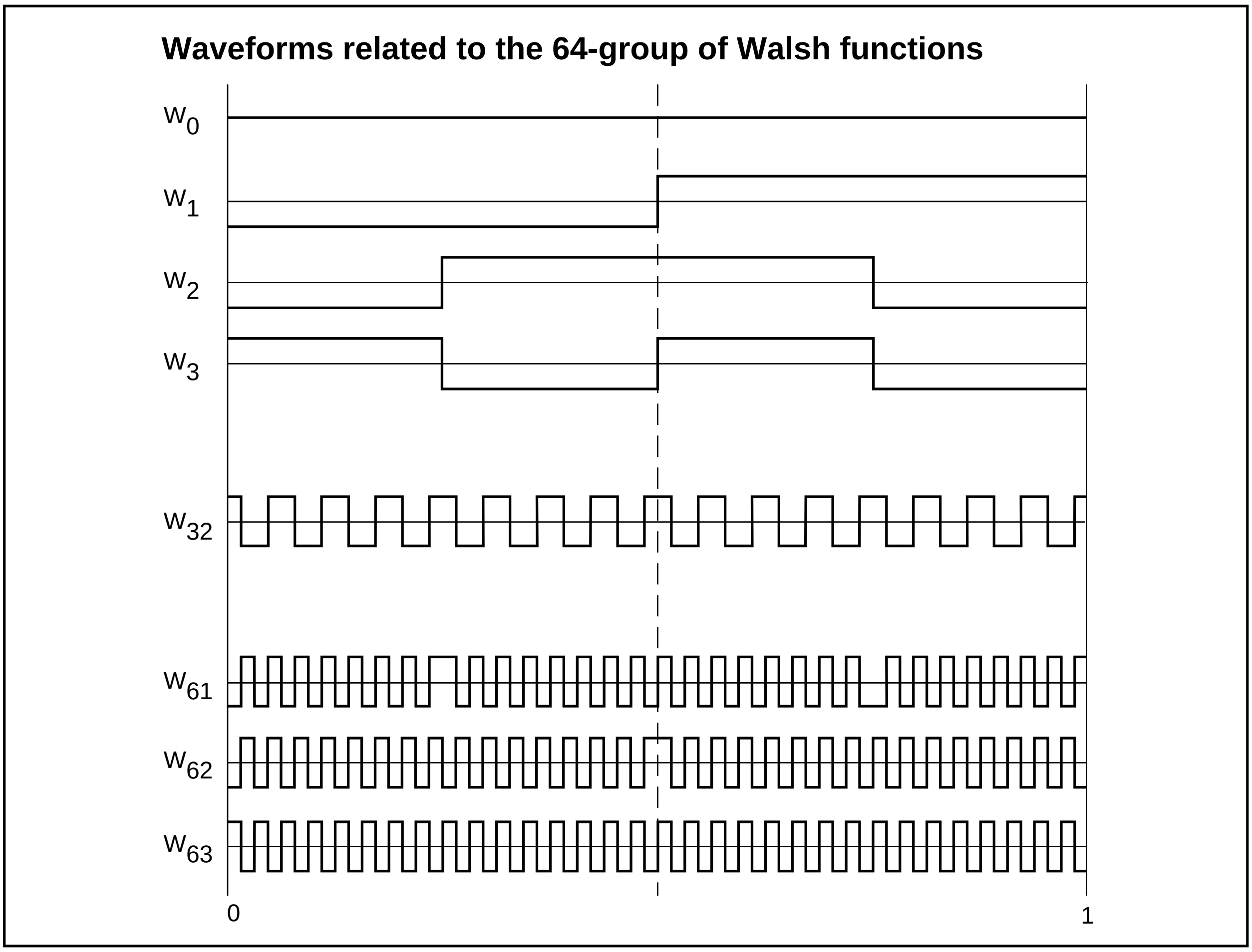


Figure A1-1

e. CDMA-based isolation against interference from other cells and also isolation between different connections in the inward direction within each cell. Thus quasi-orthogonal connections are obtained, whose mutual isolation is determined by the anti-jamming margin AJ. The connections in the outward direction from each base station are in principle orthogonal, as Walsh functions are used for the multiplexing (the orthogonality breaks partly down, if the propagation channel has strong time dispersion). In the base station transmitter are used used two modulation or coding stages in tandem, which are controlled by the short pn sequence and Walsh-functions respectively. The pn sequence gives isolation between cells, the Walsh function provides isolation between the outgoing connections from the same base.

The short pn-sequence is thus used for mutual isolation between different cells. Over one period of the sequence, a number of different time positions are defined (different for different sites), whose mutual time difference exceeds the maximum length of the impulse response of the propagation channel. Connections within each cell are isolated from each other by using different codes derived from the short pn-sequence for spreading and despreading. All the units within one cell use the same timing of the basic pn-sequence. The terminals belonging to the cell are informed over the pilot channel about this timing. Information to the terminals about the corresponding timing for adjacent cells is sent out over the syncronisation channel.

f. The rake-receiver arrangement. The discrimination between signals from different cells and between signal components with different propagation delay is through the rake arrangement. Discrimination is obtained by having different codes in the channels of the rake arrangment. The codes consist of different versions (different relative time position) of the short pn-sequence. See Fig. A1-2.

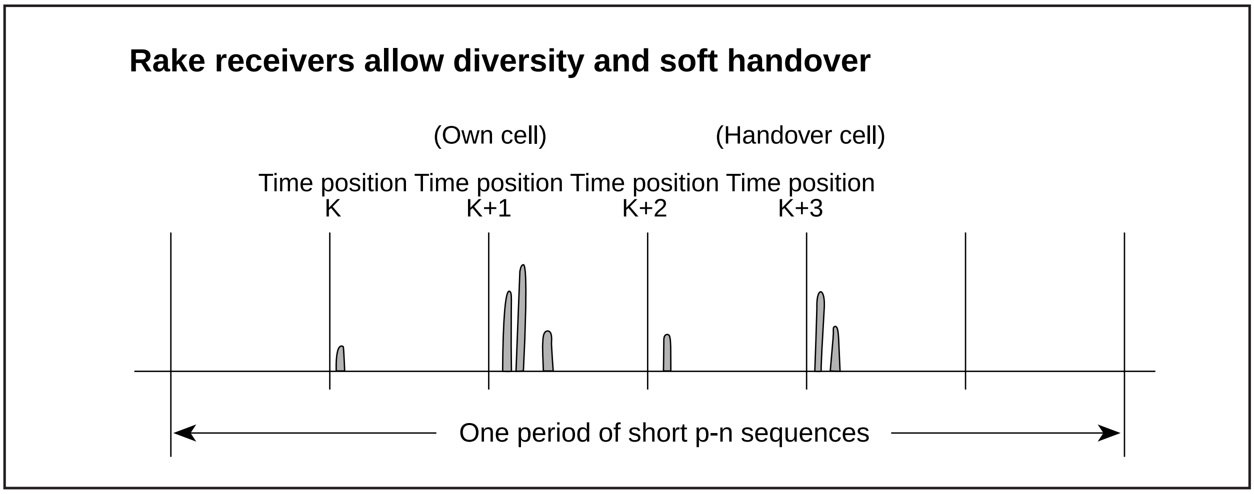


Figure A1-2

According to the figure, there is a rough structure, in as much as the radio connections in different cells differ with respect to the relative time position of the short pn sequence. Surrounding each time position is a fine structure, since the received signal from each base station is composed of components having different propagation delays.

A branch of the rake receiver of a **terminal** therefore has to be set approximately to receive signals from the wanted cell, and then fine tuned to receive a signal component having a given propagation delay. For macro diversity, the diversity branches in the rake arrangement are set for the time positions corresponding to two or more cells.

Output power is matched to the data rate from the speech coder by pulsing the **terminal** transmitter to a suitable duty cycle (e.g. 1/8 when 1.2 kb/s is output from the speech coder). The band spreading, corresponding to a chip data rate of 1.2288 Mchips/s, is achieved by combination of the long and the short pn-sequences, which both have this chip data rate. CDMA isolation of simultaneous connections within a cell is based on the use of different versions of the long pn-sequence (via different key settings).

In principle, the **base-station** receiver carries out the inverse of the signal processing done by the terminal transmitter. Microdiversity is obtained by a rake arrangement similar to that used in the terminal receiver. The symbol detection takes place in a Walsh decoder. The signals from the 64 decoders in each of the diversity branches are added. Thereafter a decision circuit selects the strongest of the 64 outputs, and a group of 6 bits is fed out.

The base receiver arrangement can be improved by antenna diversity and an increased number of diversity branches in the rake arrangement.

3. Traffic capacity

According to Qualcom, the theoretic capacity per cell in a homogeneous cell structure (cluster size one) is obtained from the following equation for the number, N, of simultaneous calls per cell:

$$N = \frac{B}{d_i} \cdot \left(\frac{E_b}{N_0}\right)^{-1}_{min} \cdot G_{DSI} \cdot L_{cell} + 1 \approx 40$$

In this equation, the following numeric values shall be put in (according to Qualcom):

$B = 1.25$ MHz	Available channel bandwidth for each transmission direction
$d_i = 9.6$ kb/s	Maximal data rate from speech coder
$^B/d_i = 130$ gr	Processing gain (exkl. DSI-gain)
$(E_b/N_o)_{min} = 7$ dB	(optimistic value)
$G_{DSI} = 2.5$ ggr	DSI gain due to variabel speech coder rate
$L_{cell} = 0.7$	Capacity loss due to interference from other cells

(The practical capacity figure is 3 to 4 times less due to several degradation factors).

Appendix 2. FH-CDMA based on GSM

1. Overview of the radio system

In connection with the discussion in the U.S. about suitable systems for the PCS band, ERA proposed an advanced GSM system as an alternative to IS-95. This system was an interference-limited version of GSM (FH-CDMA). Several comparations were made of the relative performance of these two systems. The key features were the frequency economy (traffic capacity per cell for a certain spectrum allocation) and the noise limited sensitivity which had an influence on the maximum cell sizes in rural areas. This appendix presents the results from the extensive ERA simulations of the frequency-economy of a FH-CDMA (i.e. interference-limited) version of GSM.

The system is based on the same TDMA structure and modulation as GSM. The width of the frequency slots is 200 kHz. The hopping rate is one hop per TDMA frame, i.e. 217 hops per second. The same speech coder as for GSM is assumed, and also the same arrangements for channel coding and interleaving.

The advantage of FH-CDMA compared with the normal GSM arrangement is better optimization of the different means to improve the frequency economy, which is related to the advanced frequency-hopping arrangement of GSM. In principle, the hopping structure uses the whole bandwidth, allocated to an operator (typical 5 MHz or 15 MHz). It is arranged in a way that gives cyclic, coordinated frequency hopping within each cell but a random hopping structure between different cells. This means that connections within each cell are orthogonal so that cochannel interference is only generated from connections in other cells. This interference is spread out evenly over the used frequency band. This has already been discussed in section 6.7.

The draw-back with random hop structure is the collisions between the connections at different cells, which is the cause of the cochannel interference. However, better averaging of cochannel interference is obtained and also a more random time distribution of the interference. DTX and power control increase the average value of C/I. The C/I distribution can be further improved by bandwidth expansion so that the radio channels are not loaded to 100%. A typical case is a system load of 2/3, corresponding to a spreading factor of 1.5. Thus, the system becomes interference limited.

2. Radio performance

The receiver sensitivity is often specified for FER = 2%. Frame Erasure Rate (FER) is related to the channel coding arrangement, which besides FEC includes error detection of those bits from the speech coder, which give especially large quality reduction when they are wrong. If the error detection coding indicates that these sensitive bits have been hit by uncorrectable transmission errors, a complete 20 ms speech segment is discarded (frame erasure). A FER of 2% corresponds roughly to 6% ber into the channel decoder.

Frequency hopping is an option at GSM. **Without frequency hopping,** the receiver sensitivity depends considerably on the terminal speed, as has been discussed above. The required protection ratio for FER = 2% therefore depends on the specified propagation model, incl. the terminal speed. If the quasi-stationary case is excluded, the protection ratio is typically 9 - 11 dB.

With frequency hopping and with the TU (Typical Urban) propagation model, a protection ratio of about 9 dB (local average over the fast fading) is required for all terminal speeds. This allows the cluster size 9 without the use of antenna or macro diversity.

Simulations have been made, which give the relation between the protection ratio $(C/I)_{min}$ and the FER with and without **antenna diversity**. See figure A2-1. Two different values of the diversity gain have been assumed, corresponding to uncorrelated fading for the two antennas (ρ=0) and correlated fading with a correlation factor ρ of 0.7. For the later, more typical case a diversity gain of 4 dB is obtained. The fading margin in respect of the fast fading can therefore be reduced by 4 dB through antenna diversity (further improvements could give up to 6 dB gain from base antenna diversity). This corresponds to a required protection ratio of less than 5 dB.

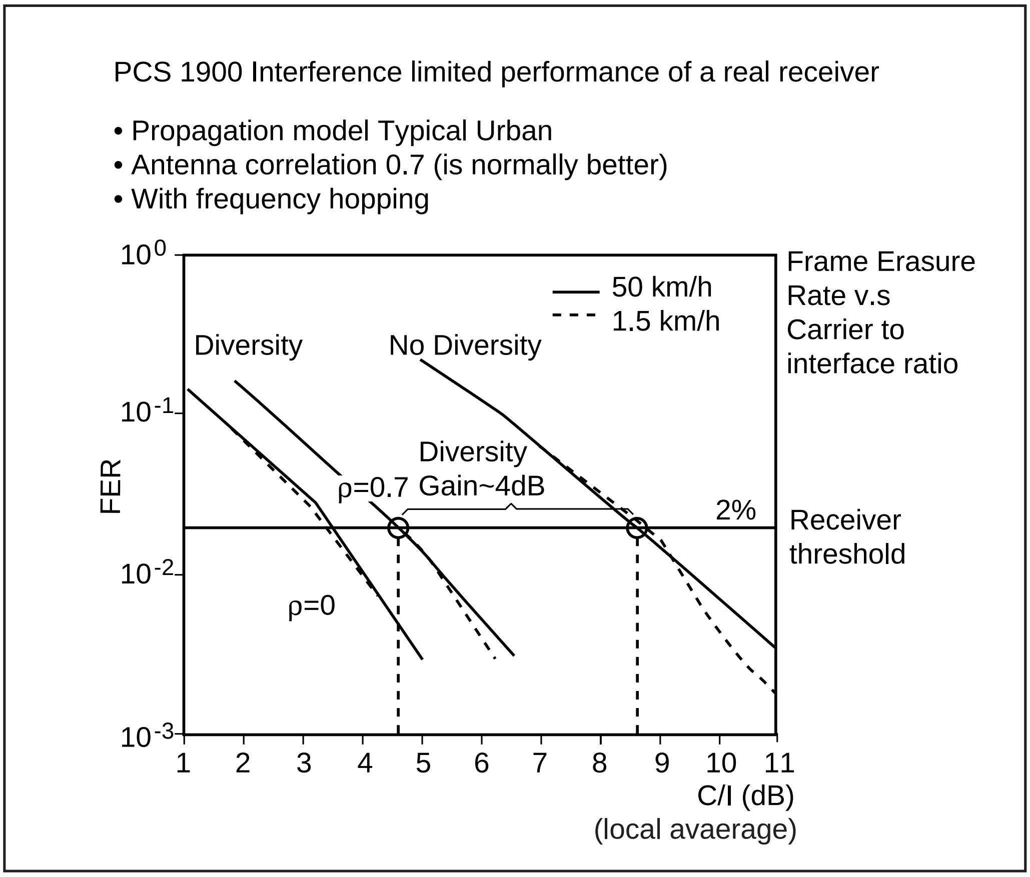


Figure A2-1

Simulation of Interference Diversity Gain

C/I target for Frame Erasure Rate (FER) decreases with random frequency hopping because of time distribution of interference

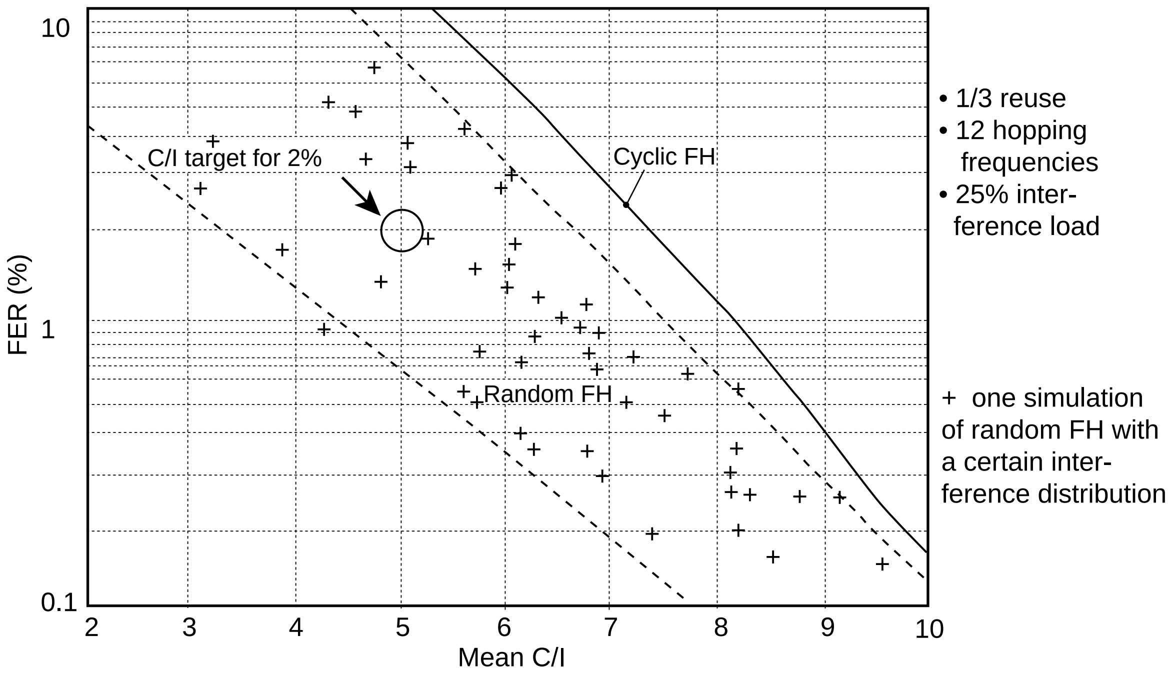


Figure A2-2

The random frequency hopping, DTX and power control together with a fairly small amount of band spreading (2/3 system load) give enough improvement of the C/I distribution that the cluster size can be reduced to 3. (It is assumed that it is enough with 90% service area reliability.) Also efficient handover algoritms give fairly good macro diversity, which reduces the shadow fading margin to 5 dB.)

As the C/I-distribution and thus the relation between reliability and protection ratio depends on very complicated statistical relations, the results above have been determined by simulations. See figure A2-2.

If we disregard the overhead for signalling etc., a system with 2x15 MHz bandwidth has room for 15x5 = 75 duplex channels. With a cluster size of 3 and 2/3 system load, the soft capacity limit becomes:

$^{75}/_{3} \times {}^{2}/_{3} = 16.7$ radio channels per cell.

With full rate traffic channels, the number of traffic channels per cell will be:

8x16.7 = 133.

The corresponding normalized capacity becomes 8.7 channels per cell and MHz.

The corresponding capacity for the original GSM system with cluster size 9 is 75/9 = 8.3 radio channels per cell or 67 full-rate traffic channels.

These simulations indicate that improved diversity arrangements and use of FH-CDMA instead of FH-TDMA would permit a doubling of the number of channels per cell. (Improved diversity arrangements but still using FH-TDMA would permit a cluster size of 7.)

Appendix 3. Overview of CODIT

CODIT (Code Division Testbed) is a large development project, which has partly been financed by EC as part of the RACE program. The program started in 1992 and ended in middle 95. Several firms and institutions have participated: ERA, Telia Research, Philips, British Telecom and IBM. The activities have consisted both of detailed system studies and of design and measurements of a 2.2 GHz testbed, which has been evaluated in Kista. The objective of the project was to assess the potential of DS-CDMA for UMTS.

One of the requirements of UMTS is to make possible a universal system solution through suitable cooperation and coordination between different independent operators, which differ with respect to coverage areas (partly corresponding to different layers of the hierarchical cell structure) and the offered range of services, especially with respect to bandwidth. This overall concept means that mutual syncronization between different base stations often is not possible. Also, within an overall DS-CDMA concept there shall exist versions with different radio frequency bandwidths. The reason is the need to match the bandwidth allocation of different classes of operators, and also that systems for narrower radio channels will be less expensive to implement.

Some of the key DC-CDMA concepts from IS-95 have been used, but several improvements have been introduced, i.e. several optional channel widths, handover capability also within hierarchical structures, possibility to utilize packet transmission efficiently, and no need for mutual syncronization between base stations (which is required for IS-95). The testbed only works for channel widths up to 5 MHz, but the system studies also cover 20 MHz radio bandwidth.

An advantage with a large radio bandwidth is better frequency diversity for a given coherence bandwidth of the propagation channel. The largest bandwidth (20 MHz) has the advantage that it gives sufficient band spreading also for 2 Mb/s services. As mentioned above, the services with the highest data rates should only be used in micro and pico cells. The following coupling has been assumed between source and chip data rates:

source data rate	**chip data rate**
kb/s	Mchips/s
0.4 - 16	1 5 20
128	5 20
2000	20

Figure A3-1

As is used at IS-95, a pilot channel is used in the outward direction, in order to allow coherent reception at the terminal receivers and measurement of the impulse response of the propagation channel. This information is used to set optimal timing delays and phases for the branches of the rake receiver. CODIT also uses a pilot channel in inward direction in order to permit coherent base station receivers.

The power control arrangement is about the same as used for IS-95. An other similarity is that both long and short spreading sequencies are used. Different versions of the long spreading sequence, with differ with respect to the relative delay, are used for multiple access between the different traffic channels and most of the signalling channels. The exception is that two short spreading sequencies of different length are used for the RACH (Random Access Channel for paging from terminal), the syncronization and pilot channel (Physical Control CHhannel, PCCH).

Instead of the requirements of mutual syncronization of nearby base stations (which is required at IS-95 to make possible macro diversity with soft handover in outward direction) the only requirements of mutual time syncronization at CODIT applies to the diversity channels via different base stations.

The contact between base station and terminals for signalling and system control is through the continuous PCCH. The traffic and signalling channels are structured in 10 ms segments. For each segment the PCCH sends information about source data rate, which determines the processing gain and thus normalized transmit power. In the outward direction, the PCCH is also used for the dynamic power control (adaption to the fast fading). The output power for a connection depends both on the source rate and the propagation attenuation. An example of variable source data rate is the used speech coder. Another case is packet transmission, where there generally are idle intervals between the packages.

Different radio channels shall be used for the different hierachical levels, in order to make possible selection of the optimum hierachical level, irrespective of the relative signal strengths. Time compression during the handover interval is used to implement macro diversity with soft handover between different hierarchical levels, which use different radio channels. During a 10 ms segment, 5 ms is used for transmission over one radio channel and 5 ms for transmission over the other. See Fig. A3-2. As mentioned above, macro diversity in the outward direction has the drawback of additional loading of the allocated radio spectrum, but this is more than compensated for by the decrease of the shadow fading margin.

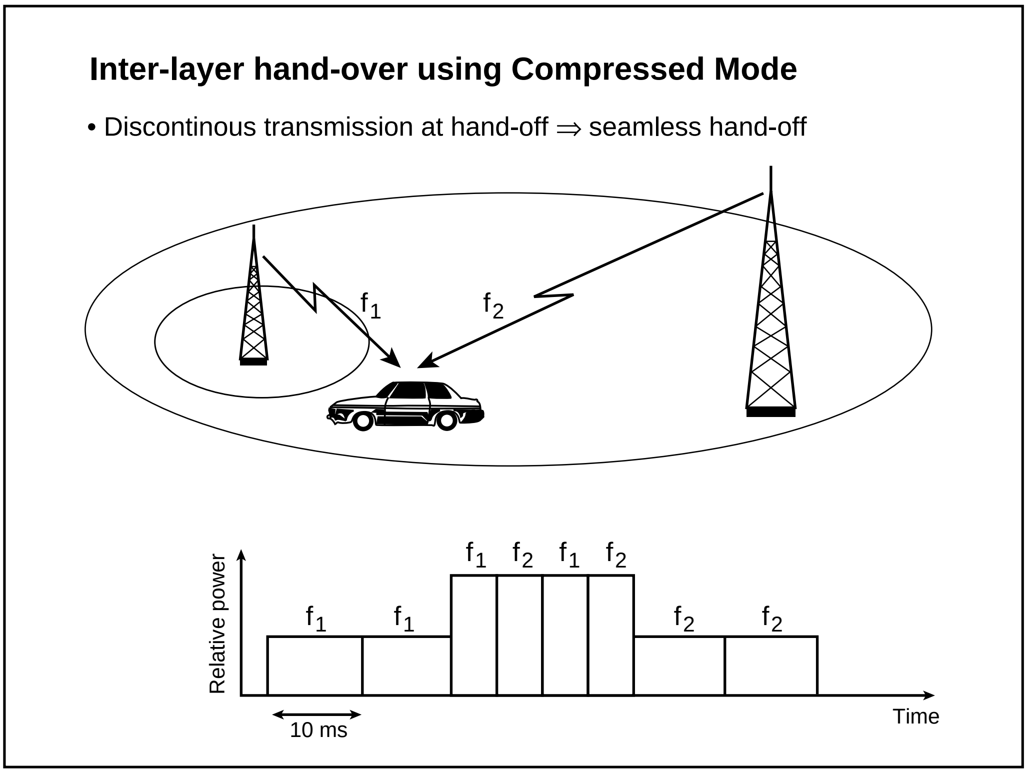


Figure A3-2

An overview of the radio specification for CODIT is shown in Fig. A3-3.

CODIT. Overview of radio part

Several optional channel bandwidths:

(1 - 20 MHz)

(20 MHz only for micro and pico cells)

Traffic channel rate up to 2 Mb/s

Data rate for the speech coder: 0.4 - 16 kb/s

(average rate 8 kb/s)

Packet transmission

No need for mutual time syncronization between base stations

Macro diversity with soft handover

(requires time syncronization between diversity branches

(handover also between hierarchical layers)

Frequency diversity

(if sufficient channel bandwidth)

Figure A3-3

Appendix 4. Overview of ATDMA

ATDMA (Advanced TDMA) is a parallel RACE projekt to CODIT. The experiences from CODIT and ATDMA will be used by the ETSI study group, which is preparing an European proposal for UTMS. The results from the ATDMA project might also be used in connection with the further development of GSM, i.e. packet transmission. The project started in 1992 and ended in 95. Several firms have participated, such as Nokia, Siemens and Alcatel.

Considerable improvements have been introduced in comparison with traditional TDMA-systems such as used for GSM:

- adaption to the new requirements for UMTS in respect of several different types of services and of hierarchical cell structures by dynamic adaption of the modulation and the channel width and utilization of packet transmission. See below.
- dynamic channel allocation (DCA), which both improves the frequency economy and eliminates the need for frequency-cell planning.
- macro diversity with soft handover, i.e. parallel connections from two or more base stations to a terminal are established by the use of different time slots in the TDMA frame or corresponding packet arrangement. The procedure is compatible with a hierachical cell structure.

Different types of modulation and several channel bandwidths and also different number of time slots per frame are utilized for the different hierarchical cell classes: large macro cells, normal macro cells (umbrella cells), micro cells and pico cells. The large macro cells are used in rural areas with no frequency shortage. For these are used GMSK, which gives good efficiency for the transmitter output stage. More frequency economic OQAM is used for the other cell sizes.

For the two types of macro cells are used a channel spacing of 277 kHz, which permits a symbol data rate of 360 kilobaud with GMSK and 450 kilobaud with OQAM.

For micro and pico cells is used the channel spacing 4 x 277 = 1108 kHz, which permits 1800 kbaud symbol data rate. In order to be able to offer data services up to 2 Mb/s, a multi carrier arrangement will be used, i.e. several radio channels can be allocated to a connection. Each time slot comprises 120 - 125 symbols, which means that the number of time slots per frame varies between 15 and 72 for the different cell sizes.

The TDMA structure has a high degree of flexibility, permitting statistical multiplex and packet transmission. Dependent on the transmission requirements of different services (speech, data transmission which tolerates only small transmission delays and data transmission with larger allowed maximum transmission delay) the TDMA structure is combined with different types of channel coding and interleaving. An additional transmission mode for data services, which are not sensitive to even large transmission delays, is selective ARQ. Asymmetric traffic channels can be set up, such as an one-way channel with only ARQ signalling in the other direction. The TDMA structure is shown in Fig. A4-1 and A4-2.

PRMA++ (Packet Reservation Multiple Access)

- Varying data rates by allocating different number of time slots per frame to different users
- The allocation is done on a short time basis and may vary during the call

A A B C A C A A B - A C A A B B A - A - B B A B A C B B A B

Figure A4-1

PRMA++ Frame structure (micro/pico)

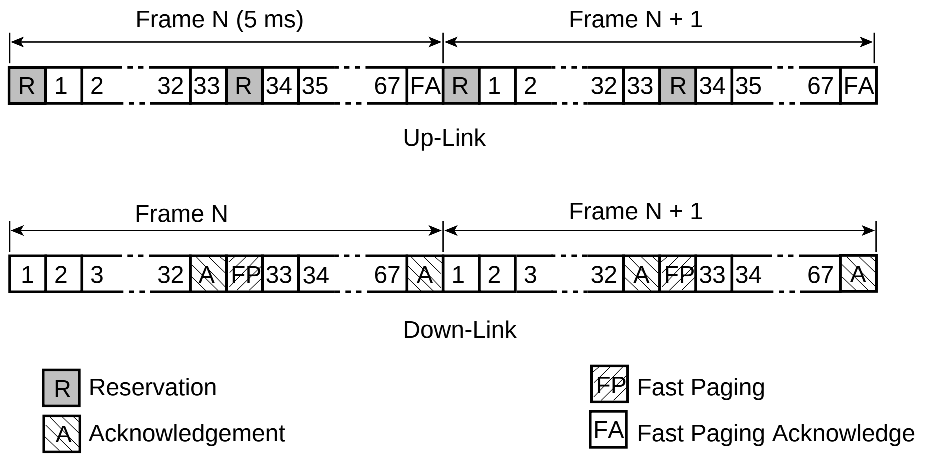


Figure A4-2

Statistical multiplex is based on demand assignment of time slots in the TDMA structure. The reservation is made on a short term basis. This means in principle that the reservation of time slots is released during speech pauses. Different logical channels are included in the time structure. Terminals ask for reservation of channel capacity in specially assigned time slots in the inward direction. There is a certain probability for collisions, i.e. it is a Aloha channel. If there is available channel capacity for traffic, the base sends back an acknowledgement with allocation of traffic time slots. Time slots are also assigned for other types of signalling including acknowledgement, e.g. for call set up (Fast Paging + Fast Paging Acknowledge). Remaining timeslots are reserved for traffic. The signalling protocols are in principle quite similar to the corresponding GSM protocols.

Appendix 5.
List of acronyms and abreviations

AAA	Adaptive Antenna Array
AAL	ATM Adaption Layer
ABR	Available Bit Rate
ACTS	Advanced Communication Technologies and Systems
AMPS	Advanced Mobile Phone Standard
ARIB	Association of Radio Industries and Business (Japan)
ASIC	Application-Specific Integrated Circuit
ARQ	Automatic Repeat Request
ATM	Asynchronous Transfer Mode
BEC	Backward Error Control (ARQ)
B-ISDN	Broadband ISDN
BRAN	Broadband Radio Access Networks (ETSI)
BSC	Base System Controller
BTS	Base Tranceiver System
CBR	Continuous Bit Rate
CDPD	Cellular Digital Packet Data (U.S.)
CLR	Cell Loss Ratio
CRC	Cyclic Redundancy Check
CS	Convergence Sublayer (part of AAL)
DAB	Digital Audio Broadcasting
DCA	Dynamic Channel Assignment
DECT	Digital Enhanced Cordless Telephone
DLC	Data Link Control
DLL	Data Link Layer
DSL	Digital Subscriber Line
EIA	Electronic Industries Association (U.S.)
ETSI	European Telecommunication Standards Institute
FCC	Federal Communication Commission (U.S.)
FDD	Frequency Division Duplex
FEC	Forward Error Correction
FFT	Fast Fourier Transform
FPLMTS	Future Public Land Mobile Telecommunication Systems
FSK	Frequency Shift Keying
GFSK	Gaussian FSK
GMSK	Gaussian Minimum Shift Keying
GPRS	General Packet Radio Service (GSM)
GRAN	Generic Radio Access Network (GSM)
HCS	Hierarchical Cell Structure
HIPERLAN	High-Performance Radio LAN (ETSI)

HLR	Home Location Register
HSCSD	High-Speed Circuit-Switched Data (GSM)
IBC	Integrated Broadband Communication
IFFT	Inverse FFT
IMT-2000	International Mobile Telecommunications 2000
IN	Intelligent Network
INAP	Intelligent Network Applications Part
INMARSAT	International Maritime Satellite Organization
IS-136	U.S. TDMA standard (D-AMPS) incl. digital system signalling
IS-95	U.S. DS-CDMA standard
ISDN	Integrated Services Digital Network
ISM	International, Scientific and Medical (frequency band)
ISP	Internet Service Provider
ISUP	ISDN User Part
IFHO	Inter-Frequency Handoff
IP	Internet Protocol
ITU	International Telecommunications Union
ITU-R	ITU Radiocommunication sector
ITU-T	ITU Telecommunication sector
LAN	Local Area Network
LEO	Low Earth Orbit
LLC	Logical Link Control
LMDS	Local Multipoint Distribution System
LR	Location Register
MAC	Medium Access Control
M-ATM	Mobile ATM
MBS	Mobile Broadband System
MSC	Mobile Switching Center
MSP	Mobile Switching Point
MSS	Mobile Satellite System
NNI	Network Node Interface
OFDM	Orthogonal Frequency Division Multiplex
OSS	Open System Interconnection
OVSF	Othogonal Variable Spreading Factor
PACS	Personal Access Communication Services
PBX	Private Branch Exchange
PCN	Personal Communication Network (Europe)
PCS	Personal Communication Services (U.S.)
PDC	Personal Digital Cellular (Japan)
PDU	Protocol (Payload) Data Unit
PHS	Personal Handy phone System (Japan)
PMP	Point to Multi Point

PMR	Private Mobile Radio
PNNI	Private NNI
POTS	Plain Old Telephone System
PRMA	Packet Reservation Multiple Access
PS	Personal Station
PSTN	Public Switched Telephone Network
QAM	Quadrature Amplitude Modulation
QoS	Quality of Service
RACE	R&D in Advanced Communication Techn. in Europe
RAL	Radio Access Layer
RLC	Radio Link Control
RLL	Radio Local Loop
RM	Radio Module
SAMBA	System for Advanced Mobile Broadband Applications
SAR	Segmentation and Reassembly Sublayer (part of AAl)
SCP	Service Control Point
SDP	Service Data Point
SDH	Synchronous Digital Hierarchy
SMG	Special Mobile Group (ETSI)
SSCOP	Service-Specific Connction-Oriented Protocol
TACS	Total Access Communication System (Europe)
TCP	Transmission Control Protocol
TCP/IP	Protocols for INTERNET
TDD	Time Division Duplex
TE	Terminal Equipment
TETRA	Trans European Trunked Radio System
TIA	Telecommunication Industry Association (U.S.)
UBR	Unspecified Bit Rate
UMTS	Universal Mobile Telecommunication System
UNI	User Network Interface
U-NII	Unlicenced National Information Infrastructure
UPT	Universal Personal Telecommunication
vBNS	Very high speed Backbone Network Service (U.S.)
VBR	Variable Bit Rate
VC	Virtual Circuit
WAN	Wide Area Network
WAND	Wireless ATM Network Demonstrator
WARC	World Administrative Radio Conference
WATM	Wireless ATM
WBA	Wireless Broadband Access
WCDMA	Wideband (Direct-Sequence) CDMA
WLL	Wireless Local Loop
WWLA	Wireless Wideband Local Access

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