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Networking, Protocols and Traffic Performance

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Bernhard H. Walke

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Mobile Radio Networks

Networking and Protocols

To

Antonie,
Thomas and Christoph

Mobile Radio Networks
Networking and Protocols

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Preface

Until the late 1980s cellular mobile radio networks for public and private users in Europe were company-specific solutions and not intended for the mass market. The broader technical–scientific world therefore limited its interest to familiarizing itself with the systems and their concepts, without involving itself in the details.

Since the development of European standards for digital systems in the late 1980s and the subsequent introduction of these systems around 1990, mobile radio has become a mass market commodity. Digital mobile radio has evolved from being an add-on business to being a key sales sector for certain large telecommunications companies, making them market leaders worldwide. This has resulted in the technical–scientific world taking a greater interest in mobile radio.

The success of mobile radio is due to the great advances made in information technology, such as the microminiaturization of integrated circuits and components and the dramatic increase in the integration density of semiconductor devices on chips, which have been particularly important in the development of hand-portable mobile radio devices (handhelds): a mobile terminal essentially consists of a very powerful signal processor that incorporates as programs all the algorithms of transmission technology required for receiving and transmitting and for electric signalling. ASICs are used instead to reduce the power consumption.

On the other hand, the advances in information technology are also evident in the development of these algorithms for signal (de)modulation, synchronization of communicating parties, channel coding and channel equalization, i.e., receiver technology that allows the reliable reception of signals with a few microvolts of amplitude over the radio channel, which perceptually can be described as an intermittent electrical contact.

Another just as important contribution to information technology has been the development of services and protocols for the organization and operation of communications networks. Along with multiplexing functions that enable a large number of mobile terminals to communicate quasi-simultaneously over the radio interface of the mobile radio system, these networks comprise a telecommunications network that contains intelligent network functions for mobility management, as well as cryptographic procedures for data protection and data security of a level never previously available in any network.

Thinking about the GSM (Global System for Mobile Communications: ETSI-Standard) which has been successful worldwide, one forgets that, in addition to this cellular system, many other concepts exist for new digital mobile radio systems that are attempting to repeat the success of GSM, with

some of them aiming for applications other than narrowband voice communication, e.g., paging, trunked radio, cordless communication, wireless local networks, wireless ATM broadband communication and satellite-supported personal communication. This book explores all these systems.

Since 1983, my research group has specialized in the development of services and protocols for private and public mobile radio systems, and has produced an extensive set of tools for software design, modelling and stochastic simulation of mobile radio systems. Through these tools, the mobile radio systems now being used in Europe, under discussion or in the process of being introduced, and described in this book, have been reproduced in a highly accurate form as large simulation program packages at my chair. These tools allow us to study the existing or forthcoming systems in their natural environments with the appropriate radio coverage, mobility and typical traffic volumes of their subscribers, and, based on this, to test our own approaches to the improvements and introduction of new services and protocols. Our proposals and the results of our work have successfully influenced the ongoing standardization at the European Telecommunciations Standardization Institute (ETSI).

The tools referred to are the outcome of the work of on average about 25 scientific assistants and 60 students working for their Master's theses per year, without whom it would not have been possible to incorporate so many details of so many systems. The work involved was not limited to the implementation of protocols for the respective systems, but ranged from the development of radio planning tools on the basis of empirical and ray-tracing techniques for given scenarios to Markov chain-based modelling of the radio channel, exemplary research into the modelling of receivers, studies on the effectiveness of adaptive channel coding, prototypic implementation of equalizers, development of models for bit error characteristics of different systems, development of procedures for dynamic channel allocation in large-scale systems and for the decentral organization of systems with wireless base stations, etc., all the way to the development of value-added services. This supplementary work proved to be necessary in order to establish with sufficient realism the difficult process of modelling real systems. It would not have been possible to present a description of the systems with the desired degree of detail without actually having implemented the services and protocols in realistic models for simulation of the systems.

The text has been gradually expanded from a first comprehensive presentation of GSM [1]. The text and many of the figures in this book are based on the input of many students whose names it would not be possible to mention individually. All I can do is convey my gratitude to all of them for their enthusiasm and for the thoroughness of their work in this collaboration. Their contribution was in modelling and evaluating the different systems and their modifications, and their input has helped my research assistants and me to develop a better understanding of the characteristics of the systems which have been considered.

The individual chapters of the book have been written in close cooperation with the research assistants responsible for the respective system models and they have been named. The chapters reflect the results of extensive research and development and in some cases incorporate material from final or earlier versions of eight cycles' worth of lecture notes. I should like to take this opportunity to give my warmest thanks to them for the thoroughness of their contributions on the respective topics, for their assistance in dealing with the relevant Master's theses that they have supervised and for their role in creating such an excellent work atmosphere. I should particularly like to mention Peter Decker and Christian Wietfeld, who in earlier years helped to provide some of the background information for the lecture notes and later provided the nucleus around which the book crystallized by integrating existing text modules. Contributions to individual sections of the book have been made by the following members of my research group:

- Branko Bjelajac (Chapter 14)
- Götz Brasche (Section 3.11)
- Peter Decker (Chapter 3)
- Matthias Fröhlich (Chapter 15)
- Alexander Guntsch (Chapter 14)
- Andreas Hettich (Chapter 12, Section 13.9)
- Martin Junius (Section 3.6)
- Arndt Kadelka (Chapter 5, 12)
- Matthias Lott (Chapter 5)
- Dietmar Petras (Section 2.8, Chapter 12)
- Christian Plenge (Chapter 9, Section 13.2)
- Markus Scheibenbogen (Section 3.6, Chapter 9)
- Peter Seidenberg (Chapter 5)
- Matthias Siebert (Chapter 11)
- Martin Steppler (Sections 2.8, 6.3)
- Christian Wietfeld (Chapter 3)
- Eckhardt Geulen (Section 3.10.5)

I especially want to thank my student Dirk Kuypers, who has shown so much dedication in the preparation of the manuscript. In addition to taking great care in incorporating years' worth of corrections and additions to the manuscript, he has made an effort to ensure that there is a homogeneity to the content and presentation—a task that particularly entailed extensive and frequent revisions to the tables and illustrative material. The precise correction work of Frank Mueller and Thomas Lammert and the careful reading of the final version of the manuscript by Carmelita Goerg are greatly appreciated.

I also want to convey my warm thanks to Mrs Jourdan von Schmoeger for the careful translation performed on the basis of a German version of the book into English. This work contributed very much to establish this book, although a lot remained to be done by the author, especially to translate the text in the figures, identify the misunderstandings introduced and align some wordings to the technical terms used by mobile radio specialists. I sincerely hope that I have not corrupted too much the style of presentation introduced by the professional translator.

Aachen, December 1998 Bernhard Walke

Reference

[1] B. Walke. *Technik des Mobilfunks, in: Zellularer Mobilfunk, J. Kruse (Hrsg)*, pp. 17–63. net-Buch, Telekommunikation edition, 1990.

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1
Introduction

During the first half of this century, the transmission of human voice through the telephone was the dominant means of communication next to telegraphy. Radio-supported mobile communication has constantly grown in importance during the last few decades and particularly the last few years to technical advances in transmission and switching technology as well as in microelectronics. Table 1.1 presents an overview of the chronological development of mobile radio systems.

In contrast to wireline networks, mobile radio networks that comply with the wish for geographically unrestricted communication can be used anywhere where it is not economic or possible to install cabling. Whereas the limiting factor with wireline networks is the network infrastructure that has to be created, the capacity of radio networks is determined by the frequency spectrum available and the physical attributes of radio waves in the earth's atmosphere.

The development of radio systems is influenced considerably by the scarcity of an important resource—frequencies. For instance, spectral efficiency can be improved through the digitalization of speech and the use of source and channel coding. Existing analogue radio systems are therefore being replaced more and more by digital mobile radio networks. Modern digital techniques used in modulation, coding and equalization enable bandwidth-efficient transmission and offer better interference behaviour and lower susceptibility to noise than analogue-modulated signals. Digital voice and data can be processed and stored before being transmitted.

This allows the use of multiplexing methods such as TDM (*Time-Division Multiplexing*), FDM (*Frequency-Division Multiplexing*) and CDM (*Code-Division Multiplexing*) that enable services to be provided to many users. For example, with TDM a large number of users in a specific frequency bandwidth are able to exchange information without extremely high selectivity of the receiver. This means that fewer steep-edged filters and resonating elements are needed, thereby resulting in a cost reduction, whereas modems transmit in burst mode and therefore are more costly. Digital modulation techniques often produce a higher level of transmission quality and are also more compatible with existing digital fixed networks.

Mobile communication today is available from a broad spectrum of technological and service-specific forms. The aim of this book is to provide the reader with an overview of the digital communications networks that have been introduced over the last few years, along with the services these networks offer

Table 1.1: Chronological development of mobile radio systems

Year	Paging system standards	Cordless phone system standards	Mobile terrestrial system standards	Mobile satellite system standards
1980	POCSAG	CT0	NMT 450 Nordic Mobile Telephone	Inmarsat-A
1985		CT1	AMPS (USA) Advanced Mobile Phone System RC2000(F) Radio Communication C450 (D,P) Cellular TACS (UK) Total Access Communication System	Inmarsat-C
1990		CT2	GSM Global System for Mobile Communication DCS 1800 Digital Cellular System at 1800 MHz	Inmarsat-B Inmarsat-M Inmarsat-Paging
1994	ERMES		TFTS Terrestrial Flight Telephone System	
1998				Inmarsat-P21, Iridium, Aries, Odyssey, Globalstar, Ellipso
2000			UMTS Universal Mobile Telecommunication System FPLMTS Future Public Land Mobile Telecommunication System	
2000+			Mobile Broadband System/ Wireless ATM	

and their protocols. Special emphasis is given to systems in Europe that are currently being used, are being standardized or whose introduction is imminent.

Deregulation and liberalization of the telecommunications market, along with the various agreements on standardization, are having a major effect on the development of mobile radio systems. Detailed specifications are necessary in order to achieve compatibility between the products of different system and terminal suppliers. International and European standards bodies are defining mobile radio systems that can be used and operated across country boundaries. This will enable users to be reachable wherever they are roaming and will result in the cost-effective production of terminals per unit, thereby opening up the market to different types of customers. The most important standards organizations active in the mobile radio area are covered in Appendix B.

Physical connections over a radio channel are far more complex than those in a fixed network. Some of the main characteristics of radio transmission are therefore presented in Chapter 2.

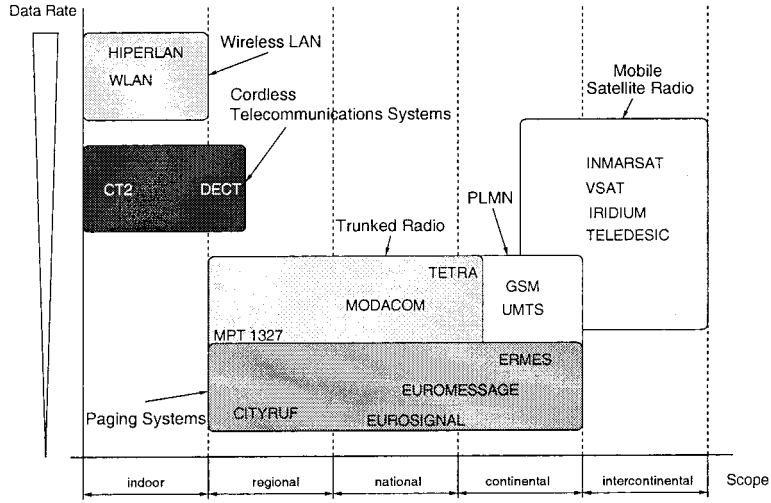


Figure 1.1: Different types of mobile radio systems

Digital radio networks that currently exist, are at the planning stage or in the process of being installed are described in the chapters that follow. Networks are differentiated according to the services they offer their users, their technical design and how they are used (see Figure 1.1). Significant differences exist between them in how and to what extent they support user mobility (see Figure 12.1).

Public cellular mobile radio systems extend the telephone service of wireline networks area-wide to mobile users. Older mobile radio networks were designed for pure voice transmission and use analogue transmission techniques. In 1958 the first public mobile radio network, the A-Netz, was installed in Germany and subsequently followed by the cellular B and C networks (see Figure 1.2). Whereas the B-Netz is restricted to telephony, the C-Netz also offers data services. These networks are currently being replaced by digital mobile radio networks based on the ETSI/GSM standard. Other analogue networks in Europe (see Table 1.2 and Figure 1.3) likewise are currently being replaced by GSM.

Trunked mobile radio systems are optimized for commercial application and provide services like closed user group, e.g., for companies with mobile field staff. These systems enable voice and/or data communication over a half-duplex channel with a central site, typically between mobile users in an area geographically restricted owing to the transmitter power of the base station; see Chapter 6.

Paging systems allow the direct paging of subscribers with mobile, pocket-sized receivers through the transmission of a signal or short message. Users with the appropriate pager can receive a call but not use the receiver to respond; see Chapter 7.

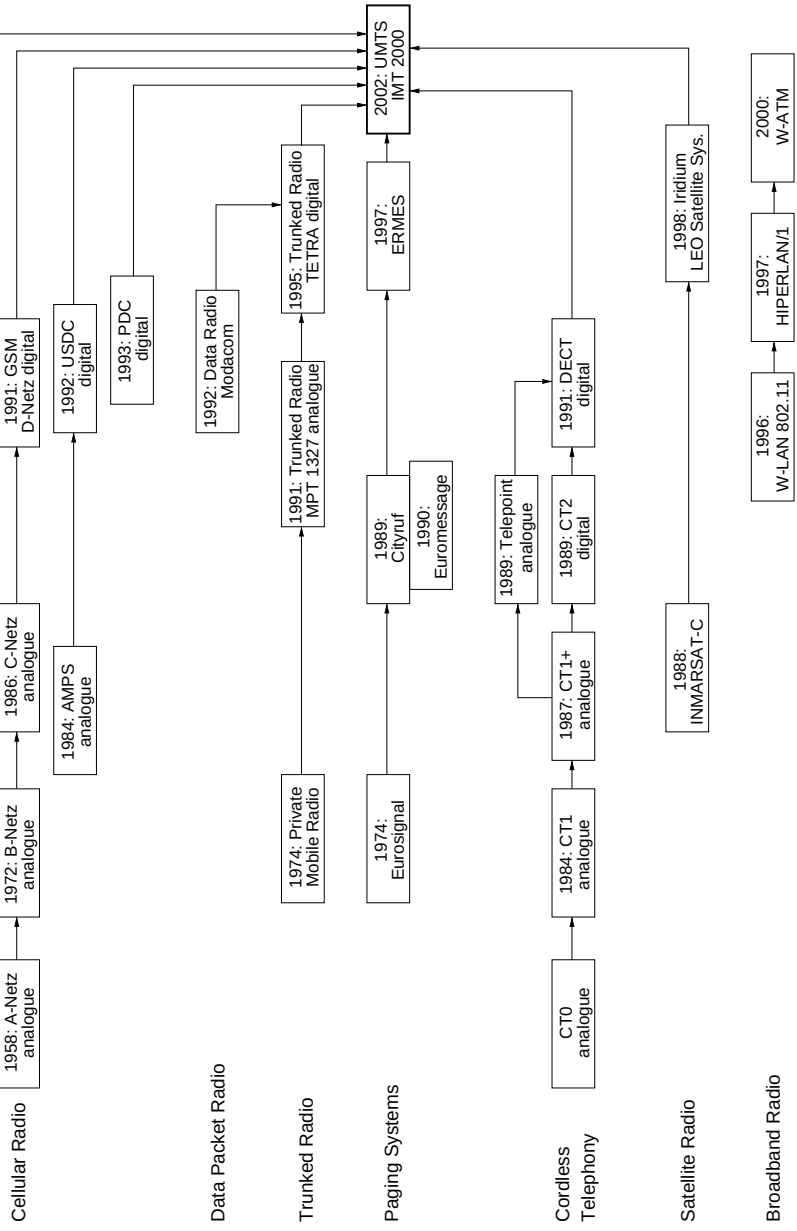


Figure 1.2: Evolution of mobile radio systems (dates refer to introduction in the field)

Table 1.2: An overview of analogue cellular mobile radio

Parameter	C 450	NMT 450	NMT 900	TACS	E-TACS	AMPS
Original country	Germany	Scandinavia	Scandinavia	GB	GB	USA
Standardized by	DBP Telekom			CRAG	CRAG	FCC
Introduced in	1985	1981	1986	1984		1983
Uplink [MHz]	450.3–454.74	453–457.5	890–915	890–915	872–905	824–849
Downlink [MHz]	461.3–465.74	463–467.5	935–960	935–960	917–950	869–894
Channel spacing [kHz]	20	25 (20)	25 (12.5)	25	25	30
Duplex range [MHz]	11	10	45	45	45	45
Access method	FDMA	FDMA	FDMA	FDMA	FDMA	FDMA
Modulation	FSK	FFSK	FFSK	PSK	PSK	PSK
MAH[a]	Yes	No	No	No	No	No
Cell diameter [km]		15–40	2–20			
Frequencies [#]	222	180 (220)	1000 (1999)	1000	1320	833
Data services [kbit/s]	2.4	–	–	–		2.4 (no HO)
Traffic capacity [Erl./km^2] (3 km distance)				14	14	12

[a]Mobile Assisted Handover

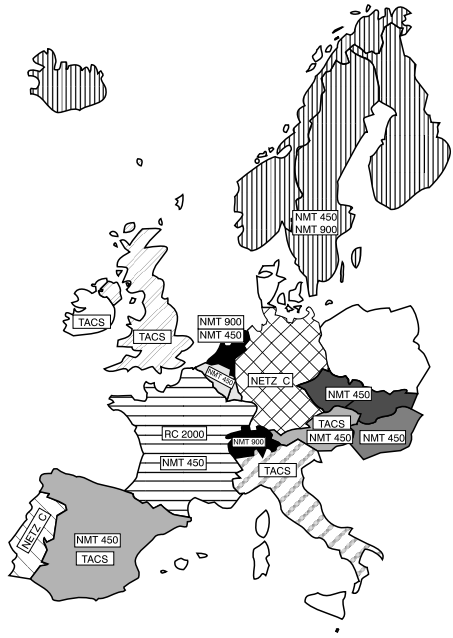


Figure 1.3: Distribution of analogue cellular systems in Europe

The concept of *cordless communication systems* summarizes services and applications based on cordless telephony. In principle, with cordless telephones, the cable between the telephone terminal and the handset was merely replaced by a radio path that allows a radio connection of up to 300 m/50 m (outdoors/indoors); see Chapter 9.

Wireless local area networks take into account the growing demand to avoid cabling of workstation computers; see Chapter 12, whereas *mobile satellite radio systems* provide global communication and accessibility; see Chapter 14.

The mobile communications market is currently developing at a rapid pace, and it is anticipated that the next few years will bring a dramatic growth in the number of users and an increased demand for quality. As a result, the standardization bodies are already developing new standards with the aim of providing a universal, service-integrated mobile telecommunications system in the near future; see Chapter 5.

1.1 Existing and New Networks and Services

1.1.1 GSM/DCS 1800 System

The spectacular growth of the GSM-based cellular mobile radio networks, including networks based on the DCS 1800 standard, convey the impression

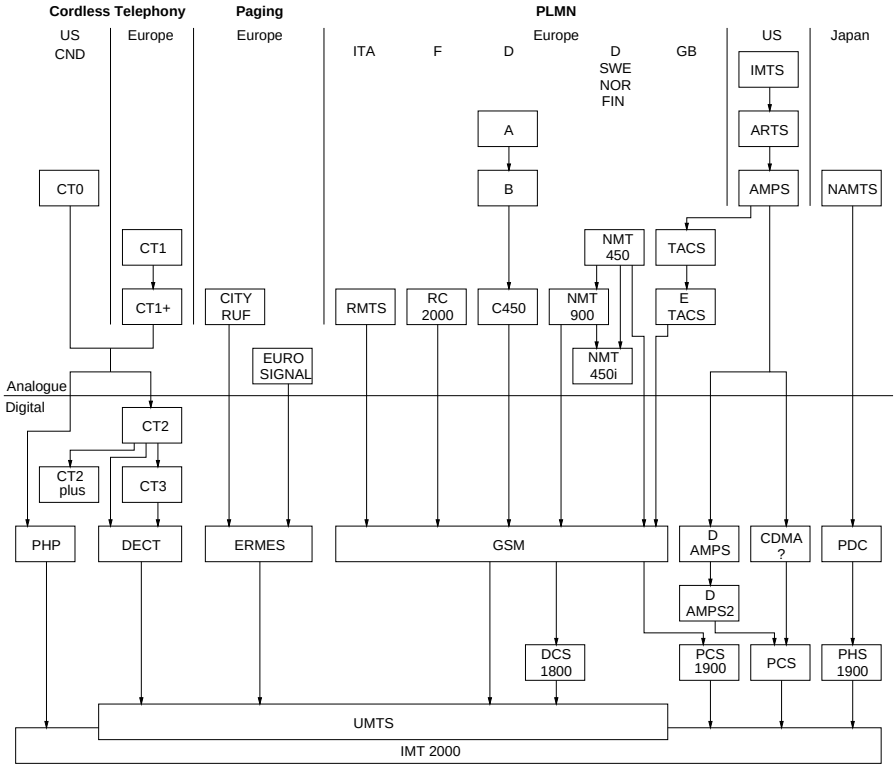


Figure 1.4: Overview of worldwide standards for mobile radio systems

that the essential development needed in this area has been accomplished through the introduction of these cellular mobile radio networks. What one forgets is that these networks have been designed as an "extension" of ISDN to the mobile user, but only address the needs to a limited degree: instead of two B-channels per user, only one with a considerably lower user data rate (13/6.5 kbit/s for voice and 9.6 kbit/s for data) is available. Likewise the ISDN-D channel has only been reproduced to a point: an X.25 packet service (X.31) on the D_m channel is not possible in GSM. The primary rate connection (2.048 Mbit/s) available with ISDN does not exist. The situation is a similar one with competing systems in the USA and Japan (see Table 1.3 and Figure 1.4).

A number of concepts of GSM 900/DCS 1800 systems will have to be developed further in order to head off the competitive pressure of other concepts for cellular networks (UMTS, IMT 2000, Spread Spectrum CDMA) and to provide better support to mobile image and data services. The anticipated demand for ISDN-compatible mobile data services (64 kbit/s) is pressuring

Table 1.3: An overview of digital cellular mobile radio

Parameter	GSM	DCS 1800	D-AMPS	CDMA	PDC
Orig. country	Europe	Europe	US	US, Korea	Japan
Standardized by	ETSI	ETSI	TIA 54	TIA 95	MPT
Introduced in	1992	1993	1991	Test 1991	
Uplink [MHz]	890–915	1710–1785	824–849	824–849	810–826
Downlink [MHz]	935–960	1805–1855	869–894	869–894	940–956
Channel spacing [kHz]	200	200	10	1230	25
Duplex range [MHz]	45	95	45	45	130
Access method	TDMA/FDD	TDMA/FDD	TDMA/FDD	CDMA	TDMA/FDD
Modulation	GMSK	GMSK	$\frac{\pi}{4}$-PQJPSK	QPSK/DQPSK	$\frac{\pi}{4}$-PQJPSK
Min. C/I [dB]	> 9	> 9	> 12		> 13
Speech codec	RPE-LPC-LTP	RPE-LPC-LTP	VSELP	ADPCM	VSELP
Speech data rate [kbit/s]	13 (6.5)	13	7.25	8.55	6.7
Frequencies [#]	124	374	832	10	1600
Time slots/freq.	8 (16)	8	3	–	3
Data service [kbit/s]	9.6	9.6	8	?	9.6–14.4
Max. speed MS [km/h]	250	250	100	100	100
Output power HU [W]	2–5–8–20	0.25–1	0.6–1.2–3–6	0.5–2–6.3	0.3–0.8–2
Traffic cap. [Erl./km^2] (3 km site dist.)	40 (84)		41		91
Spectr. efficiency $\left[\frac{\text{Erl.}}{\text{km}^2 \cdot \text{MHz}}\right]$	1.1 ... 1.6	1.1 ... 1.6	1.64 ... 2.99		2 ... 3.63
Diversity meth.	Interleaving	Interleaving Slow FH	Interleaving Slow FH		Interleaving Antenna Div.

for rapid continued development of the radio interface. Relevant work is being carried out by ETSI GSM/2+ or is being scheduled. Examples include:

- GPRS(*General Packet Radio Service*) for multiplex data transmission of a large number of virtual connections over one or more traffic channels (TCH).
- Multipoint, voice and data services for group communications as usual with trunked radio systems.
- Higher bit rate voice and services for images and data over several traffic channels used in parallel (HSCSD, *High-Speed Circuit-Switched Data*) and use of the total 200 kHz bandwidth per carrier with advanced modulation to provide high-bit-rate services up to 384 kbit/s (EDGE, *Enhanced Data rates for GSM Evolution*).

These further developments will result from or be supported by research in areas such as radio planning, adaptive antennas, propagation modelling, modulation, source and channel coding, combined coding and modulation (*Codulation*) for the improvement of spectral efficiency, data compression, signal processing, microelectronics and switching technology, and electrical power storage. After the basic services have been agreed on and introduced on the network side, new services and applications will be provided through modifications to the radio interface and the incorporation of relevant options for terminals, e.g., mobile computing; limited support of multimedia applications will also be available.

This development of cellular networks should not be viewed in isolation but must be seen from the standpoint of the continuing development of other mobile radio systems that will be introduced at the same time or later and in some cases will represent competition (see Sections 1.1.2–1.1.7). Another point to be made is that cellular networks still do not offer a satisfactory solution for the radio coverage required within buildings and that there is still a great deal of scope for improving the planning tools for radio coverage prediction and algorithms for dynamic channel allocation in hierarchical cell structures. Likewise the organization of intelligent mobile radio networks in terms of signalling, data storage and distribution, availability of (cross-network) value-added services, network management, etc., is still at a relative early stage of development.

It is expected that TETRA-based trunked radio networks will experience demands similar to those of the GSM/DCS 1800 systems, with the added complication of the algorithms and protocols for direct communication between mobile stations (*direct mode*) optionally available from the standard.

1.1.2 DECT

Digital cordless systems based on the ETSI/DECT standard already in use (as "small" systems) are just now conquering the market as "large systems" provid-

Table 1.4: An overview of analogue cordless telephony

Parameter	CT0	CT1
Original country	US, JPN	Europe
Frequency band [MHz]	1.6 / 47	914–916 959–961
Number of channels	8	40
Frequency range [MHz]	0.4	4
Channel spacing [kHz]		25
Access method	FDMA	FDMA
Duplex method	FDD	FDD
Channel allocation	Fixed	Dynamic
Cellular capability	No	Limited
Max. power HU [mW]		10
Range [m]	< 1000	< 300
Handover	No	No
Traffic capacity [Erl./km^2]	1	200

ing partial coverage to the application areas of larger private branch exchange systems, thereby replacing the analogue predecessor systems (see Table 1.4). These systems are extremely suitable for mobile applications indoors and in close proximity to the respective base station (up to approximately 300 m outdoors).

Because only a fraction of future mobile users will be using communications services outside metropolitan areas, DECT systems could be used as personal communications systems in densely built-up areas to reach a high percentage of all mobile radio subscribers if sufficient radio coverage were available, and, with the function mobility management implemented, might provide considerable competition to the cellular networks discussed in Section 1.1.1. The *Personal Handyphone System* (PHS) in Japan (see Table 1.5) is a good example.

The inherent strength of cordless systems lies in their suitability for providing coverage indoors, which easily lends itself to providing a limited range outdoors, also.

Compared with GSM, DECT systems are less restrictive in terms of the services that can be specified at the radio interface, and therefore allow a higher level of flexibility, e.g., for use in a local area network (LAN) or as the technology for wireless access to the telecommunications network (*Radio in the Local Loop*, RLL) (see Figure 9.64).

On one hand, DECT systems are preventing users from migrating from fixed network operators to mobile radio network operators, but, on the other hand, they promote a migration to mobile radio network operators because users already have a fixed network.

Table 1.5: An overview of digital cordless telephony

Parameter	CT2	DECT	PHS
Original country	GB	Europe	Japan
Standardized by	BT (ETSI)	ETSI	TTC/RCR
Introduced in	1992	1993	
Frequency range [MHz]	864.1–868.1	1880–1990	1895–1918
Radio carrier spacing [MHz]	0.1	1.728	0.3
Channels [#]	40	120	
Transmitted data rate [kbit/s]	72	1152	
Channel allocation method	DCA	DCA	DCA
Speech data rate [kbit/s]	32	32	32
Speech coding	ADPCM G.721	ADPCM G.721	ADPCM
Control channels	In-call emb.	In-call emb. (logical channels: C, P, Q, N)	Fixed control carriers
In-call control channel data rate [kbit/s]	MUX 1.2, 0.75, 1.4, 1.5	4.8 (+ 1.6 CRC)	
Total channel data rate [kbit/s]	33/34	41.6	
Duplexing technique	TDD	TDD	TDD
Multiple access TDMA [Timeslots]	1 TDD	12 TDD	4 TDD
Carrier Usage FDMA/Multicarrier [# Carriers]	40	10	
Bits/TDMA timeslot (speech/data+emb. ctl.) [bit]	66/68	420 (4 bit Z-field)	
Timeslot duration (incl. guard time) [µs]	1	417	
TDMA frame period [ms]		10	
Modulation technique	GFFSK	GFFSK	π/4-DQPSK
Modulation index	0.4–0.7	0.45–0.55	
Traffic capacity [Erl./km²]	250	10,000	
Handover	No	Yes	
Cellular capability	Limited	Yes	
Peak output power [mW]	10	250	10
Mean output power [mW]	5	10	

1.1.3 Radio Networks as a Bypass to the Local Loop

Deregulation in Europe ends the monopoly on voice services of the incumbent operator, and is resulting in the expansion of former corporate networks by new network operators in competition with the respective incumbent (some of them also using lines leased from the incumbent), who are providing services to large customers and (eventually) all conveniently located corporate and private customers. This expansion is being accompanied by the development and establishment of local cellular radio networks that use point-to-multipoint radio relay or fixed radio user connections (see Figure 10.6), offer ISDN-based and primary rate multiplex interfaces and can be used as access networks (*Radio in the Local Loop*, RLL) to fixed networks of the incumbent's competitors.

GSM and DCS 1800 only have limited application in this area because of their noticeably lower transmission rates compared with ISDN. In multichannel operation DECT can offer ISDN interfaces; the appropriate standards were drawn up by ETSI/RES 03 in 1996. Radios in the local loop networks are closely related to the systems described in Sections 1.1.1 and 1.1.2 but require further development to enable them to make better use of frequencies and operate more cost-effectively.

Along with cellular networks that provide sectorial or radial coverage in the proximity of a base station, chains of base stations (DECT relay) and tree-like arrangements of radio links, starting from the fixed network access point, are also expected to bridge the "last mile" between fixed networks and customers in the local network area. The same frequency band used by cellular systems (e.g., with DECT) or public radio relay bands (e.g., 2.6/3.4/10/17/23/27/38 GHz) will be used.

All the systems described above will place considerable and, in some cases, new demands on radio network planning, on the procedures for dynamic channel allocation and on hierarchical cell structures, for which flexible solutions will have to be sought. In all probability, RLL technology will be developed and tested in Great Britain, and will then expand to the rest of Europe, with appropriate export opportunities to other continents.

1.1.4 Wireless Local Area Networks (IEEE 802.11 WLAN, Wireless LAN, ETSI/HIPERLAN/1)

There is a considerable demand today for the wireless connection of (*movable*) workstation computers to provide flexibility in how and where equipment is installed in order to use standard Internet applications, which today are often accessed over a local area network (LAN). Standardization has just produced solutions that constitute the first fast step in this direction. So-called single-hop solutions are currently possible; these tend to require the connection of a base station to a fixed network (e.g., LAN) for each office room served at the frequencies 2.4/5.3/40/60 GHz. Further development is possible and necessary to reduce the cabling required.

Since these networks permit data transfer rates comparable to LANs (typically up to 20 Mbit/s), they are more suitable for replacing LANs and less appropriate for supporting new multimedia services. These new services place real-time demands on a transmission system that in principle cannot be supported by the Internet, or at least not until considerable further development has been carried out in this area. Movable workstations along with mobile terminals can be supported. In addition to radio, media such as infrared and visible light are also being considered for wireless LANs. Terminal mobility (or movability) is placing new demands on Internet protocols.

Consequently, there is a considerable need for research and development to evaluate and improve the mobile Internet protocol (*mobile IP*) to be used with wireless systems. It should be mentioned here that in the future Internet protocols will increasingly be used from terminals to fixed and mobile radio networks. The work being carried out on mobile IP is also important for the mobile radio networks discussed in Sections 1.1.1–1.1.3 (and to those which follow).

1.1.5 Wireless Networks for Process Control

This area of application has recently been of special interest, and could be ready for the introduction of wireless communications systems, because existing wireline networks are usually proprietary solutions and users are demanding open communication architectures. A major upheaval is imminent that could also create opportunities for open radio-based systems. Certain industrial environments have special requirements for transmission techniques and protocols that are not easily or readily accommodated by the systems that exist in the other areas described in Sections 1.1.1–1.1.7.

What will be characteristic of the new wireless communications systems in process control is that standard PCs and LAN technology, supplemented by wireless systems, will displace the current solutions based on storage-programmed controllers (SPC) connected by so-called field-bus systems.

1.1.6 Universal Mobile Telecommunications System UMTS

The MoU UMTS (*Memorandum of Understanding for the Introduction of UMTS*) group promotes a revolutionary (non-evolutionary) approach to the further development of current systems, and their integration into existing systems and networks is desirable, even if the technical implementation will be costly. The critical factor with current mobile communications systems is the bit rate, which is not sufficient for the new applications of the future and should be allocated flexibly as required. UMTS was regarded by some people less as a totally new system but more as a further development of GSM. The current development is deviating to some extent from that view in that at least the radio interface will be designed newly, but a number of services kept as they are (see Chapter 5).

1.1.7 Wireless Broadband Systems

Owing to the increasing introduction and growing use of broadband services over fibre optic networks based on ATM transmission technology (broadband ISDN) with transmission rates of 34 (E3), 155, 622 and 2400 Mbit/s, a broadband option is required for connecting movable or mobile terminals, similar to GSM/DCS 1800 to connect to the narrowband ISDN. The current state of technology enables the implementation of radio-supported, cellular mobile broadband systems with 25 Mbit/s user data rates. In contrast to the systems mentioned in Section 1.1.4, these are real-time wireless ATM systems based on ATM cell transmission that logically are most comparable to DECT (related to ISDN). As soon as ATM networks (with real-time capability) directly connect terminals, many of the Internet protocols that were developed for heterogeneous, error-prone non-real-time-capable networks and services will have to be rethought.

It will take several more years of effort to resolve the problems that arise when wireless broadband systems are introduced in all the areas already mentioned in Section 1.1.1. ETSI/BRAN has been developing W-ATM standards for RLL, radio LANs and cellular systems since 1996. The ATM Forum has been developing protocols for mobility management in ATM networks also since 1996 (see Chapter 12). Carrier frequencies of 5.3/17/40/60 GHz are being planned because of the large frequency bandwidth required.

1.1.8 Mobile Satellite Radio

Geostationary satellites are preferred for providing coverage to slowly moving stations (ships) because of the large receiving antennas required owing to high signal attenuation. Various groups of companies are planning global mobile radio networks on the basis of low (700–1700 km height, LEO, *Low Earth Orbit*) and medium–high (10000 km height, ICO, *Intermediate Circular Orbit*) flying satellites (see Tables 1.6 and 1.7). The aim is to guarantee radio coverage at 1.6 GHz for hand-portable satellite receivers (300 g). Although these systems are primarily geared to providing coverage to rural and suburban areas, it is evident that plans exist to provide wide-area coverage with high capacity, including areas that are also well supplied by ground-based cellular networks. This means that, in addition to the efforts involved in the development and evaluation of these systems, issues concerning the cooperation with terrestrial mobile radio and fixed networks will also have to be resolved. Handover procedures in hierarchical cell structures, from picocells to satellite umbrella cells, will have to be developed (see Section 1.1.1).

In addition to the switching functions on board satellites for the connection of mobile stations to a suitable ground base station, other problems still need to be resolved, such as routing between mobile satellites and the control of the radio links between the satellites. The IRIDIUM system is a first example of this. Satellite networks, like terrestrial mobile radio networks, will endeavour

Table 1.6: An overview of mobile satellite telecommunications, Part 1

Name	IRIDIUM	Project 21	Globalstar	The Calling Network (Brilliant Pebbles)	Odyssey
Prime company, country	Motorola, US	Inmarsat, GB	Qualcom, US (LQSS)	Global Com. Inc.	TRW, US
Services offered	Voice, Data, Fax, GPS, Paging	Voice, Data, GPS, Paging	Voice, Data, Fax, GPS, Paging	Voice	Voice, Data, GPS
Coverage	Global	Global	Global	Underdev. Regions	Global
Orbit type	LEO	ICO	LEO	LEO	MEO
Orbit height	778 km	10000 km	1389 km	600 km	10370 km
No. of orbits	6		8	21	3
Sats. per orbit	11		6	40	4
Total no. of sats.	66	12–15	48	840 (+80 spares)	12
Cells/sat.	48		6		37
Channels/sat.	4070		2700		2300
1st sat. launched	1996		1997	1996	
Full operation	1998	2001	1999	1999	
Costs	3.4–3.7 bill. $	2 bill. $	0.82–1.5 bill. $	6.5–7 bill. $	1.3–1.4 bill. $
Charge for voice service per min.	3 $		0.35–0.45 $		0.65 $
Terminal mode	Dual-mode	Dual-mode	Dual-mode		
Access method	FDMA/TDMA	Not yet decided	CDMA	FDMA/TDMA	

Table 1.7: An overview of mobile satellite telecommunications, Part 2

Name	Inmarsat A	NAME	Telesat	Arles	Ellipsat
Prime company, country	Inmarsat, GB	American Mobile Sat. Corp., US	Telesat Mobile, CND	Constellation Communication, US	Ellipsat Corp., US, GB, ISR
Services offered	Voice, Data, Telex	Voice, Data, Fax	Voice, Data, Fax	Voice, Data, Fax	Voice
Coverage	Global	Global	Global		Global (North and South Zone)
Orbit type	GEO	GEO	GEO		
Orbit height	36000 km	36000 km	36000 km		
Number of orbits	1	1	1		3 (North)
Sats. per orbit	3	3	3		5 (North)
Total no. of sats.		3	3	48	15 (North) + 6 (South)
Cells/sat.				7 . . . 19	
Channels/sat.				50	
1st sat. launched	1976	1995	1996		
Full operation	1979	1995	1996		
Costs				290 bill. $	219 bill. $
Charge for voice service per min.					0.5 $
Terminal mode	Sat. only	Dual-mode	Dual-mode		Dual-mode

to pick up traffic close to the source and transfer it close to the destination, with no use or minimal use of other fixed networks. Research interest is focused on the problem of interference between space segments of the same or of different satellite systems and between space and ground segments.

1.1.9 Universal Personal Mobility

In addition to radio and transmission-related functions, mobile communication requires special services from the fixed networks. Mobile radio systems usually consist of a radio and a fixed part. The mobility management of users is essentially implemented through functions in the fixed network based on functions of *Common Channel Signalling System Number 7* (SS 7).

Architectures for *Universal Personal Telecommunication* (UPT) and *Intelligent Networks* (IN) are currently being developed worldwide for fixed networks and standardized by the ITU-T (see Chapter 15). This means that it will eventually be possible to reach a person anywhere in the world under one personal telephone number, for all services and over fixed and mobile radio networks, independently of the network service provider. The concepts for mobility across network domains still have to be developed.

Advantages are to be exploited and disadvantages avoided, with users given control in each specific situation over which callers are allowed to reach them, which services can be used and what should be done with other calls or incoming messages (concept of the subscriber's role) (see Chapter 15).

All services marked as not to be switched to the subscriber will be dealt with according to his instructions, e.g., transferred into another form of service, routed to a storage device or diverted to a third party. These types of services will initially be primarily developed for subscribers of mobile radio networks, because only they will have universal access to the network. Consequently, these services will be implemented and introduced within the context of mobile communications. So there is a need to:

- Consider services in the context of new generations of mobile terminals.
- Develop intelligent services with, for example, functions based on previous history and according to terminal, time and place of use (e.g., the so called location aware services).

Finally, there should be some mention of the work involved in the future *Telecommunication Information Network Architecture* (TINA) that is being developed by the international consortium TINA-C to increase the flexibility in using communications networks.

1.2 Systems with Intelligent Antennas

Studies have recently been conducted into all types of mobile radio systems to examine possibilities for increasing efficiency [(bit/s)/(MHz·km^2)] through

the use of *smart antenna arrays*, see also Section 14.4. For obvious reasons (dimensions, complexity, ability to use existing mobile devices without requiring changes), this technology is initially being discussed for use in the base stations of cellular systems. The range (and consequently the cell radius) can be increased or the transmitter power (and consequently the interference) reduced owing to the array gain achieved through adaptive forming of the antenna diagram. This might result, at the end of development, in dynamic, radio-relay-like point-to-multipoint mobile communication. A first step into this direction is represented by the phased array antenna systems used by LEO satellite systems, e.g., IRIDIUM and TELEDESIC, to establish a set of cells on earth by means of a small number of antenna systems only.

Over and above this increase in efficiency through a reduction in the transmitting power and/or an increase in coverage range, it appears possible to implement true *Space-Division Multiple Access* (SDMA) for a dramatic increase in spectral efficiency and network capacity [Erl./(MHz·km²)]. This access procedure is not to be seen as an alternative to the established procedures *Code/Time/Frequency-Division Multiple Access* (C/T/FDMA) but instead as a compatible extension to them. The idea is that a receiver with an *antenna array* receives the signals of several users who are using the same time/frequency/code channel, and from this calculates the geographical *directions of arrival* (DoA). This directional information is used for spatial filtering on the uplink and beam forming on the downlink, which can be imagined as a simultaneous adaptive forming of the antenna directional diagram for each user with only one antenna array.

Extensive research and development will be necessary into the overall concept and into almost all the system components before these types of systems are introduced. This applies to the antenna arrays themselves, the associated transmit and receive parts (*front ends*), and above all the algorithms for processing the signals (parameters estimation, data estimation, beam forming) and intelligent (dynamic) channel allocation. It is clear that a directionally based separation of users (SDMA) can only be achieved if users in the same channel can be spatially well separated from each other. The ability to create these spatially well-separated user groupings is an important task of channel allocation.

It is also obvious that the protocols for the radio interfaces of existing mobile radio systems will have to be adapted to these new concepts and that radio resource management in the network will derive considerable advantages from a dynamic channel allocation procedure that minimizes the transmit power while guaranteeing a minimum interference level. This optimized channel allocation within a cell or beyond cell boundaries creates the expectation of a considerable increase in spectral efficiency and network capacity, and in fact irrespective of whether the basic system is a F/TDMA type (GSM), a CDMA type (IS-95) or a hybrid form (such as the UMTS under discussion). However, the results for the different basic systems will vary depending on the detailed formulation and optimization of the overall concept.

1.3 Mobile Radio Systems with Dynamic Channel Allocation and Multiple Use of Frequency Spectrum

Dynamic channel allocation is an intelligent method for allocating radio resources as required for wireless communication between a terminal and a base station. This measure on its own can increase the capacity of an ETSI/DECT system (one with standardized dynamic channel allocation) in indoor application cost-effectively by a factor of two to four compared with an ETSI/DCS 1800 system (one that uses fixed channel allocation). Some initial publications indicate it could be possible to achieve comparable or somewhat lower capacity increases with mobile radio systems. This would apply to all the systems mentioned in Section 1.1, thereby making the research and development work for each of the systems an attractive proposition.

Dynamic channel allocation produces higher capacity, which in turn enables a larger number of communications relationships to be implemented simultaneously in the available frequency ranges. This will be possible in the existing GSM900/DCS 1800 systems when new procedures for channel allocation are developed and tested.

Owing to the scarcity of frequency spectrum for mobile radio applications, the Federal Communications Commission (FCC) (USA) and European Radio Office (ERO) have made inital allocations for joint use of the same spectrum for public mobile radio services.

Little is known at present about the compatibility of mobile radio systems in adjacent frequency bands and the compatibility of systems operating in the same frequency band. Here too it obviously comes down to improving spectral efficiency through measures allowing the competitive use of the same frequency band. Figure 1.5 shows the neighbouring relationships between some of the radio systems affected.

1.4 Other Aspects

1.4.1 Self-Organizing 4th-Generation Systems

Decentralized organizational forms (eliminating a centralized base station) would appear to have an advantage for applications with a high local density from wireless communicating stations that operate in frequency bands above 2.5 GHz and therefore require a line-of-sight connection between each other. Ad hoc networks are being discussed, the key feature of which is fully decentralized self-oganization. Other characteristics of these systems include:

- Use of some or all stations as relays on the multi-hop route between communicating stations.

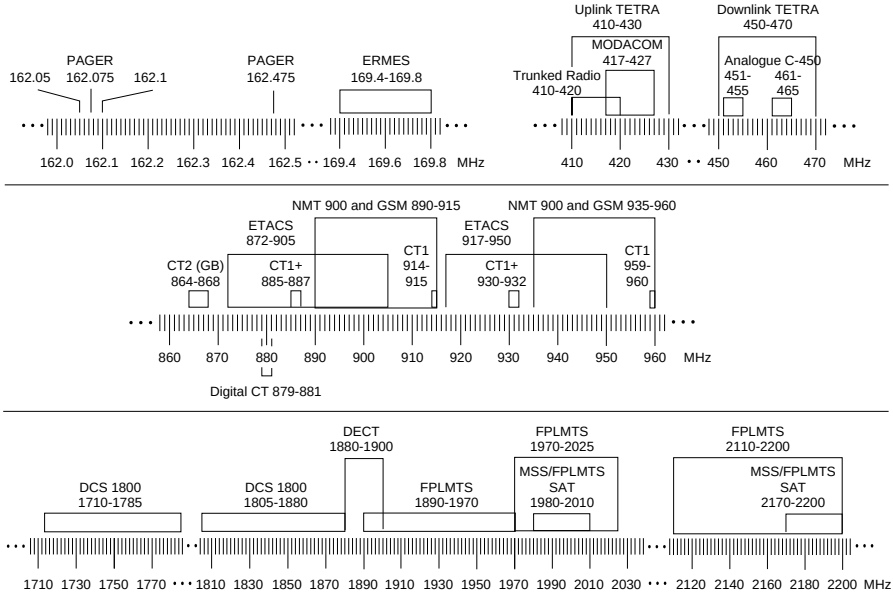


Figure 1.5: Frequency use for mobile radio services

- Support of synchronous and asynchronous transmission services, such as those customary with ISDN and local networks.
- Autonomous route selection and operation of the stations, including sleep mode.
- Installation of gateway stations for the link to fixed network.
- Decentralized network management.
- Local restriction to areas with, for example, a diameter of very few kilometres.
- Dynamic reuse of radio resources in accordance with the cellular principle.

These kinds of systems typically require several radio sections for each communication relationship; in other words, they place a higher demand on the spectrum than conventional (mobile) radio systems, which only need one radio section per communications relationship. Hence multi-hop systems are dependent on measures that increase capacity, such as adaptive antennas and the use of SDMA, to achieve comparable efficiency. Their application domains will be connectivity to sensors, actuators, clients, and servers for multiple applications where communication might be asynchronous, i.e., from time to time with possibly a low bit rate only. Investigations on these 4th-generation systems have just been started.

1.4.2 Electromagnetic Environmental Compatibility

Conventional mobile radio systems use omnidirectional antennas, which adversely affect the environment because of the electromagnetic field produced ("electrosmog"). Intelligent antennas deliberately direct the transmitter power towards the receiver, which, compared with omnidirectional antennas at the same range, reduces transmitter power considerably.

The effects of electromagnetic waves on biological systems have been examined scientifically for many years with no indications of a negative impact on human health when the equipment is operated according to the regulations defined in the respective standards. The best knowledge on the use and control of radio waves is being taken into account in the development of new technologies for mobile radio systems.

1.5 Historical Development

Communications networks began their triumphal march in 1843 when approval was granted by the American Congress for the first test section for Morse telegraphy along a rail route between Washington and Baltimore. Wired voice transmission was first possible through the invention of the telephone by Alexander Graham Bell in 1876.

1879 *Hughes* presented the phenomenon of electromagnetic waves to the Academy of Natural Sciences in London. Because the Maxwellian rules on the propagation of electromagnetic waves had not yet been recognized at that time, Hughes' results were rejected [1, 2].

1881 The first public telephone network in Berlin was installed. Point-to-point voice transmission was made possible with the help of the switchboard operators called *Fräuleins vom Amt*, who switched calls from one terminal to another terminal. Previously this service had only been offered by telegraphists.

During the following century, telephone networks were installed and constantly extended. Then they were equipped with automatic switching and expanded into regional, national and, finally, worldwide networks. The telephone became a part of daily life, although its use was restricted to fixed wired networks.

1888 *Hertz* was successfully able to reproduce and confirm the Maxwellian theory. He demonstrated that a spark produced from a transmitter at a nearby receiver produced a voltage. During the 1890s, *Tesla* extended the bridgeable distance.

1897 *Marconi* developed the first usable system for wireless telegraphic transmission over large distances. A Morse key was used to produce a spark in the transmitter. The receiver contained a coherer—a tube filled with

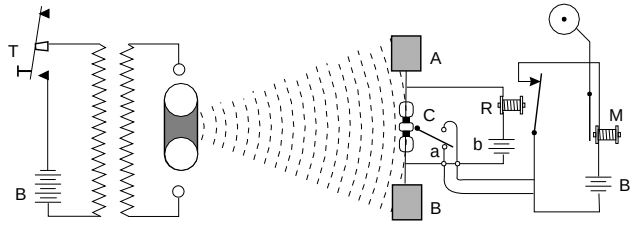


Figure 1.6: Wagner hammer

ferrous powder that was connected to a direct current supply. The voltage was set so that the electric circuit, including an electromagnetic printer, did not close.

The received electromagnetic wave created by a spark in the transmitter causes the receiver circuit to be closed. A so-called Wagner hammer (see Figure 1.6) ensures, through shaking the coherer (C), that the conducting receiver circuit is opened again. Antennas A and B are adjusted to the oscillator frequency of the resonant circuit of the transmitter.

1901 *Marconi* succeeded in transmitting wireless signals over the Atlantic. However, the transmitting and receiving equipment used was so large that it could only be used in stationary locations.

1902 A radio from the company Telefunken (Germany) used in the military is shown in Figure 1.7. The top cart is carrying a 3 kW gas motor, which is driving a 1 kW alternating-current generator. The transmitting and receiving device is mounted on the cart below, and the Morse key is recognizable in the foreground. The antennas, which are not pictured, were very large because of the short-wave frequencies used.

1903 At this time, the first ships were equipped with radio facilities to provide shipping companies and the military with wireless communication.

Braun, Slaby and *v. Arco* of Telefunken developed a closed resonance circuit that improved the adjustment to a given frequency. At the same time, this provided a way of bypassing the Marconi patents.

1906 It became necessary to coordinate radio frequencies as more and more ships were being equipped with radio systems and it was possible to filter the spectrum occupied by a radio transmitter. At the first *World Administrative Radio Conference* (WARC) specific frequency bands were allocated to different services in order to limit reciprocal radio interference.

With the invention of the triode by *von Lieben* in 1910, transmitters based on sparking were very quickly replaced by smaller and lighter devices.

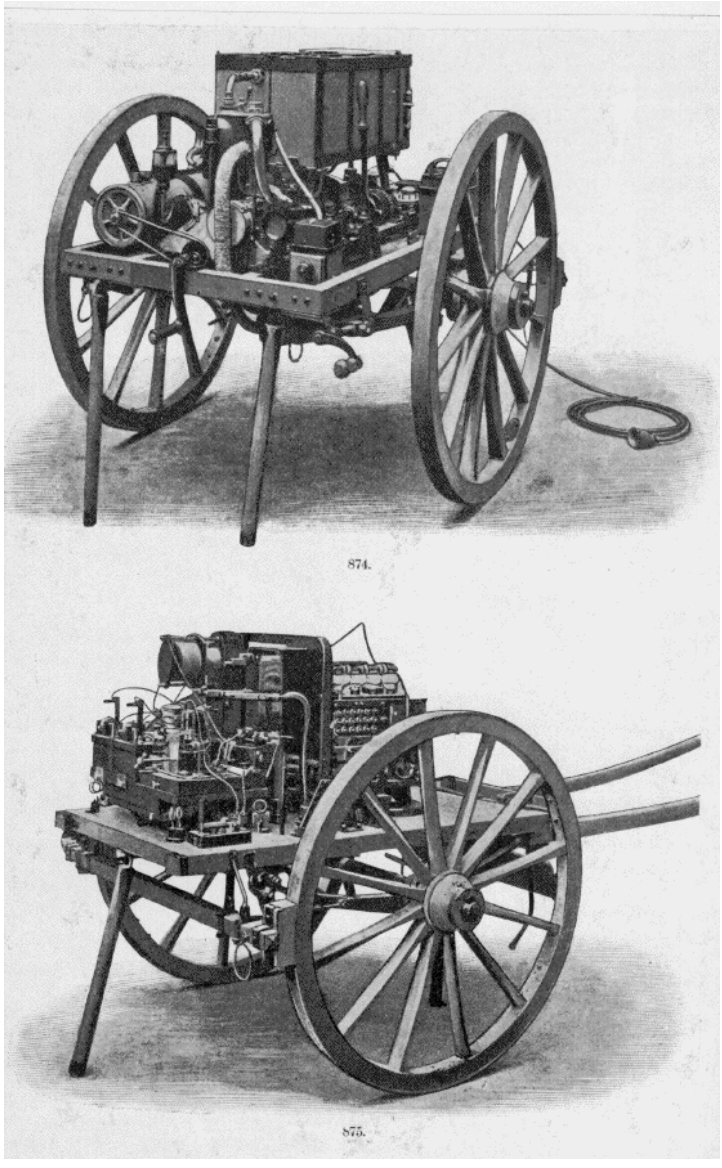


Figure 1.7: Power and equipment cart for the Telefunken telegraphy system

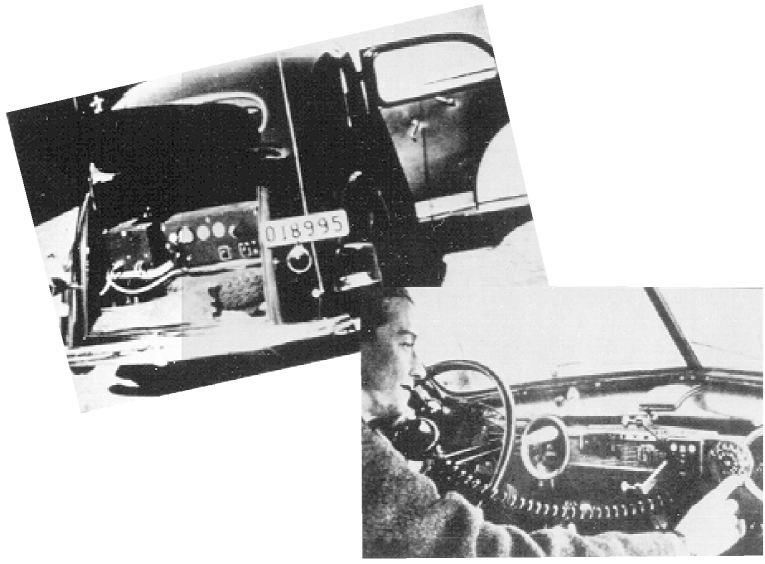


Figure 1.8: Car telephone 1935

1912 The use of frequency bands up to 3 MHz was regulated at the second WARC. Higher frequencies were judged as being commercially not viable, and were therefore released for private use by radio amateurs. However, this decision was reviewed, and during the following years the commercial use of the spectrum was quickly regulated:

1927	up to 30 MHz	1947	up to 10.5 GHz
1932	up to 60 MHz	1959	up to 40 GHz
1938	up to 200 MHz	1979	up to 275 GHz

1935 The first transmitting and receiving equipment for private users (e.g., taxis) appeared on the market. They used electronic tubes and could be installed in vehicles (see Figure 1.8). The problem was that they completely filled up all the space in the boot.

1952 It was now possible to call a user of a mobile terminal from a fixed network connection. Local radio systems started to be used more and more [2]:

- A single *base transceiver station* (BTS) was used in an area 20–100 km in diameter.
- Around 20 participants shared one voice channel.

- In the beginning manual switching was still required to establish a connection to the fixed network; later this was replaced by automatic switching.
- Calls to a mobile device from the fixed network were supported.
- A file with user-specific data (*Home Location Register*, HLR) was established, identifying the local switching centre as the gateway to the BTS.
- Voice communication was carried out semi- or full-duplex.

1958 Isolated systems typically supplied coverage to city areas, and were restricted to their terminals. When users left the city, they could not use the mobile radio devices in another city, even if the same system was installed there (the frequency used was a different one and there was no roaming agreement between the individual operators).

Countrywide mobile radio systems allowed subscribers to establish a connection with any BTS of the system. Instead of one frequency channel, a whole bundle of frequencies was available to the mobile terminal. Any free channel could be used for transmission. Through the utilization of trunking gain, it was possible for more traffic to be carried with the same blocking probability. Base stations were either linked individually over gateway nodes to the fixed network or intermeshed together and connected over a central gateway to the fixed network.

The rest of this chapter describes the development of cellular radio networks in Germany. The A-Netz in Germany is an example of an early *Public Land Mobile Radio Network*, PLMN. An operator supported the switching, and every BTS was available to radio telephones.

1972 The B-Netz was introduced in Germany, Austria, the Netherlands and Luxemburg. It supports fully automatic switching of the incoming and outgoing calls of mobile stations (MS) and roaming between the four participating countries. A caller from the fixed network has to know the number of the base station where the mobile user is currently located. The mobile user's complete number consists of the location area code of the BTS, the number of the gateway node and the user identification. The MS is paged on a system-wide frequency, and receives a radio channel when it responds to the call.

A subscriber can restrict outgoing calls from an MS to certain frequency channels in order to optimize operating costs. A radio channel is used for inband signalling to establish a connection. If the mobile station leaves the coverage area of its BTS, the connection is broken off. There is no handover—either from the frequency channel of the BTS to another one of the same BTS or to a neighbouring cell.

1989 The C-Netz was the first mobile radio network in Germany in which automatic interruption-free *handover* was executed for mobile users chang-

ing from one radio supply zone (cell) to another cell. The network has fully automatic mobility management so that the location areas of the switched-on terminals are constantly updated and a user can automatically be located over the corresponding database when there is an incoming call, without any operator support. The network had a maximum of 850 000 subscribers in 1995 that is now continuously decreasing in favour of the GSM systems.

1992 The D1-Netz based on the European ETSI/GSM standard was introduced. It transmits digitally, and eliminates the incompatibility that previously existed between national mobile radio networks in Europe. The first operator was T-Mobil, a subsidiary of Deutsche Telekom AG.

1993 As a result of the deregulation of mobile radio in Europe, the D2-Netz began operating as an area-wide GSM network in Germany. The operator is Mannesmann Mobilfunk GmbH.

1995 The E1-Netz based on the ETSI/DCS 1800 standard was launched as another area-wide mobile radio network. The operator is E-Plus Mobilfunk.

1997 A license was granted for the operation of a DCS 1800 network E2, which must achieve 75 % area coverage in Germany and has begun operations in October 1998.

Many other mobile radio systems have been successfully introduced along with these public cellular systems (see Figure 1.2 and Tables 1.1–1.7).

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2
System Aspects

2.1 Fundamentals of Radio Transmission

In mobile radio systems, unlike wired networks, electromagnetic signals are transmitted in free space (see Figure 2.1). Therefore a total familiarity with the propagation characteristics of radio waves is a prerequisite in the development of mobile radio systems. In principle, the Maxwell equations explain all the phenomena of wave propagation. However, when used in the mobile radio area, this method can result in some complicated calculations or may not be applicable at all if the geometry or material constants are not known exactly. Therefore special methods were developed to determine the characteristics of radio channels, and these consider the key physical effects in different models. The choice of model depends on the frequency and range of the radio waves, the characteristics of the propagation medium and the antenna arrangement.

The propagation of electromagnetic waves in free space is extremely complex. Depending on the frequency and the corresponding wavelength, electromagnetic waves propagate as ground waves, surface waves, space waves or direct waves. The type of propagation is correlated with the range, or distance, at which a signal can be received (see Figure 2.2). The general rule is that the higher the frequency of the wave to be transmitted, the shorter the range.

Based on the curvature of the earth, waves of a lower frequency, i.e., larger wavelength, propagate as ground or surface waves. These waves can still be received from a great distance and even in tunnels.

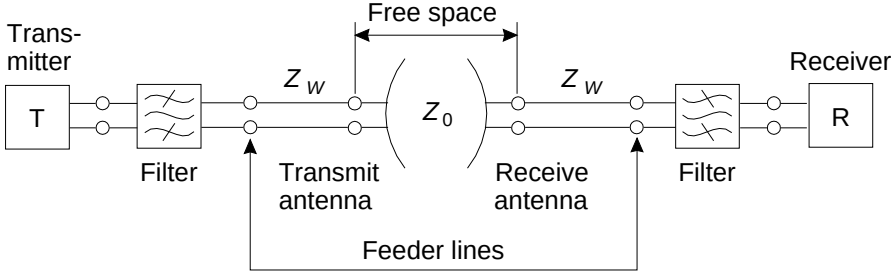


Figure 2.1: Radio transmission path: transmitter–receiver. Z_0 and Z_W are the radio wave resistances in free space and on the antenna feeder link

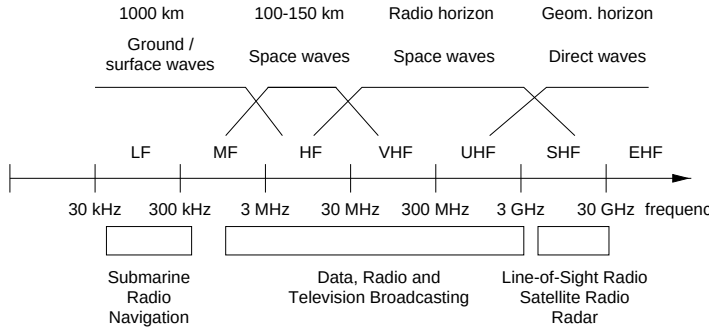


Figure 2.2: Propagation and range of electromagnetic waves in free space

In the higher frequencies it is usually space waves that form. Along with direct radiation, which, depending on the roughness and the conductivity of the earth's surface, is quickly attenuated, these waves are diffracted and reflected based on their frequency in the troposphere or in the ionosphere.

The range for lower frequencies lies between 100 and 150 km, whereas it decreases with higher frequencies because of the increasing transparency of the ionosphere, referred to as the radio horizon. When solar activity is intense, space waves can cover a distance of several thousand kilometres owing to multiple reflection on the conductive layers of the ionosphere and the earth's surface.

Waves with a frequency above 3 GHz propagate as direct waves, and consequently can only be received within the geometric (optical) horizon.

Another factor that determines the range of electromagnetic waves is their power. The field strength of an electromagnetic wave in free space decreases in inverse proportion to the distance to the transmitter, and the receiver input power therefore fades with the square of the distance. The received power for omnidirectional antennas can be described on the basis of the law of free-space propagation.

An ideal point-shaped source, a so-called isotropic radiator of signal energy, transmits its power P_T uniformly distributed into all directions. Such a transmitter cannot be realized physically. The power density flow F through the surface of a sphere at a distance d from an ideal radiator (see Figure 2.3) can be expressed as

$$F = \frac{P_T}{4\pi d^2} \ [\mathrm{W/m^2}] \tag{2.1}$$

In most cases antennas are used that focus the radiated power into one direction. The resultant antenna gain $g(\Theta)$ into the direction Θ is expressed by the radiated power normalized to the mean power, where P_0 represents the total transmit power emitted from the antenna.

$$g(\Theta) = \frac{P(\Theta)4\pi}{P_0} \tag{2.2}$$

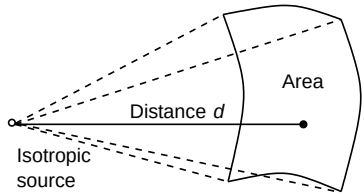


Figure 2.3: Power density flow F

The maximum signal energy radiated from the antenna is transmitted into the direction of the main lobe. The maximum antenna gain g_{max} at $\Theta = 0$ gives the amplification measure in comparison with an isotropic radiator using the same signal energy.

According to Equation (2.1), the power density flow of an ideal loss-less antenna with gain g_T is

$$F = \frac{P_T g_T}{4\pi d^2} \quad [\mathrm{W/m^2}] \tag{2.3}$$

The product $P_T g_T$ is called EIRP (*Effective Isotropically Radiated Power*). This is the transmit power necessary with an omnidirectional isotropic radiator to reach the same power density flow as with a directional antenna diagramme.

The energy arriving at the receiver is

$$P_R = P_T g_T g_R \left(\frac{\lambda}{4\pi d}\right)^2 \tag{2.4}$$

In Equation (2.4) P_T represents the power radiated by the transmitter and P_R the input power of the receiver. g_T and g_R stand for the corresponding absolute antenna gains. λ is the wavelength and d the distance between sender and receiver.

The free-space path loss

$$L = \left(\frac{\lambda}{4\pi d}\right)^2 \tag{2.5}$$

describes the spatial diffusion of the transmitted energy over a path of length d, and g_R is the receive antenna gain.

In a logarithmic representation this produces the path loss $(P_T - P_R)$

$$L_F = -10\log g_T - 10\log g_R + 20\log f + 20\log d - 20\log\frac{c}{4\pi}$$

with c representing the wave propagation speed.

In a simple case scenario with isotropic antennas the free-space attenuation L_0 is produced without antenna gain as the difference between received power and radiated power:

$$L_0[\text{dB}] = P_R[\text{dBm}] - P_T[\text{dBm}] = -10\log\left(\frac{P_R[\text{mW}]}{P_T[\text{mW}]}\right) = -20\log\left(\frac{\lambda}{4\pi d}\right) \tag{2.6}$$

2.1.1 Attenuation

Weather conditions cause changes to the atmosphere, which in turn affect the propagation conditions of waves. Attenuation is frequency-dependent and has a considerable affect on some frequencies, and a lesser one on others. For example, in the higher-frequency ranges above about 12 GHz attenuation is strong when it is foggy or raining because of the scattering and absorption of electromagnetic waves on drops of water.

Figure 2.4 shows the frequency-dependent attenuation of radio waves with horizontal free-space propagation in which, as applicable, the appropriate attenuation values for fog (B) or rain of different intensity (A) still need to be added to the gaseous attenuation (curve C). What is remarkable are the resonant local attenuation maxima caused by water vapour (at 23, 150, etc., GHz) or oxygen (at 60 and 110 GHz).

Based on 60 GHz as an example, Figure 2.5 shows the propagation attenuation and the energy per symbol E_s related to N_0 (noise power), referred to as the signal-to-noise ratio, for antenna gain of $g_T = g_R = 18$ dB. These gains are achieved with directional antennas with approximately $20° \cdot 20°$ beam angles. The electric transmit power in the example is 25 mW, thereby producing the value 2 dBW = 1.6 W for the radiated microwave power (EIRP). The ranges which can be achieved are 800 m in good weather conditions and 500 m in rainy conditions (50 mm/h).

2.1.2 Propagation over Flat Terrain

Free-space propagation is of little practical importance in mobile communications, because in reality obstacles and reflective surfaces will always appear in the propagation path. Along with attenuation caused by distance, a radiated wave also loses energy through reflection, transmission and diffraction due to obstacles.

A simple calculation [27] can be carried out for a relatively simple case scenario: two-path propagation over a reflecting surface (see Figure 2.6). In this case

$$\frac{P_R}{P_T} = g_T g_R \left(\frac{h_1 h_2}{d^2}\right)^2$$

$d \gg h_1, h_2$ is a frequency-independent term. The corresponding path loss L_P is

$$L_P = -10\log g_T - 10\log g_R - 20\log h_1 - 20\log h_2 + 40\log d$$

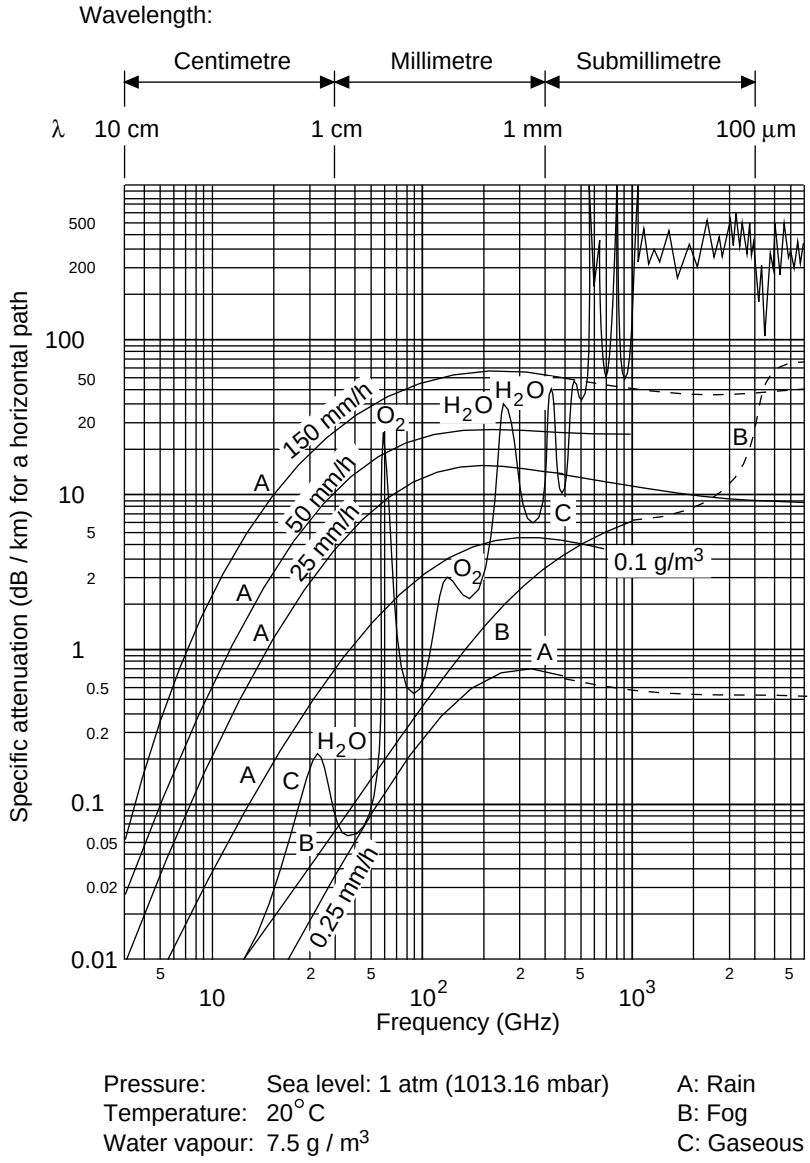


Figure 2.4: Attenuation of radio propagation depending on the frequency due to gaseous constituents and precipitation for transmission through the atmosphere, (from CCIR Rep. 719, 721)

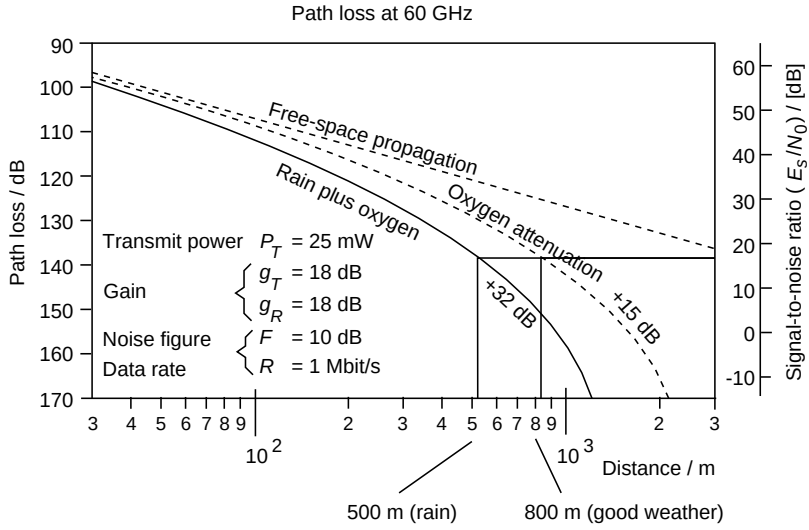


Figure 2.5: Attenuation due to weather conditions

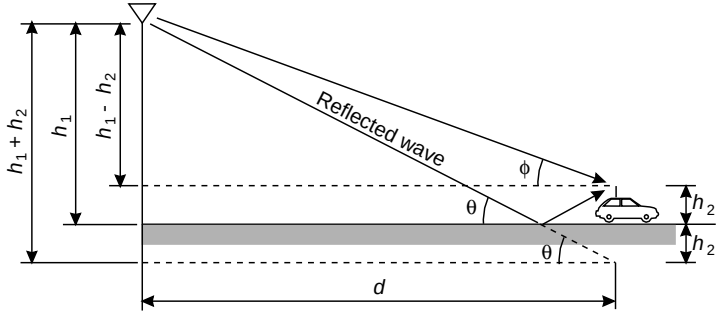


Figure 2.6: Model for two-path propagation due to reflection

and with isotropic antennas

$$\frac{L_F}{\mathrm{dB}} = 120 - 20\log\frac{h_1}{\mathrm{m}} - 20\log\frac{h_2}{\mathrm{m}} + 40\log\frac{d}{\mathrm{km}} \tag{2.7}$$

In this model the receive power decreases much faster ($\sim 1/d^4$) than with free-space propagation ($\sim 1/d^2$). This also depicts the reality of a mobile radio environment more closely but does not take into account the fact that actual ground surfaces are rough, therefore causing wave scattering in addition to reflection. Furthermore, obstacles in the propagation path and the type of buildings that exist have an impact on attenuation.

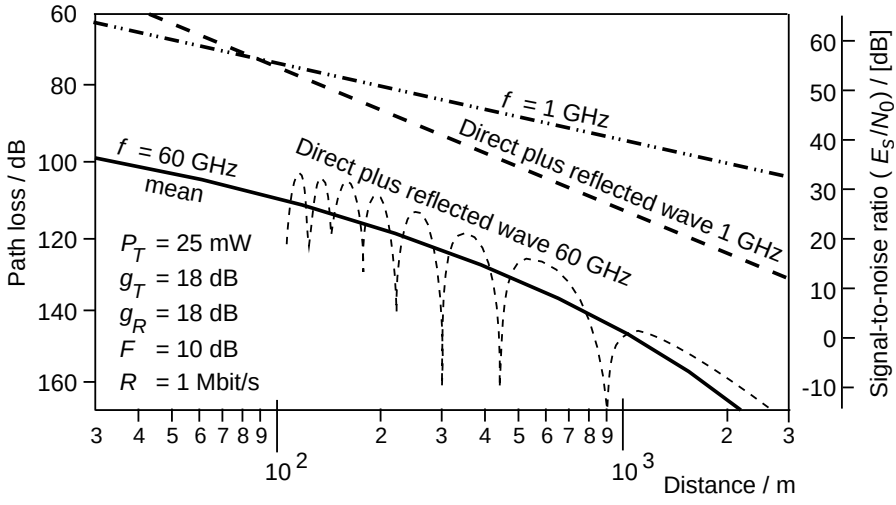


Figure 2.7: Propagation attenuation in two-path model taking into account O_2 absorption

With the introduction of the propagation coefficent γ, the following applies to isotropic antennas:

$$P_R = P_T g_T g_R \left(\frac{\lambda}{4\pi}\right)^2 \frac{1}{d^\gamma} \tag{2.8}$$

Realistic values for γ are between 2 (free-space propagation) and 5 (strong attenuation, e.g., because of city buildings).

Different models can be used for calculating the path loss based on these parameters, and are presented in Section 2.2.

Figure 2.7 compares the resulting propagation attenuation at 1 GHz and at 60 GHz, taking into account O_2 absorption and interference caused by two-path propagation. This interference leads to signal fading in sharply defined geographical areas, and this is also relevant within the transmission range.

2.1.3 Fading in Propagation with a Large Number of Reflectors (Multipath Propagation)

Fading refers to fluctuations in the amplitude of a received signal that occur owing to propagation-related interference. Multipath propagation caused by reflection and the scattering of radio waves lead to a situation in which transmitted signals arrive phase-shifted over paths of different lengths at the receiver and are superimposed there. This interference can strengthen, distort or even eliminate the received signal. There are many conditions that cause fading, and these will be covered below.

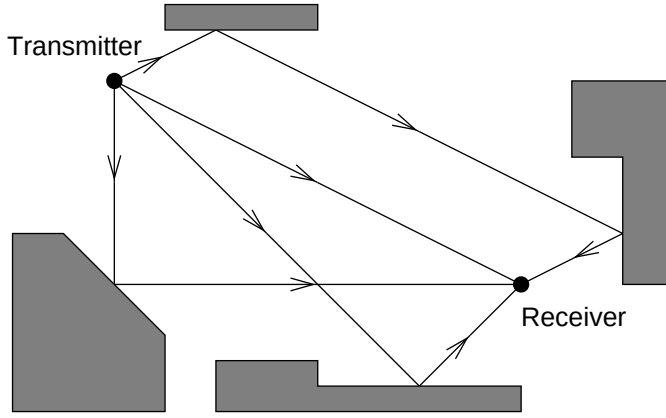


Figure 2.8: Multipath propagation

In a realistic radio environment waves reach a receiver not only over a direct path but also on several other paths from different directions (see Figure 2.8). A typical feature of multipath propagation (frequency-selective with broadband signals) is the existence of drops and boosts in level within the channel bandwidth that sometimes fall below the sensitivity threshold of the receiver or modulate it beyond its linear range.

The individual component waves can thereby superimpose themselves constructively or destructively and produce a stationary signal profile, referred to as multipath fading, which produces a typical signal profile on a path when the receiver is moving, referred to as short-term fading (see Figure 2.9).

The different time delays of component waves result in the widening of a channel's impulse response. This dispersion (or *delay spread*) can cause interference between transmitted symbols (*intersymbol interference*).

Furthermore, depending on the direction of incidence of a component wave, the moving receiver experiences either a positive or a negative Doppler shift, which results in a widening of the frequency spectrum.

In general the time characteristics of a signal envelope pattern can be described as follows:

$$r(t) = m(t)r_0(t) \tag{2.9}$$

Here $m(t)$ signifies the current mean value of the signal level and $r_0(t)$ refers to the part caused by short-term fading. The local mean value $m(t)$ can be deduced from the overall signal level $r(t)$ by averaging $r(t)$ over a range of 40–200 λ [21].

The receive level can sometimes be improved considerably through the use of a diversity receiver with two antennas positioned in close proximity to each other ($n \cdot \lambda/2$; $n = 1, 2, \ldots$). Because of the different propagation paths of the radio waves, the receiving minima and maxima affected by fading of both antennas occur at different locations in the radio field, thereby always enabling

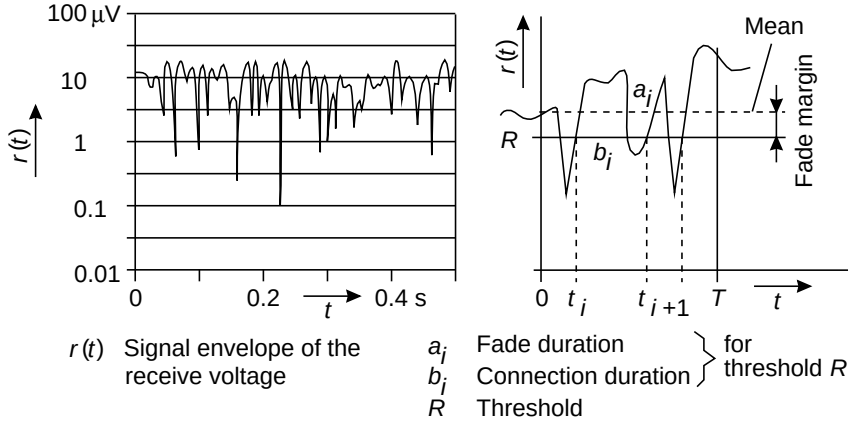


Figure 2.9: Receive signal voltage at a moving terminal under multipath fading (overall and in detail)

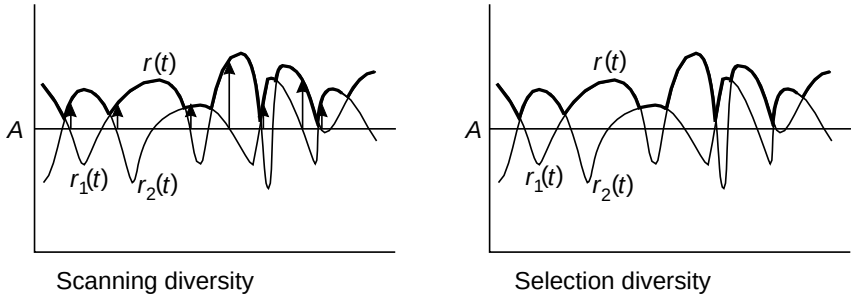


Figure 2.10: Diversity reception

the receiver to pick up the strongest available receive signal. See Figure 2.10, which shows the signal profile $r_i(t)$ of two antennas and the receive signal $r(t)$. With *scanning diversity* an antenna is replaced by a prevalent antenna when its signal level drops below a threshold A. With *selection diversity* it is always the antenna with the highest signal level that is used.

2.1.4 A Statistical Description of the Transmission Channel

It is only possible to provide a generic description of a transmission channel on the basis of a real-life scenario. In the frequency range of mobile radio being considered, changes such as the movement of reflectors alter propagation conditions. Signal statistics is another way of developing a mathematical understanding of the propagation channel.

2.1.4.1 Gaussian Distribution

The distribution function resulting from the superposition of an infinite number of statistically independent random variables is, based on the central limit theorem, a Gaussian function:

$$p(x) = \frac{1}{\sqrt{2\pi}\sigma} e^{-\frac{(x-m)^2}{2\sigma^2}} \tag{2.10}$$

No particular distribution function is required for the individual overlaid random variables, and they can even be uniformly distributed. The only prerequisite is that the variances of the individual random variables should be small in comparison with the overall variance.

A complete description of the Gaussian distribution is provided through its mean value m and the variance σ^2.

2.1.4.2 Rayleigh Distribution

On the assumption that all component waves are approximately incident at a plane and approximately have the same amplitude, a Rayleigh distribution occurs for the envelope of the signal. This assumption applies in particular when the receiver has no line-of-sight connection with the transmitter because of the lack of dominance of any particular component wave (see Figure 2.8).

The distribution density function of the envelope $r(t)$ is

$$p(r) = \frac{r}{\sigma^2} e^{-\frac{r^2}{2\sigma^2}} \tag{2.11}$$

with the mean value, quadratic mean value and variance

$$E\{r\} = \sigma\sqrt{\frac{\pi}{2}}, \qquad E\{r^2\} = 2\sigma^2, \qquad \sigma_r^2 = \sigma^2\left(\frac{4-\pi}{2}\right)$$

For the representation with $r(t) = m(t) \cdot r_0(t)$ a normalization of $E\{r_0^2\} = 1$ is common and useful. The logarithmic representation with $y = 20 \log r_0$ therefore produces

$$p(y) = \frac{10^{y/10}}{20 \log e} e^{-10^{y/10}}$$

with the mean value, variance and standard deviation ($C = 0.5772\ldots$ is Euler's constant)

$$E\{y\} = -C \cdot 10 \log e = -2.51 \text{ dB}$$

$$\sigma_y^2 = (10 \log e)^2 \pi^2/6 = 31.03 \text{ dB}, \qquad \sigma_y = 5.57 \text{ dB}$$

Figure 2.11 illustrates the distribution in half-logarithmic scaling.

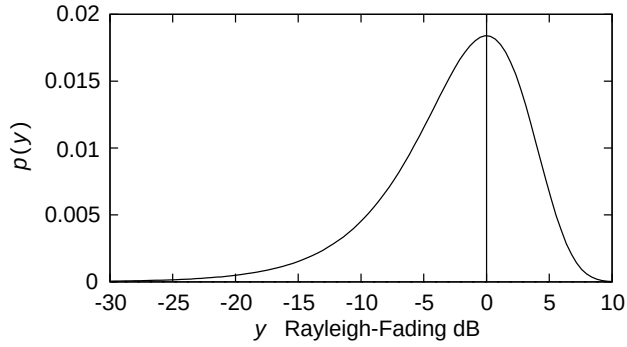


Figure 2.11: Rayleigh distribution function (dB)

Fading frequency The frequency of fading, which can be of the order of about 30 to 40 dB in depth, is dependent on the speed at which the receiver is moving, and can be described on the basis of the Doppler shift of the transmit frequency. The rate N_R at which the prescribed field strength level is exceeded is therefore calculated from

$$N_R = \sqrt{2\pi} f_m \rho e^{-\rho^2} \tag{2.12}$$

with f_m standing for the quotient arising from the vehicle speed v and wavelength λ

$$f_m = v/\lambda \tag{2.13}$$

and ρ indicating the relationship between the received signal level and the mean level.

Because the quadrature and in-phase components of the transmitted signal are Gaussian-distributed and the field strength follows a Rayleigh distribution, the signal fluctuations that arise due to multipath propagation are also referred to as *Rayleigh fading*.

The propagation paths are all of different lengths and have different reflection and transmission coefficients on the respective obstacles. This causes phase shifts on the individual incoming paths.

Signal fading due to Rayleigh fading occurs at intervals of the order of half the wavelength, $\lambda/2$.

Taking into account the attenuation and the multipath propagation with the complex elements of all the paths, the following attenuation can be observed in buildings according to [29]:

$$L = -20 \log \left(\frac{\lambda}{4\pi} \left| \sum_{i=0}^{n} \frac{\Gamma_i}{d_i} e^{\frac{2\pi d_i}{\lambda}} \right| \right) \tag{2.14}$$

L	overall attenuation in dB	n	number of incoming paths
d_i	length of ith path	λ	wavelength

Table 2.1: Parameter values for the Rice distribution

	$d \leq 6\,\text{km}$		$d > 6\,\text{km}$	
Environment	r_s	$K = r_s^2/\sigma^2$	r_s	$K = r_s^2/\sigma^2$
Woodland	0.40	0.25	0.16	0.04
Small town	0.63	0.76	0.39	0.27
Village	0.74	1.15	0.40	0.24
Hamlet	0.81	1.61	0.77	1.35
Minor road	0.77	1.19	0.75	0.96
B-road	0.78	1.23	0.74	0.92
A-road	0.86	1.37	0.55	0.55

Γ_i takes account of the reflections and transmissions experienced by the ith ray on the path between transmitter and receiver:

$$\Gamma_i = \prod_{j=0}^{r} R_j \prod_{k=0}^{t} T_k \tag{2.15}$$

R_j	jth reflection factor of ith route	r	number of reflections on jth path
T_k	kth transmission factor of ith route	t	number of reflections on kth path

2.1.4.3 The Rice Distribution

There are many cases in which the assumption of component waves having the same amplitude does not apply, especially when a line-of-sight connection dominates. The envelope is then described on the basis of a Rice distribution.

The distribution density function for the envelope $r(t)$ produces

$$p(r) = \frac{r}{\sigma^2} e^{-\frac{r^2+r_s^2}{2\sigma^2}} \mathrm{I}_0\left(\frac{r r_s}{\sigma^2}\right) \tag{2.16}$$

with I_0 is the Bessel function of 1st type and 0th order. The Rayleigh distribution is a special case of the Rice distribution for $r_s = 0$.

In concrete terms, r_s^2 represents the power of the direct, dominant component wave, and σ^2 that of the randomly distributed multipath component waves. Signal fades occur at longer intervals the further away the receiver is from the transmitter; see Figure 2.7.

Reference [24] contains parameter values for several measurements in rural areas (see Table 2.1). The values relate to the normalized signal envelope $r(t) = m(t)r_0(t)$ and a dB-mean value of 0 for r_0. Depending on the environment, σ^2 is clearly less than with Rayleigh fading. $K = r_s^2/\sigma^2 = 0$ corresponds to $r_s = 0$, i.e., there is no line-of-sight connection. $K \rightarrow \infty$ means that no multipath signals are being received. Figure 2.12 shows the Rice distribution for $\sigma = 1$.

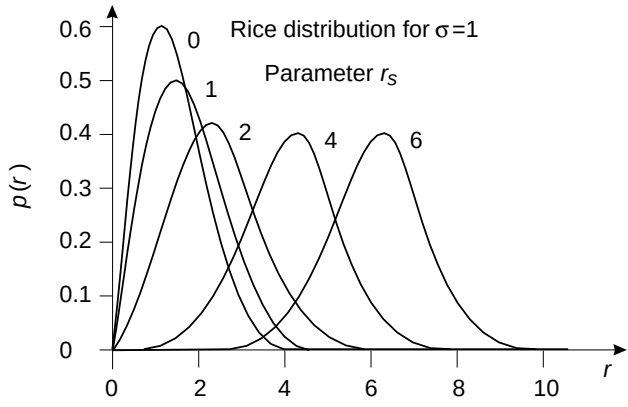


Figure 2.12: Rice distribution density function

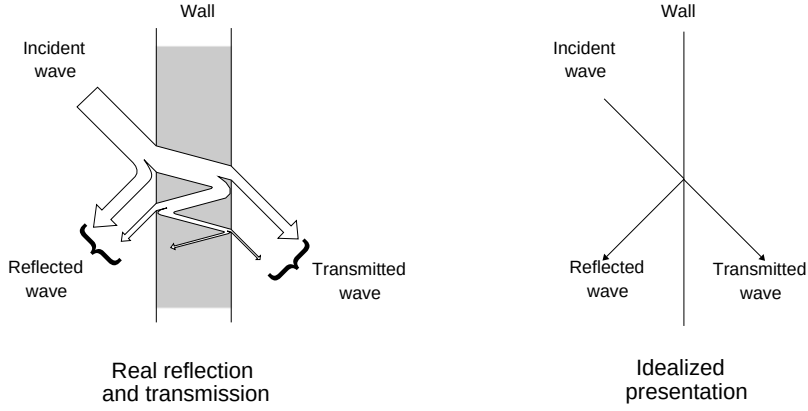


Figure 2.13: Reflection at a wall

There is no closed-form solution for the mean value and variance for the Rice distribution density function. These parameters can only be determined using approximation formulas and tables.

2.1.5 Reflection

Waves are completely reflected on smooth surfaces, but otherwise they are only partially reflected because of partial absorption—something that results in undesirable phase shifts.

If a propagating wave hits a wall, part of it is reflected and part transmitted, as is shown in Figure 2.13. The reflected part is a result of direct reflection and a multitude of multiple reflections on the inside of the wall. In this same

way the entire transmitted part consists of one direct continuous wave and many component waves reflected in the wall; see[19].

The sum total of the reflected and the transmitted wave differs from the incident wave because the multiple reflections within the wall cause attenuation loss.

In the prediction of actual radio propagation (e.g., using ray tracing techniques) it is usually the geometric conditions of reflection and transmission on a wall—albeit in the idealized form presented in Figure 2.13—that are taken into account.

Geometric errors can occur for the following reasons:

1. Owing to refraction, the exit point of the transmitted wave on the inside of the wall is shifted vertically from the exit point in the simplified representation.

2. The parts resulting from multiple reflections do not actually exit from the wall at the same place as the direct wave.

3. The point of reflection is fixed on the idealized wall and is therefore misaligned by half the thickness of the wall from the actual point of reflection.

According to [18], the reflection and transmission of an electromagnetic wave on a dielectric layer are described as follows:

$$R_{Wall} = \frac{r(e^{-2j\psi} - 1)}{e^{-2j\psi} - r^2}, \qquad T_{Wall} = \frac{1 - r^2}{e^{-j\psi} - r^2 e^{j\psi}} \tag{2.17}$$

with

$$\psi = \frac{2\pi d}{\lambda}\sqrt{\underline{\varepsilon}_r - \sin^2\varphi} \tag{2.18}$$

and

$$r_\perp = \frac{\cos\varphi - \sqrt{\underline{\varepsilon}_r - \sin^2\varphi}}{\cos\varphi + \sqrt{\underline{\varepsilon}_r - \sin^2\varphi}}, \qquad r_\parallel = \frac{\underline{\varepsilon}_r\cos\varphi - \sqrt{\underline{\varepsilon}_r - \sin^2\varphi}}{\underline{\varepsilon}_r\cos\varphi + \sqrt{\underline{\varepsilon}_r - \sin^2\varphi}} \tag{2.19}$$

R_{Wall}	complex reflection factor	λ	wavelength
T_{Wall}	complex transmission factor	d	thickness of wall
$\underline{\varepsilon}_r$	complex dielectric coefficient	φ	angle of incidence

The expressions in Equation (2.19) represent the reflection behaviour on an ideal thin layer, with $r_\perp$ describing the behaviour in vertical polarization and $r_\parallel$ in parallel polarization.

The reflection curves calculated using Equation (2.17) and illustrated in Figures 2.14 and 2.15 closely resemble those shown in [19]. No measurement results are available for the transmission values (see Figures 2.16 and 2.17); they are deduced from the reflection coefficients.

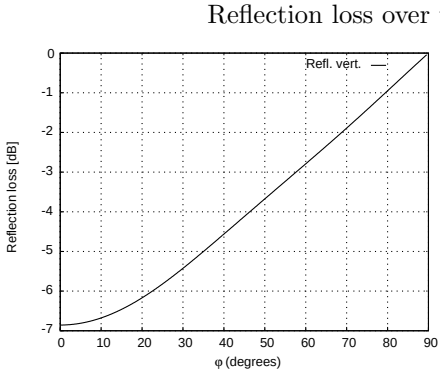


Figure 2.14: Concrete wall (wall thickness 150 mm), vertical polarization

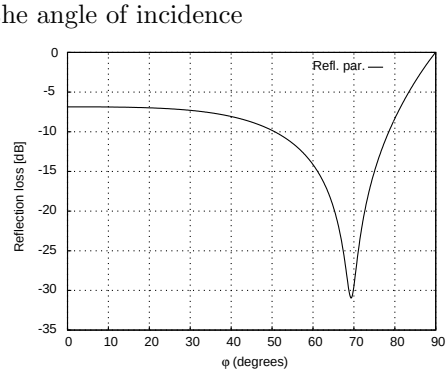


Figure 2.15: Concrete wall (wall thickness 150 mm), horizontal polarization

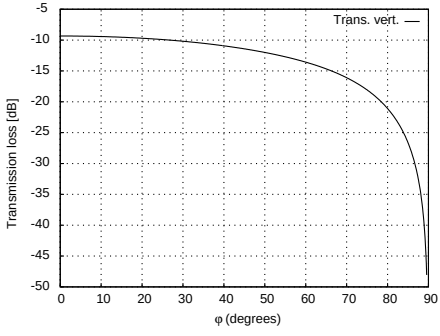


Figure 2.16: Concrete wall (wall thickness 150 mm), vertical polarization

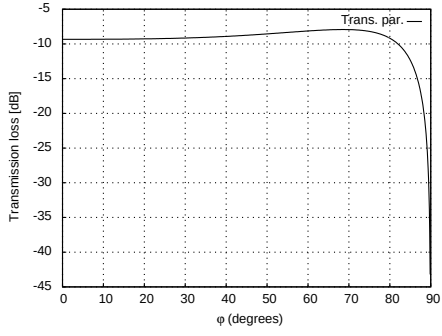


Figure 2.17: Concrete wall (wall thickness 150 mm), horizontal polarization

The figures show the attenuation of the reflection or the transmission over the angle of incidence φ, with the attenuation being $20\log|R_{Wall}|$ [dB] and $20\log|T_{Wall}|$ [dB] respectively.

The results for the different polarization directions as a function of the angle of incidence indicate a sharp drop in the *Brewster angle* area. Otherwise the reflection factor increases from a minimum value of 0° to a maximum value of almost 90°. The minimum value, the gradient of the curve and the *Brewster angle* are dependent on the thickness and material of the wall.

The reflection characteristics of different materials in the area of 1–20 GHz are presented as attenuation curves in [19].

2.1.6 Diffraction

Diffraction describes the modification of propagating waves when obstructed. A wave is diffracted into the shadow space of an obstruction, thereby enabling it to reach an area that it could ordinarily only reach along a direct path through transmission.

The effect of diffraction becomes greater as the ratio of the wavelength to the dimension of the obstacle increases. Diffraction is negligible at frequencies above around 5 GHz.

2.1.7 RMS Delay Spread

The RMS (*root mean square*) delay spread describes the dispersion of a signal through multipath propagation and takes into account the time delays of all incoming paths with relation to the first path. The respective paths are weighted with their received level:

$$\tau_{rms} = \sqrt{\frac{1}{\sum_{i=1}^{n} P_i} \sum_{i=1}^{n} (\tau_i^2 P_i) - \tau_d^2}, \qquad \text{with} \quad \tau_d = \frac{\sum_{i=1}^{n} (\tau_i P_i)}{\sum_{i=1}^{n} P_i} \tag{2.20}$$

τ_{rms}	RMS delay spread	P_i	received level of path i
τ_i	time delay of ith path	n	# incoming paths

If the value of the RMS delay spread exceeds the tolerance limits of a system, it is assumed that error-free reception is no longer possible. When this happens, the waves travel over considerably different long paths, the levels of which are not negligible. If the resultant time dispersion of the signal is greater than the symbol duration during transmission then the receiver experiences intersymbol interference and bit errors.

2.1.8 Shadowing

Obstacles in the line-of-sight path between transmitter and receiver outdoors (mountains and buildings) or inside buildings (walls) hinder direct wave propagation and therefore prevent the use of the shortest and frequently least interfered (strongest) path between transmitter and receiver, and cause additional attenuation to the signal level, which is called *shadowing*. Shadowing causes fluctuations to the signal level over a distance that, at 900 MHz for example, can be of the order of around 25–100 m. *Long-term fading* occurs when a moving receiver is lingering for a long time in the radio shadow, e.g., for 10 to 40 s.

Measurements have revealed that the local mean value $m(t)$ in Equation (2.9) follows a lognormal distribution, i.e., $L_m = \log m(t)$ is normally distributed with a standard deviation of approximately 4 dB [21, 27]. This

is also called *lognormal fading*. This approximation applies to statistics for large built-up areas.

2.1.9 Interference Caused by Other Systems

In addition to the interference caused by radio wave propagation, which has already been discussed, there is also secondary interference, such as the reciprocal effects of neighbouring radio systems on adjacent channels in the spectrum and electromagnetic impulses caused by other systems, car starters, generators and PCs—in other words *man-made noise*.

2.2 Models to Calculate the Radio Field

Reliable models for the calculation of expected signal levels are needed in the planning of radio networks, establishing of supply areas and siting of base stations. Data on terrain structure (topography) and buildings and vegetation (morphology) are required for these calculations.

Radio propagation in a mobile radio environment can be described on the basis of three components: long-term mean value, shadowing and short-term fading. The sum total of these components $L_P = L_l + L_m + L_s$ describes the resultant overall path loss between transmitter and receiver; see Figure 3.45.

Another factor to be considered is that mobile stations usually move at different speeds. The level, e.g., for determining GSM radio measurement data, is measured on a time-related basis, so that the level is also affected by the speed at which the mobile station is moving.

In the measurements by Okumura [25] the long-term mean value describes the level value averaged over a large physical area of 1–1.5 km. The effects of shadowing and short-term fading disappear through the averaging. This long-term mean value can be calculated using approximate models.

A description of the most common models used in calculating the mean value of the expected radio levels follows. A distinction is made between empirical models, which are based on measurement data, and theoretical models, which are based on the use of wave diffraction.

2.2.1 Empirical Models

The empirical approach is based on measurement data that when plotted as regression curves or analytical expressions can be used to calculate signal levels. The advantage of these models is that because of their measurement basis, they all take into account known and unknown factors of radio propagation. The disadvantage is that the models only cover certain frequencies and scenarios and sometimes have to be revalidated for other areas.

Reference [27] offers an overview of the different measurements and the models derived from them.

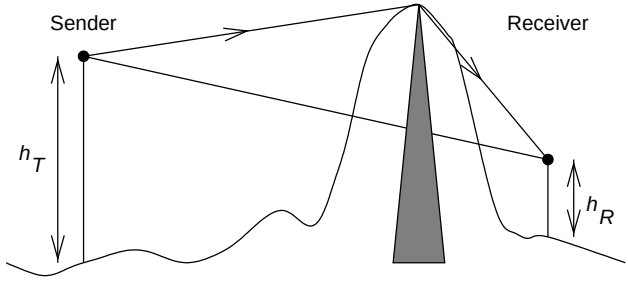


Figure 2.18: Obstacle in terrain as diffraction edge

2.2.2 Diffraction Models

Diffraction theory can be used to obtain a description of radio propagation. In this case obstacles in uneven terrain are modelled as diffraction edges. A section of terrain of the line of sight, which can usually be obtained from a topographical database, is required for calculating diffraction loss. Figure 2.18 illustrates the principle for an edge.

For less steep forms of terrain, such as hills, cylinder diffraction can be used as a model to produce better results. All types of terrain must be represented using several diffraction edges. Many different methods are available for calculating the resultant diffraction loss (see the overview in [27]).

Diffraction models have the advantage that they can be calculated without reference to any particular frequency or scenario, and consequently, in comparison with empirical models, can be used in a wider range of application (frequencies, distances). The disadvantages are that the accuracy of the calculation depends strongly on the accuracy of the topographical database and that the different approaches produce widely different results for terrains with several obstacles.

Because morphology plays an important role in the calculation of radio propagation, empirical correction factors are also required for the diffraction models. In practice, therefore, hybrid calculation methods are used with radio network planning tools.

2.2.3 Ray Tracing Techniques

The long-term mean value of a signal level can be calculated using empirical models and diffraction models. Some applications, such as the calculation of radio propagation in networks with microcells (<1 km radio range), require calculations with a more accurate resolution. The surroundings of the mobile and base station (such as the geometry of the buildings) must be taken into account if there is an interest in more than just the mean value over a large area.

There are empirical and theoretical approaches to carrying out calculations. Good results can be achieved using *ray tracing* techniques, which can, however, prove to be complicated for larger scenarios.

2.2.4 The Okumura/Hata Model

In 1962/63 and in 1965 Okumura carried out extensive measurements in and around Tokyo for the frequency range from 500 MHz to 2 GHz. The results of these measurements were published as regression curves [25]. To simplify radio field predictions, Hata linearized some of these curves and approximated them through analytical equations [12].

The basis for the calculation is an equation for the path loss in relatively flat terrain with city dwellings (see Section 2.2.4.1) and isotropic antennas:

$$\begin{aligned}\frac{L_P}{\mathrm{dB}} &= 69.55 + 26.16\log\frac{f}{\mathrm{MHz}} - 13.82\log\frac{h_T}{\mathrm{m}} \\ &\quad -a(h_R) + \left(44.9 - 6.55\log\frac{h_T}{\mathrm{m}}\right)\log\frac{d}{\mathrm{km}} \end{aligned} \tag{2.21}$$

This equation applies to frequencies f from 150 to 1500 MHz, (effective) transmitting antenna heights h_T from 30 to 200 m and distances d from 1 to 20 km. $a(h_R)$ is a correction factor for the height of the receiving antenna h_R.

Based on a frequency of 900 MHz, a 30 m base station antenna height and a 1.5 m mobile station antenna height in a medium-sized city ($a(h_R) \approx 0$), Equation (2.21) gives

$$\frac{L_P}{\mathrm{dB}} = 126.42 + 35.22\log\frac{d}{\mathrm{km}}$$

A comparison with free-space propagation (also see Section 2.1)

$$\frac{L_P}{\mathrm{dB}} = 91.52 + 20\log\frac{d}{\mathrm{km}}$$

and with propagation over a flat area (Equation (2.7))

$$\frac{L_P}{\mathrm{dB}} = 86.94 + 40\log\frac{d}{\mathrm{km}}$$

shows the path loss to be clearly higher than in these two theoretical models; with a value of 3.5, the propagation exponent is somewhat lower than it is with propagation over a flat area.

2.2.4.1 Terrain Profile

Okumura distinguishes between two types of terrain:

- *Quasi-flat terrain*
  Flat terrain with a maximum 20 m altitude difference.

- *Irregular terrain*
  All other irregular types of terrain, divided into:
  - *Rolling hilly terrain*
  - *Isolated mountain*
  - *General sloping terrain*
  - *Mixed land–sea path*

Diagrams with correction factors for calculating the radio propagation for each of these categories are provided.

2.2.4.2 Types of Morphology

Okumura refers to three types of morphology:

- *Open area*
  Open area without any major obstacles: farmland, etc.
- *Suburban area*
  City or town with houses and trees, low-density housing
- *Urban area*
  City with high rises or at least two-storey high buildings

The diagrams with the correction factors for the *open* and *suburban* morphology types in Okumura's work have been approximated into formulas by Hata. In each case the correction factors must be added to the basic path loss Equation (2.21):

Suburban

$$\frac{K_{\text{suburban}}}{\text{dB}} = -2\left(\log\frac{f}{28\text{MHz}}\right)^2 - 5.4$$

Open

$$\frac{K_{\text{open}}}{\text{dB}} = -4.78\left(\log\frac{f}{\text{MHz}}\right)^2 + 18.33\log\frac{f}{\text{MHz}} - 40.94$$

The values for a frequency of 900 MHz are $K_{\text{suburban}} = -10$ dB, $K_{\text{open}} = -28.5$ dB, and for 1800 MHz they are $K_{\text{suburban}} = -12$ dB, $K_{\text{open}} = -32$ dB.

2.2.5 Radio Propagation in Microcells

Today's radio networks use hierarchical cell structures with small microcells below the level of conventional macrocells. This increases network capacity in areas with high traffic volumes (city centres, trade fairs, etc.).

Microcells cover areas extending up to several 100 m, with radio illumination greatly affected by the geometry of the base station surroundings. The Okumura/Hata model only lends itself to the calculation of the mean value for large areas and cell sizes with a radius of several kilometres. Other methods are required for calculating radio illumination in microcells [26, 36].

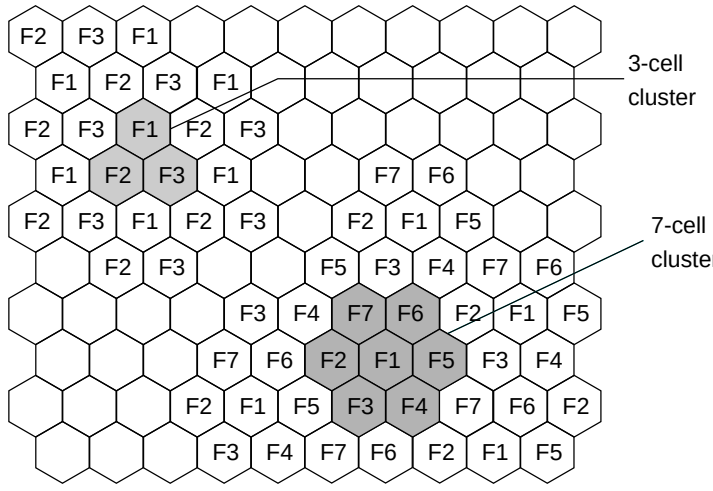


Figure 2.19: A cellular network supplying an extended area

2.3 Cellular Systems

Traditional radio networks (1st-generation) that try to provide coverage to large areas by increasing the transmitting power of the individual base stations are only able to serve a limited number of subscribers because of the bandwidth used. In these radio networks an allocated radio channel is retained as long as possible, even if a receiver has already moved into another supply area. Because the physical boundaries of a supply area are not precisely defined, neighbouring supply areas must use different radio channels in order to avoid interference. Where user density is high, this results in a heavy demand for frequencies, which, however, are restricted because of the scarcity of available spectrum.

The poor utilization of frequency spectrum in these radio networks and the increase in the number of mobile radio users, which these systems could no longer handle, led to the development of cellular networks.

Cellular networks are based on dividing the entire area over which a network is to be operated into radio cells, each of which is served by a base station. Each base station is only allowed to use a certain number of the total available frequency channels, which can only be reused after a sufficient interval to avoid any interference from neighbouring cells (see Figure 2.19). F_i $(i = 1, 2, \ldots, n)$ indicates the groups of frequency channels used in the respective cells.

In cellular networks the low transmitting power of the base stations enables them to use allocated frequencies only in a strictly defined area of the radio cell, thereby allowing these frequencies to be reused after a predetermined reuse distance.



Figure 2.20: Simulated best server cells of a metropolitan scenario

Cells are generally represented as idealized regular hexagons, but because of topographical and environmental conditions, this is only an approximation of what actually occurs; see Figure 2.20. In reality radio cells are very irregular in their external shape and furthermore are designed to overlap with one another by approximately 10–15%. This enables mobile stations operating near the boundary of a cell to choose with which base station they set up a call.

2.3.1 Cluster Patterns and Carrier-to-Interference Ratio

Radio cells are combined into *clusters*, and each frequency is used once per cluster. Cells of a cluster neighbouring any other cluster may reuse all frequencies occupied in the cluster in a regular way and thereby repeat the cluster. Because the clusters must supply an entire area with coverage, only certain cluster patterns consisting of, e.g., 3, 4, 7, 12, 15 or 21 cells, are possible. The lower the number of cells per cluster, the greater the number of radio-frequency channels that can be used per cell for the entire mobile radio system in a given frequency band. However, frequencies reused within a short distance increase the co-channel interference across clusters. The carrier-to-interference ratio is the ratio of the received *carrier signal* C to the received *interference signal* I from the co-channels.

Figure 2.21 shows an example of frequency planning using a 3-cell cluster (frequency groups 1, 2, 3) and a regular hexagonal cell structure. This kind of uniform cluster structure always has six co-channel cells in close proximity to any base station. The interference usually originates from these cells; co-channel cells further away will be of less harm.

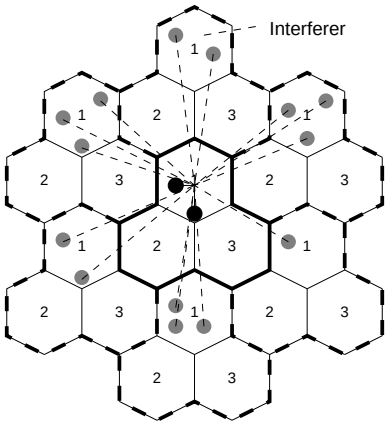


Figure 2.21: Interference by co-channel cells

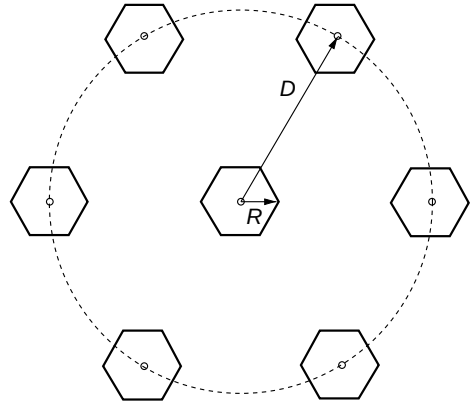


Figure 2.22: Co-channel cells in a cellular network

The distribution of interference level is caused by the overlapping of many independent sources of interference and therefore can be regarded as a normal distribution. On the downlink the base stations are the source of interference; on the uplink it is the mobile stations from the co-channel cells.

2.3.2 C/I Ratio and Interference-Reduction Factor

Depending on the modulation technique employed and the technical equipment of the receiver, the smallest acceptable C/I ratio for a cellular network is based on the number of co-channel cells, the distance between these cells, the transmitting power and the terrain characteristics.

The C/I ratio required for cell planning is calculated below on the basis of an idealized concept for the model. The calculation considers a system such as the one in Figure 2.22. In the central cell with radius R a mobile station is in radio contact with its base station. The interference cells in which the mobile and base stations are transmitting on the same frequency and therefore can cause interference are arranged in a circle around the cell at distance D. Co-channel interference by cells further away can be ignored because of the larger associated path loss. On the assumption that statistically all mobile stations transmit independently of one another, the total interference power for a base station with K_I interfering mobile stations is calculated on the basis of a combination of the individual interference contribution I_k and the noise power N:

$$I = \sum_{k=1}^{K_I} I_k + N \tag{2.22}$$

The noise power N in the receiver is

$$N = FK\Theta v \tag{2.23}$$

F stands for the noise factor of the receiver (which is between 5 and 10), K for Boltzmann's constant ($1.38 \cdot 10^{-23}$ W·s/K), Θ for the temperature in K and v for the bit rate in bit/s. The ratios in the model produce a noise power of approximately $5.6 \cdot 10^{-12}$ mW, or a level of -112 dBm.

If one ignores the noise power in a homogeneous system in which all mobile stations are transmitting at the same power then the co-channel carrier-to-interference ratio only depends on where the mobile stations are located in relationship to one another:

$$\frac{C}{I} = \frac{R^{-\gamma}}{\sum_{k=1}^{K_I} D_k^{-\gamma}} \tag{2.24}$$

In Equation (2.24) D_k indicates the distance to the kth mobile station and γ indicates the path loss parameter for the receive power. Because $K_I = 6$ cells are located at the same distance from the central cell in the model, and for purposes of simplification, all distances can be considered as equal, and it follows that

$$\frac{C}{I} = \frac{R^{-\gamma}}{6D^{-\gamma}} = \frac{1}{6}q^{\gamma} \tag{2.25}$$

The ratio $q = D/R$ is called the *co-channel interference reduction factor* and is

$$q = \frac{D}{R} = \left(6\frac{C}{I}\right)^{\frac{1}{\gamma}} \tag{2.26}$$

Aiming towards a co-channel carrier-to-interference ratio C/I of, e.g., 11 dB, using a realistic value of, e.g., $\gamma = 4$, produces a reduction factor of $q = 2.85$.

The next step in cell planning is the selection of a suitable cluster type to meet the requirement for a specific ratio of cell radius R to reuse distance D. If large clusters are used, the distance between the interfering cells will also be great; however, each cell will be able to carry less traffic because the trunk of frequency channels of the available band must be distributed among the clusters. Therefore, for maximum spectrum utilization efficiency, there has to be a balance between the number of cells per cluster and achievable transmission quality. The reduction factor produces the following actual values for the cluster sizes:

Cluster size	3	4	7	12
D/R	3	3.464	4.58	5.6

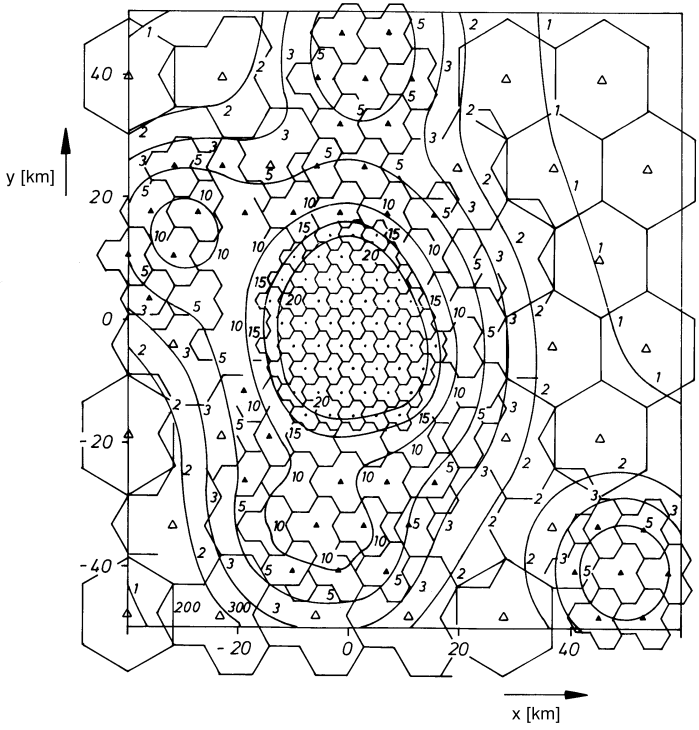


Figure 2.23: Adapting cell size to traffic density

2.3.3 Traffic Load and Cell Radius

Another parameter in radio network planning is the size of the cell radii. By reducing the size of the cells, a system can be adapted to cope with a higher level of user density (see Figure 2.23). However, the system dictates the minimum possible size of a cell derived from cost/benefit considerations. In practice networks are usually initially set up with relatively large cell radii. If it turns out that this structure is no longer able to carry existing traffic, the cells with high traffic density are reduced in size through the process of cell splitting and equipped with additional base transceiver stations. This approach always provides an optimal solution in terms of network utilization, necessary signalling for handover, service quality for the user, as well as costs of the network infrastructure. The number of base station locations can be reduced by setting up three neighbouring cells, with the cells serving three 120° sectors (= cells) with different channel groups (also see Section 2.4).

The main advantage of cellular systems is that the same radio channels can be reused, and consequently coverage can be provided to areas of any size.

The following requirements must be met to ensure that the system functions without any problems:

- Continual measurement of the field strength of the received signal ensures that a communicating mobile station is using the serving base station of the corresponding cell. The network is alerted as soon as a mobile station leaves a cell and the connection is automatically switched to the neighbouring base station. Cellular systems must therefore be capable of carrying out a change in radio channel as well as in base station during a connection. This process is called *handover* (see Section 3.6).
- A mobile radio network must be aware of which mobile radio users are currently roaming in its radio coverage area to enable it to page them if necessary. Mobile stations are therefore always assigned to a location area and reassigned to another area by the network when they change the service area. This service characteristic of cellular networks, which is called support of *roaming* (see Section 3.7.1), ensures that users can be reached at all times.

Complicated signalling protocols are required for operating cellular networks.

2.4 Sectorization and Spectral Efficiency

Cellular systems such as GSM are considered to be spectrum-efficient. Spectral efficiency relates the traffic capacity to frequency unit and surface element:

$$\text{Spectral efficiency} \left[\frac{\text{bit/s}}{\text{MHz} \cdot \text{km}^2}\right] = \frac{\text{Traffic [Erlang]}}{\text{Bandwidth} \times \text{Area}} \tag{2.27}$$

Traffic refers to all the traffic to be handled in a cellular system (i.e., the number of simultaneous voice and data connections), bandwidth is the total available channel capacity, and area refers to the total system area A [km^2].

The efficiency is dependent on the following parameters:

- Number of required radio channels per cell
- Cluster size or size of interference group

The sectorization of a given cluster reduces co-channel interference, because the power radiated backward from a directional base station antenna is very small and the number of interfering cells is reduced, i.e., the C/I ratio increases, owing to the directivity. On the other hand, the number of channels/sector drops and along with it the trunking gain.

2.4.1 Efficiency and Traffic Capacity

The following abbreviations are used in the next section:

V	average number of MS per km^2	n	number of channels per cell
t	traffic of a MS in Erlang	x	number of channels per sector
f_s	channel grid (30 kHz* in GSM)	S	number of sectors per cell
N	number of cells per cluster	A_s	sector area
Z	number of cells in system (A/A_h)	R	cell radius

The area of a hexagon with side length R is

$$A_h = \frac{3\sqrt{3}}{2}R^2 \tag{2.28}$$

This produces the following areas for three or six-sector cells:

$$A_3 = \frac{\sqrt{3}}{2}R^2, \qquad A_6 = \frac{\sqrt{3}}{4}R^2 \tag{2.29}$$

2.4.1.1 Measures for the Spectrum Utilization Efficiency

Two measures of spectrum utilization efficiency are to distinguish:

- Efficiency of a modulation method, which is usually expressed by the quotient of transmission rate [bit/s] and frequency bandwidth [Hz] used:

$$\eta_{Mod} = \frac{\text{Transmission rate of a channel}}{\text{Bandwidth per channel used}} \left[\frac{\text{bit/s}}{\text{Hz}}\right] \tag{2.30}$$

- Efficiency of a cellular system seen from the transmission specialist's viewpoint, where the transmission rate [bit/s] of the radio channel is normalized to the system bandwidth:

$$\eta_{cell} = \frac{\text{Transmission rate} \times \text{No. of channels in system}}{\text{System bandwidth}} \left[\frac{\text{bit/s}}{\text{Hz}}\right] \tag{2.31}$$

In both cases a bit error ratio results that is characteristic for the modulation scheme and other receiver parameters selected. The two views are more or less identical.

The teletraffic engineer has a different view on spectrum utilization efficiency. He relies on additional parameters to calculate the traffic capacity per square kilometre for a given system concept under some quality of service constraints, and is then able to compare different cellular concepts.

*A GSM frequency channel is 200 kHz wide and carries 8 TDMA channels. With 25 kHz per TDMA traffic channel and an additional 12 % of channel capacity for signalling, the result is approximately 30 kHz/TDMA channel.

According to Section 2.3.2, any calculations of the traffic capacity are possible only under the constraints of some C/I ratio. In [33] the spectrum utilization efficiency is claimed to be best characterized by calculation of the local traffic intensity per bandwidth unit; see Equation (2.27). The traffic intensity is exactly one Erlang if the radio channel is continuously being used. A channel being used up to $x\,\%$ only is carrying a traffic of $x/100$ Erlang. From this, the spectrum utilization efficiency can be defined as follows:

$$\eta = \frac{\text{No. of channels/Cell} \times \text{Offered traffic/Channel}}{\text{System bandwidth} \times \text{Area of a cell}} \left[\frac{\text{Erlang}}{\text{kHz} \cdot \text{km}^2}\right] \tag{2.32}$$

Here the system bandwidth is the product of the cluster size and the bandwidth per cell. The offered traffic differs from the carried traffic by the amount of traffic lost according to blocking of the system; see Appendix A.1.2.

Using the abbreviations introduced above, the spectrum efficiency for systems with sectorized and non-sectorized antennas can be defined, see [6]:

- With omnidirectional antennas, the maximum traffic capacity is the result of the traffic of a cell multiplied by the number of cells in the system:

$$T_{max} = VtA_hZ \tag{2.33}$$

  The efficiency is then

$$\eta_{Omni} = \frac{VtA_hZ}{2f_snNA} = \frac{Vt}{2f_snN}\left[\frac{\text{Erlang}}{\text{km}^2 \cdot \text{kHz}}\right] \tag{2.34}$$

- With sector antennas, the maximum traffic capacity is calculated as

$$T_{max} = VtA_sSZ \tag{2.35}$$

  The efficiency is then

$$\eta_{dir} = \frac{Vt}{2f_s(xS)N} \tag{2.36}$$

2.4.2 The Effect of Sectorization with a Given Cluster Size

Figures 2.24–2.26 show different clusters with 3-site and 6-site sectorization. The effects of sectorization can be examined on the basis of the following factors:

- Only the first ring of co-channel interference cells M is taken into account (see Figure 2.27). The effect of the second ring of interference cells is negligible.

- The propagation coefficient or attenuation factor is approximately $\gamma = 4$, so the received power is $P_R \sim P_T/R^4$.

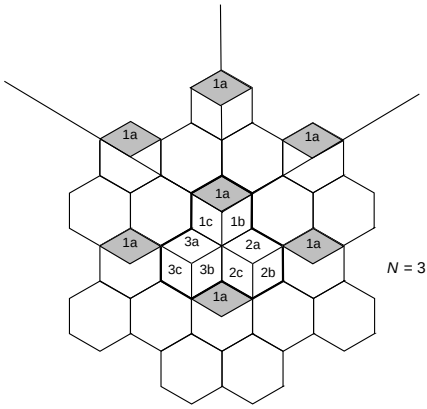


Figure 2.24: 3-sector cluster with 3-site sectorization

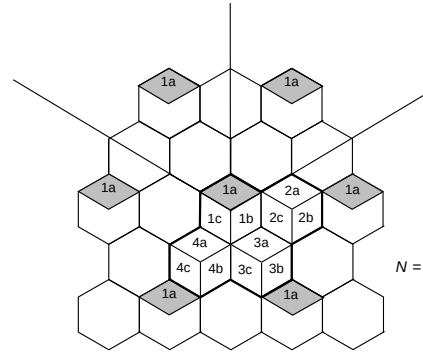


Figure 2.25: 4-sector cluster with 3-site sectorization

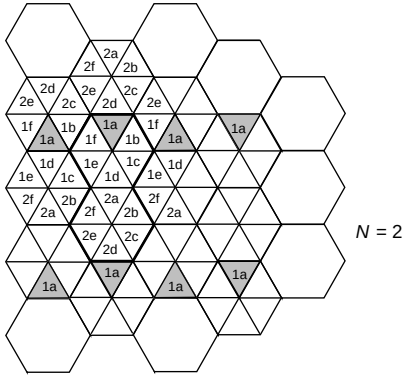


Figure 2.26: 2-sector cluster with 6-site sectorization

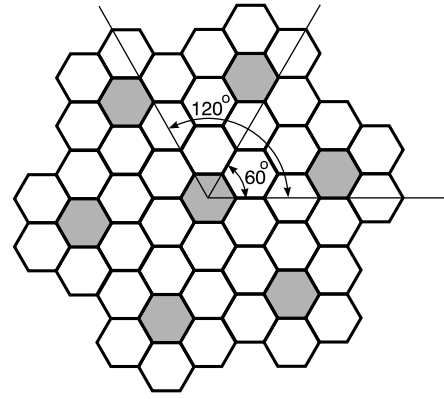


Figure 2.27: Distribution of co-channel interference cells

- The interference reduction factor, per Equation (2.26), is $q = 4.6$ for $N = 7$ (see the table in Section 2.3.2).

According to Figure 2.27, the relationship between the number of sectors and the number of interference cells for a 7-sector cluster is

$S \hat{=}$ number of sectors	$M \hat{=}$ number of co-channel cells
1 (omni)	6
3	2
6	1

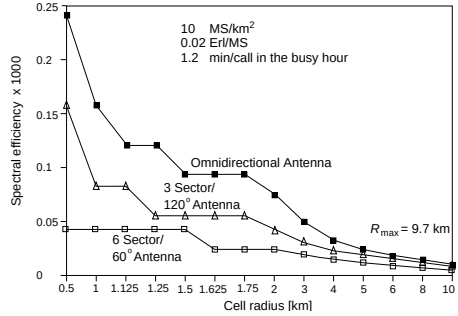


Figure 2.28: Spectral efficiency for $Vt = 0.2$ Erlang

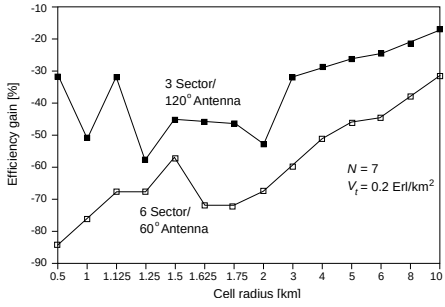


Figure 2.29: Efficiency gain with sectorized systems

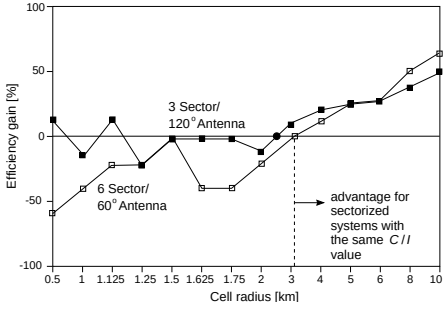


Figure 2.30: Efficiency gain with unchanged/reduced clusters

The same applies to other cluster sizes (see Section 2.3). Analogously to Equation (2.25), the C/I ratio for different clusters is [20]:

$$C/I = \frac{q^4}{M} = \frac{(3N)^2}{M} \tag{2.37}$$

The C/I ratio for the sectorizations being considered is therefore

$N = 7$	Omni	3-sector	6-sector
C/I [dB]	17	25.3	29

The spectral efficiency can therefore be calculated from Equations (2.34) and (2.36). Figure 2.28 shows the spectral efficiency over the cell radius for a traffic value of $Vt = 0.2$ Erlang/km^2, which corresponds to a traffic density of 10 MS/km^2 with a traffic volume of 0.02 Erlang per MS (in other words, each MS talks an average of 1.2 minutes per hour on the mobile telephone).

Figure 2.29 reveals a negative percentage increase in efficiency for sectorized systems found for the cell radii shown. According to Appendix A.1.2, the

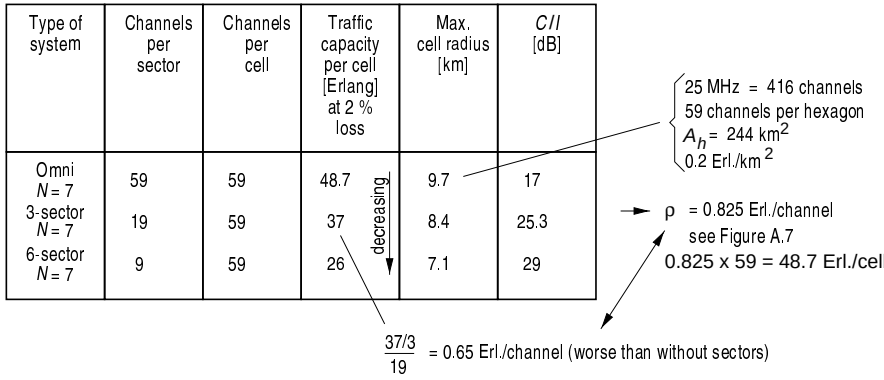


Figure 2.31: Maximum traffic capacity

traffic capacity per cell $Y = f(n, p_l)$ can be calculated analytically with given loss probability p_l:

$$Y = f(n, p_l) = VtA_sSp_l, \qquad \text{whereby } n = f(Vt, S, A_s)$$

For example, with a loss probability of 2 % it is possible to calculate the overall maximum traffic capacity for a system with a 2·12.5 MHz bandwidth or with 416 duplex channels; see Figure 2.31. The maximum cell radius for a traffic load per user of 0.2 Erlang/km^2 is also given.

In a 3-sector cell $n = 19$ channels result per sector, for which at $p_l = 2$ % a channel utilization of $\rho = 0.65$ can be determined from Appendix A.2.2, Figure A.7. The traffic capacity per cell (3 sectors) is then $\rho n \cdot 3 = 37$ Erl.

It is noticeable that sectorization in a given cluster size reduces spectral efficiency and maximum traffic capacity because of a decrease in trunking gain; the main advantage of sectorization is an improved C/I ratio!

2.4.3 Efficiency and Traffic Capacity with Sectorization and a Well-Chosen Cluster Size

Efficiency can be increased at the same time that sectorization occurs if the sectorized cluster is reduced in size but the C/I ratio is not allowed to drop below the value of, say 17 dB, achieved with omnidirectional antennas and 7-sector clusters. (The same calculations can be done for any other assumption for the C/I ratio.)

According to Equation (2.37), for the smallest allowable cluster size N (based on a minimum value $C/I = 17$ dB for the different sectorizations) there follows:

- Without sectorization, the cluster size must be $N = 7$, so that $C/I = 17$ dB.
- For $S = 3$ ($M = 2$), $N = 4$ is allowable, with $C/I = 18.5$ dB resulting.

- For $S = 6$ ($M = 1$), $N = 3$ is allowable, with $C/I = 19$ dB resulting.

The spectral efficiency described in Equation (2.36) produces the result shown in Figure 2.30:

- From a cell radius of $R = 3$ km (at 0.2 Erl./km^2 traffic volume), sectorized systems offer advantages over omnidirectional systems.
- From a 6 km radius, six sectors tend to be better than three sectors.
- With higher traffic volumes, 3-sector antennas have an advantage with cell radii from:

Vt [Erlang/km^2]	0.5	1
R [km]	2	1.7

The values for maximum traffic capacity clearly indicate the advantages of sectorized systems (see the first table in Figure 2.32). Despite the loss in trunking gain, sector systems prove to be superior to non-sectorized systems when a comparison is made on the basis of equal values for the C/I ratio.

2.4.4 Sectorization with Shadowing

The necessary C/I values can be maintained in spite of radio shadowing if there is a large overlapping of cells and larger clusters with, e.g., $N = 12$, i.e., the interference reduction factor $q = 5.6$. The cluster size with sectorization shown at the bottom of Figure 2.32 is necessary in order to achieve a C/I of 17 dB with an interference probability of 0.1 for $\sigma = 6$ dB (σ is the standard deviation of the lognormal distribution).

The percentage of efficiency increase with sectorization for these systems can be displayed in a similar way as in Figure 2.30 [22]. This shows that at 0.2 Erl./km^2, even with a 13 dB shadowing reserve, sectorized systems are better than omnidirectional ones starting from a radius of approximately $R = 3$ km. At 0.5 or 1 Erl./km^2, even sector systems with a 2 or 1.7 km cell radius are better than omnidirectional systems. The traffic capacity per cell follows from Figure 2.32 (table on right), calculated on the same basis as before.

2.5 The ISO/OSI Reference Model

Since the exchange of information between communicating partners is complex in structure and difficult to understand, the entire communications process has been universally standardized and organized into individual well-defined hierarchical *layers*.

Each layer, with the exception of the top one, offers *services* to the layer directly above it. The way these services are implemented is through the

Maximum cell radius at 0.2 Erl./km^2 and various sectorization

7-sector Cluster: with $p_I = 0.02$ (without shadowing)

Type of system	Channels per sector	Channels per cell	Traffic capacity per cell [Erlang]	Max. cell radius [km]	C/I [dB]
Omni $N=7$	59	59	48.7	9.7	17
3-sector $N=4$	34	104	76.6	12.1	18.5
6-sector $N=3$	23	138	94.6	13.5	19

(Traffic capacity per cell: increasing ↓)

27-sector Cluster: with $p_I = 0.02$ (with a 13 dB shadowing reserve)

Type of system	Channels per sector	Channels per cell	Traffic capacity per cell [Erlang]	Max. cell radius [km]
Omni $N=27$	-	15	9.0 ↓	4.2
3-sector $N=16$	8	26	10.9	4.5
6-sector $N=12$	5	34	9.9 ↑	4.4

Necessary cluster size with a 13 dB shadowing reserve

Type of system	Interfering channels M	Reuse distance ratio q	Cluster size N	C/I [dB]
Omni	6	8.7	27	30.4
3-sector	2	6.8	16	30.6
6-sector	1	5.69	12	31.1

Figure 2.32: Traffic capacity with shadowing reserve

passing of information between the *peer entities* of the respective layer of the communicating systems by means of *protocols*. In this process a layer uses the services of the next layer below it. Therefore within a process each entity communicates directly only with the entity immediately above it or below it. The higher-ranking layer is referred to as the service user and the lower-ranking layer as the service provider.

The hierarchical model facilitates communication between developers, suppliers and users of communications systems. If a change is undertaken in one of the layers, it does not affect the others. Furthermore, the structure of the layers makes it easier for protocols to be implemented and standardized.

Taking these aspects into account, the International Standardization Organization (ISO) specified a generally accepted layered model, the *ISO/OSI Reference Model*, for *Open Systems Interconnection*, OSI, a description that refers to almost all the communications systems in use today. This model is called OSI, because it describes the connection of open digital systems compliant with the respective ISO standards.

The OSI model is based on different principles. Each layer carries out a precisely defined function, and each function has been stipulated in line with internationally standardized protocols. The boundary lines between the individual layers have been established to minimize the information flow over the interfaces. Each higher layer represents a new level of abstraction from the layers below it. To keep the number of layers and interfaces to a minimum, several different functions have also been added to the same layer. A seven-layer model (see Figure 2.33) was created as a result of these considerations.

The following is a brief description of the different tasks of the seven layers of the OSI Reference Model [13]:

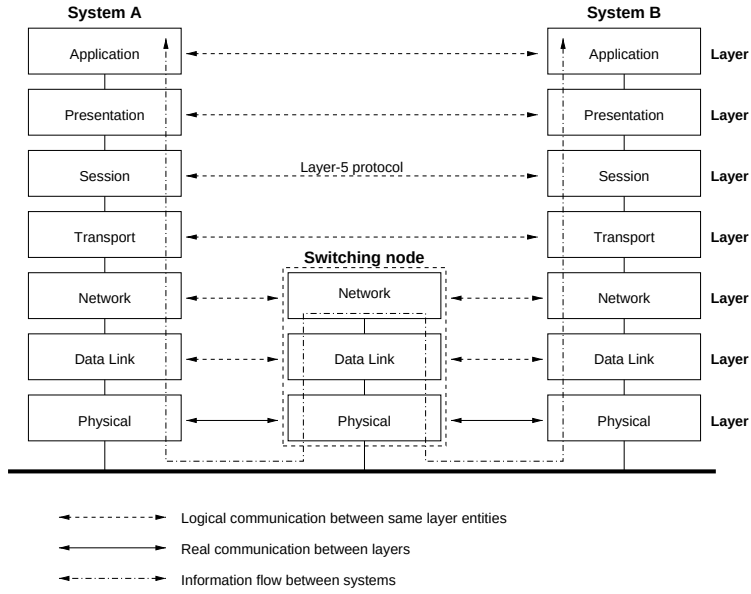


Figure 2.33: The ISO/OSI Reference Model

Physical layer: Layer 1 The bit transmission layer (*physical layer*) provides the basis for communication and facilitates the transmission of bits over a communications medium. Layer 1 describes the electrical and mechanical characteristics, e.g., standardized plugs, synchronized transmission over cable or radio channels, synchronizing techniques, signal coding, and signal levels for the interface between terminal equipment and line termination.

Data link layer: Layer 2 The task of the *data link layer* is to interpret the bit stream of layer 1 as a sequence of data blocks and to forward them error-free to the network layer. Error-detection or correction codes are used to protect data from transmission errors. Thus, for example, systematic redundancy that is used at the receiving side for error detection is added by the transmitter to the data, which is transmitted in blocks (frames).

These frames are transmitted sequentially between peer entities of layer 2. If a transmission error is detected then an acknowledgement mechanism initiates a retransmission of the block and guarantees that the sequence will be maintained.

The data link layer adds special bit patterns to the start and to the end of blocks to ensure they are recognized. Because of flow control on both sides, the logical channel can be used individually by the communicating partner entities.

Layer 2 contains the access protocol for the medium and the functions for call set-up and termination with regard to the operated link.

Network layer: Layer 3 The *network layer* is responsible for the setting up, operation and termination of network connections between open systems. In particular, this includes *routing* and address interpretation, and optimal path selection when a connection is established or during a connection.

Layer 3 also has the task of multiplexing connections onto the channels of the individual subnets between the network nodes.

Transport layer: Layer 4 The *transport layer* has responsibility for end-to-end data transport. It controls the beginning and the end of a data communication, carries out the segmentation and reassembly of messages, and controls data flow. Error handling and data security, coordination between logical and physical equipment addresses and optimization of information transport paths also fall within the range of this layer's tasks.

The transport layer represents the connecting link between the network-dependent layers 1–3 and the totally network-independent overlaid layers 5–7, and provides the higher layers with a network-independent interface. The transport layer provides a service with a given quality to the communicating applications processes, regardless of the type of network used.

Session layer: Layer 5 The *session layer* controls communication between participating terminals, and contains functions for exchange of terminal identification, establishing the form of data exchange, dialogue management, tariff accounting and notification, resetting to an initialized logical checkpoint after dialogue errors have occurred, and dialogue synchronization.

Presentation layer: Layer 6 The *presentation layer* offers services to the application layer that transform data structures into a standard format for transmission agreed upon and recognized by all partners.

It also provides services such as data compression as well as encryption to increase the confidentiality and authenticity of data.

Application layer: Layer 7 The *application layer* forms the interface to the user or an applications process needing communications support. It contains standard services for supporting data transmission between user processes (e.g. file transfer), providing distributed database access, allowing a process to be run on different computers, and controlling and managing distributed systems.

2.6 Allocation of Radio Channels

Utilization of the capacity of a transmission medium can be improved through different methods that involve transmitting several connections simultaneously in multiplex mode. Multiplexing is a technique permitting multiple use of the transmission capacity of a medium. The following techniques are used in radio systems:

- *Frequency-division multiplexing* (FDM)
- *Time-division multiplexing* (TDM)
- *Code-division multiplexing* (CDM)
- *Space-division multiplexing* (SDM)

In addition to these multiplexing methods, which facilitate the multiple use of the capacity of a transmission medium by many communications channels, there are access techniques to the respective frequency, time, code and space channels, which are abbreviated as follows:

- *Frequency-division multiple access* (FDMA)
- *Time-division multiple access* (TDMA)
- *Code-division multiple access* (CDMA)
- *Space-division multiple access* (SDMA)

As layer 2 protocols based on the ISO/OSI Reference Model, these access techniques are only indicators of the respective class of protocols, and are specified in each individual case (for each system).

2.6.1 Frequency-Division Multiplexing, FDM

With frequency-division multiplexing, the spectrum available to the radio system is divided into several frequency bands that can be used simultaneously (see Figure 2.34).

Each frequency band is regarded as a physical channel that is allocated to two or more stations for communication. Each station is able to send or receive using the entire available transmission rate of the frequency band.

The frequency spectrum is divided into frequency bands through the modulation of different carrier frequencies when messages are being transmitted. On the receiving end the signals are separated through appropriate filtering. Because real filters have a finite slope characteristic, *guard bands* are necessary to prevent interference (cross-talk).

Therefore full utilization of the available frequency band is not possible. Many public land mobile radio systems, such as the GSM network and trunked radio systems, use frequency-division multiplexing techniques.

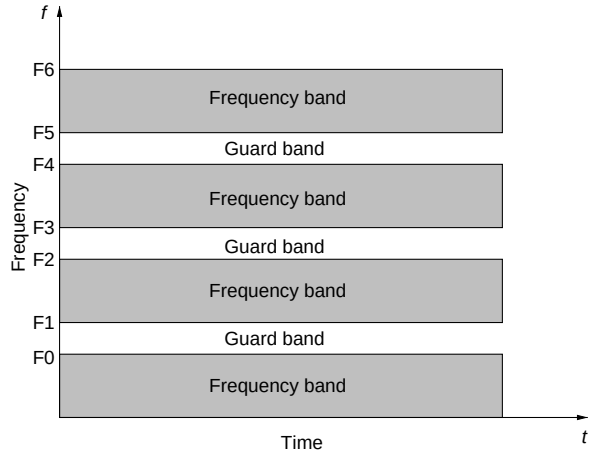


Figure 2.34: Frequency-division multiplexing technique, FDM

2.6.2 Time-Division Multiplexing, TDM

An FDM channel can sometimes offer more capacity than is required for a communications link. Periodically the frequency channel can then be allocated alternately to different communications links. This is the concept behind time-division multiplexing, in which the entire bandwidth of a radio channel is used but is divided into *time slots* that are periodically allocated to each station for the duration of a call (see Figure 2.35). The transmitting station can accommodate a certain number of data bits in a slot. The sequence of slots used by a station forms a time channel.

In some applications this regular allocation of time slots to stations, which results in constant engagement of the transmission medium, can be a disadvantage, especially when long transmission pauses occur. When this happens, the time slots are allocated centrally or decentrally to individual users as required. Instead of synchronous, asynchronous time slots are then allocated.

Access to a transmission medium using the TDM technique requires a multiplexer and, on the receiver's side, a demultiplexer, which must be capable of working together in perfect synchronization to enable transmitted messages to be allocated to the right time channels. Similarly to FDM, TDM systems must also have a *guard time*, in this case between the individual slots, to prevent synchronization errors and intersymbol interference resulting from signal propagation time differences. This guard time prevents the use of very short time slots and therefore reduces the utilization of the theoretically available capacity. Further, a delay is introduced to collect the data to be transmitted in a time slot. The radio modems must transmit in a burst mode now, since the frequency channel is available only during time slots with intermittent silence periods, and time compression of the message transmitted is necessary.

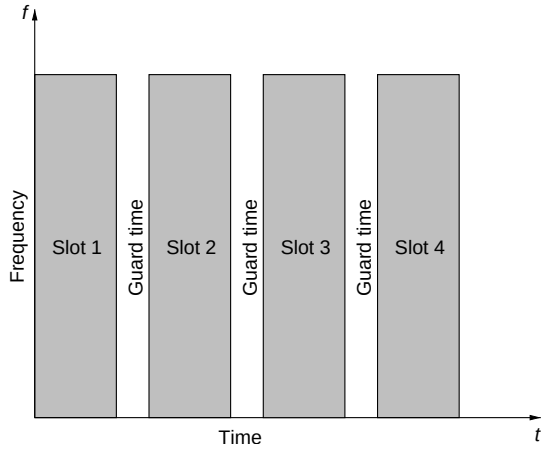


Figure 2.35: Time-division multiplexing, TDM

Although the TDM method is more frequency-economic than FDM, it requires very precise synchronization between the communicating parties, and therefore is technically more complex than FDM. Most of the digitally transmitting mobile radio systems also use TDM techniques in addition to FDM.

2.6.3 Code-Division Multiplexing, CDM

Code-division multiplexing separates the transmission channels neither into frequency areas nor into time slots. What is characteristic about this method is that it transmits narrowband signals in a wide frequency spectrum in which the narrowband signal is spread to a broadband signal through the use of a unique user code. This is called code spreading (see Figure 2.36). By this method a communications channel is defined by an unique code.

Each user in the radio communications systems is allocated a unique user code, which is used to spread the signal spectrum to be transmitted into multiples of the original bandwidth. These signals are then transmitted by the transmitters of the communicating terminals at the same time in the same frequency band, typically to/from the same base station. The user codes used by the transmitters must be selected in such a way that the receivers experience minimal interference despite the fact that transmission is taking place simultaneously. The use of an orthogonal *pseudo-noise* (PN) code for carrier modulation of the information being transmitted fulfils this condition.

The receiver, which must know the transmitter's user code, searches the broadband signal for the bit pattern of the transmitter's PN sequence. Through an autocorrelation function (ACF), the receiver is able to synchronize with the transmitter's code channel and de-spread the signal back to its original bandwidth. The signals of the other transmitters, the codes of which

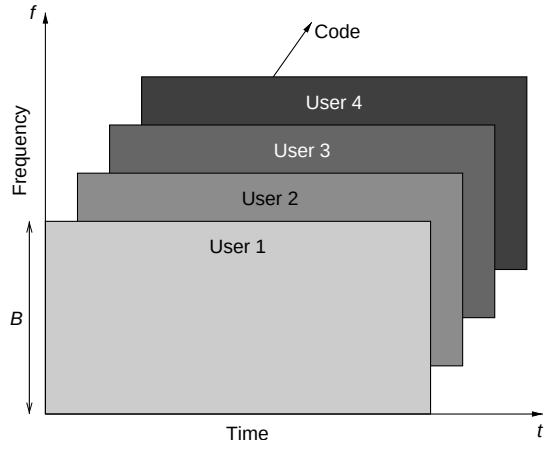


Figure 2.36: Code-division multiplexing, CDM

are different from the PN sequence selected, are not de-spread by this procedure, and therefore only contribute to the noise level of the received signal.

With a certain number of code channels on the same frequency channel the (*signal-to-noise ratio*, SNR) can fall below the value required for reception by the correlator. The CDM method thereby restricts the number of users who can use the same frequency channel.

In practice mainly two different methods are used for the spread spectrum transmission of a signal:

- *Direct sequencing* (DS)
- *Frequency hopping* (FH)

An advantage of CDM is that because of the coding the user data remains confidential, thus cancelling out the need for a cryptographic method for the protection of transmitted data. Systems using CDM (e.g., IS-95 and UMTS) have better protection against interference than pure FDM or TDM systems: this applies to atmospheric as well as to deliberate interference to the system. A jamming transmitter station usually does not have enough transmission power in order to cover an entire frequency spectrum nor the necessary information in order to pick up a particular call to jam. Another advantage over the TDM method is that the different transmitters in a CDM system do not need to synchronize their time. Because of the codes, they are self-synchronizing.

A system-inherent disadvantage of CDM is that transmitters and receivers must generate pseudo-random code sequences. When several stations are transmitting simultaneously (this is the normal way of operation), random statistical superimposition can occur, causing errors and creating the need for error-detection and correction measures. Furthermore, each receiver must

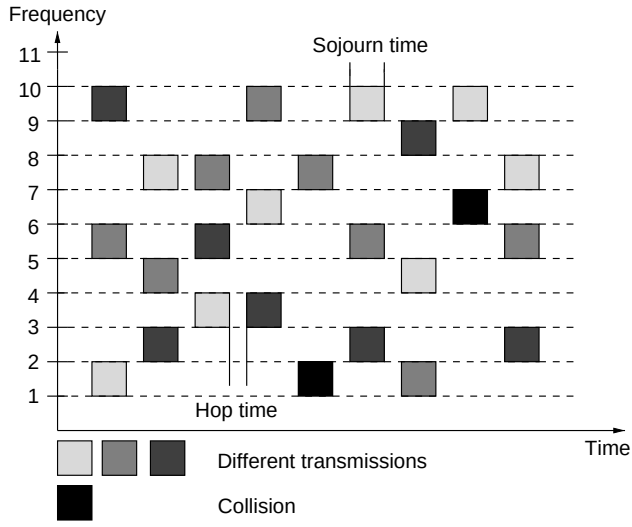


Figure 2.37: Frequency hopping spread spectrum technique

utilize fast power control to prevent transmitters with strong signals from causing interference to the signals of weaker transmitters.

2.6.3.1 Direct Sequencing

Direct sequencing is a spreading technique in which the binary signals to be transmitted are added modulo two to the binary output signal of a pseudo-noise generator and then used to phase-modulate the carrier signals. A combination of the data bits with the pseudo-random bit sequence (*chip sequence*) converts the narrowband information signal to the large bandwidth of the PN signal, thereby producing a code channel [30]. The PN signal can use a Barker, Walsh or Gold code.

2.6.3.2 Frequency Hopping

In *frequency hopping* transmitters and receivers change transmission frequency synchronously and in quick succession (see Figure 2.37). The already-modulated information signal is added modulo two to the signal of a code generator, which controls a frequency synthesizer. This causes a considerable widening of the original bandwidth. Frequency hopping takes place either quickly (many *hops* per information bit) or slowly (one *hop* for a large number of information bits).

It is possible for several transmissions to take place at the same time in the available frequency range, but this can lead to collisions if two or more transmitters happen to be using the same frequency simultaneously. These

collisions, which can reduce the carrier to interference ratio to an unacceptable low level, are preventable through the use of orthogonal hopping codes.

The advantage of this method is that interference in the transmission channel, e.g., caused by co-channel interference, will usually only affect a small part of a message. The defective part of the communication can, if necessary, be compensated for on the receiving side through error-protection methods.

2.6.4 CDMA Technique for 2nd-Generation PLMNs

Since the early 1980s, CDMA has been used in experiments involving wireless communication in buildings and outdoors as well as over satellites [3, 30].

This technology has generated special interest since early 1991 because of an announcement by QUALCOMM USA in San Diego about a cellular CDMA system claiming to offer a particularly high level of spectral efficiency in cellular radio technology. Telecommunications Industry Association (TIA) Interim Standard 95 (IS-95) has been produced since then (see Section 4.3).

Its characteristics are:

- A frequency band B for all code channels.
- A large number of simultaneous digital signals in the same band B.
- Use of (almost) orthogonal signal sequences.
- Each signal sequence in a channel can be detected and decoded in the receiver if the code sequence is known.
- Interference limitation: The number of simultaneous transmitters is restricted owing to reciprocal interference at the receivers.

Orthogonality can be achieved:

with	through
TDM	Signal separation in the time domain
FDM	Signal transmission in different frequency bands
CDM	Channel-specific code for signal sequences

The bandwidth B of a user's CDMA signal is considerably wider than his channel bit rate R. The larger B/R is, the greater the number of simultaneous code channels in the same band B. The performance per bandwidth unit is reduced through spreading.

Example: If a 1 MHz signal is spread 100 times to $B = 100$ MHz and is transmitted with 1 W power then

- a receiver with a bandwidth of 1 MHz only receives 10 mW of interference power (attenuated because of the distance).
- a transmitter with a 1 MHz bandwidth causes interference to a receiver with a 100 MHz bandwidth with only 1/100 of its power.

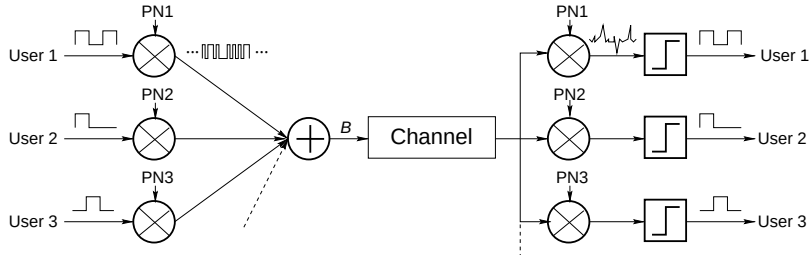


Figure 2.38: Simplified version of a CDMA communications system

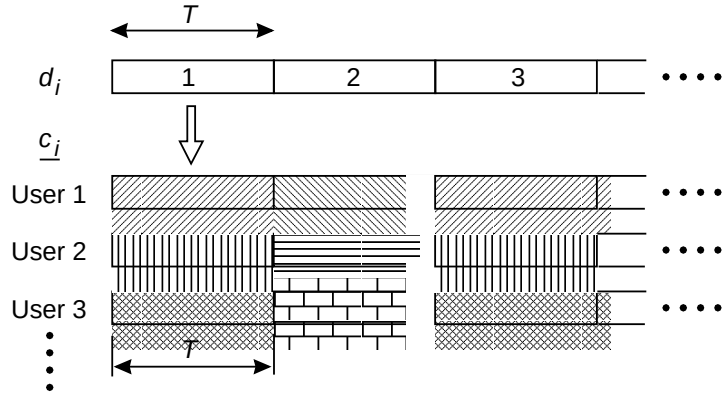


Figure 2.39: Orthogonal spreading of symbols

Figure 2.38 shows a diagram of a CDMA communications system (without a modulator or a demodulator):

- User signal sequences are multiplied by individual spreading sequences. PN_1, PN_2, ... and transmitted over the same band.
- The signal sequences with errors are multiplied by the channel-specific codes in the (correlation) receiver and restored by an equalizer
- The overlaid signals of multiple users resembles Gaussian noise.

CDMA signals are not sensitive to narrowband interference. The method claims to offer spectral efficiency several times greater than what is possible with FDMA/TDMA.

Problem with CDMA: All signal sequences received by the receiver must be of the same strength within approximately 1 dB, otherwise stronger signals will suppress weaker signals (*near/far problem*). An adaptive (fast) *power control* for the transmitter is required by the receiver.

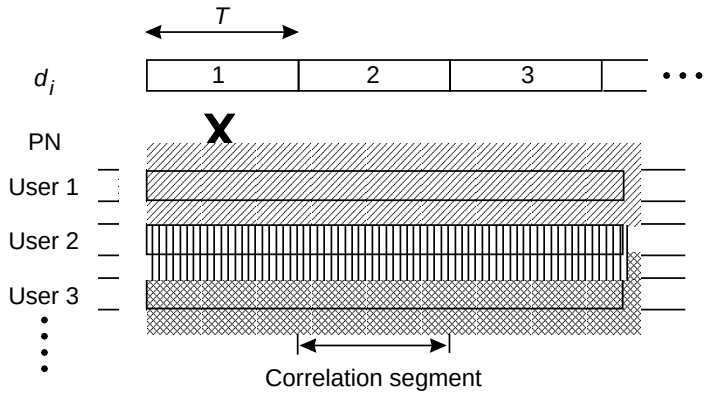


Figure 2.40: Code spreading based on the DS method

2.6.4.1 Different Popular CDMA Techniques

Orthogonal spreading of symbols The data symbol d_i with the duration T of each user is individually mapped onto a set of orthogonal code sequences each of length T [3]:

$$\underline{c_k} = \{c_{k1}, c_{k2}, \ldots, c_{kL}\}, \tag{2.38}$$

The duration of each chip c_{kj} in the sequence is T/L, with L indicating the length of the code sequence (see Figure 2.39).

If data symbols for the basis m (m-ary) are used and purely orthogonal coding of the symbols is applied then an individual quantity of m orthogonal code sequences $\underline{c_k}$ is required for each user (channel).

The main advantages of symbol-by-symbol code spreading are as follows. The orthogonality and cross-correlation (i.e., channel interference) of the code sequences depend strongly on the code selected and therefore are controllable. If transmission and the reception of symbols of different channels occur simultaneously, e.g., within a cell, then capacity (channels/bandwidth) is increased substantially (i.e., the orthogonality is fully effective).

Initially, there was no satisfactory solution for establishing extensive code sequences with good cross-correlation properties on the basis of limited code lengths L, as required for cellular networks. Recently it has been shown that there exist large code families with the codes being nearly perfectly orthogonal to each other [31].

Spreading based on the direct sequencing (DS) method This involves the DS multiplication of each user's data signals with the user-specific PN binary chip sequence having a period clearly longer than that of the data symbol T (see Figure 2.40). The chip length is then $T/L \ll T$.

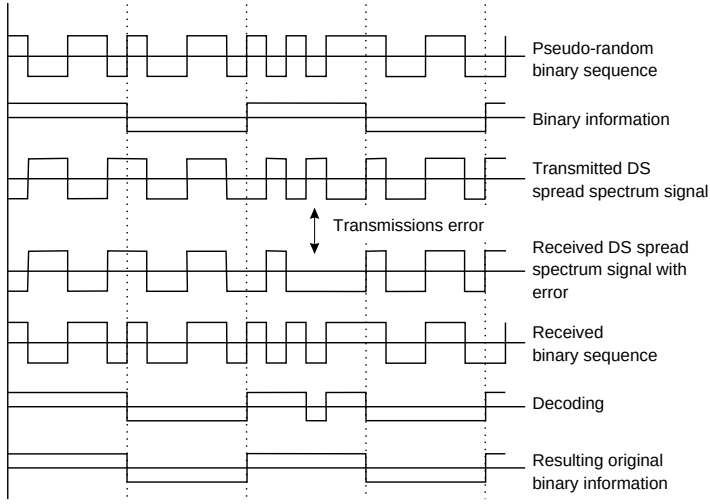


Figure 2.41: Example of DS spreading method

Figure 2.41 shows the effect of code spreading in the baseband. The interference caused by spread signals (*chips*) during transmission can be detected and suppressed after the despreading.

Advantage: The selection of code sequences and their allocation to users is greatly simplified (e.g., IS-95 uplink: a common PN sequence of length $2^{42}-1$ for all users in all cells, with a user-specific starting sequence taken from this PN sequence; see Section 4.3.1).

Disadvantage: The selection of code sequence has little influence on the cross-correlation of the different code channels. *Reason*: Random cross-correlation of spread code segments of symbol length T, which only amount to a small part of the whole PN sequence.

Hybrid methods A PN sequence is symbolically multiplied by a Walsh-function (with period T) that is user-specific. A PN sequence plus a complete set of orthogonal Walsh functions produce a finite set of periodic DS spreading functions with guaranteed orthogonality with regard to cross-correlation during the symbol length T (see Figure 2.42). This method is applied by IS-95 on the downlink for each cell, see Section 4.3.2.

Frequency hopping techniques This is a code spreading system in which transmission is spread over different frequency bands, which are used sequentially: *fast frequency hopping* with many hops per symbol (bit).

Advantage: The system is robust with respect to different receiving levels—better than DS-CDMA.

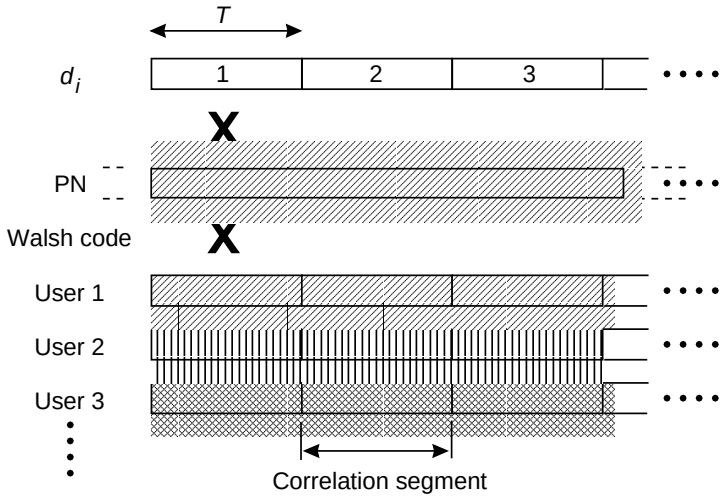


Figure 2.42: Combined DS and orthogonal symbolic spreading

Disadvantage: The code allocation is complicated when there are a large number of users per cell. The technical complexity of frequency changing of up to 1 Mhop/s is required, and the complexity of the FH receiver should also be mentioned.

Example: In GSM (see Section 3.3.2), *slow frequency hopping* is used to avoid the consequences of signal fading on a small number of frequencies of the hopping sequence. The hopping frequency has 1 hop/burst (burst$\hat{=}$157 bit $\hat{=}$4.6 ms) $\rightarrow$ 217 hop/s.

CDMA combined with FDMA and TDMA The effectiveness of this method is underlined by its use with the UMTS; see Section 5.6.1.

The assumption that the spectral efficiency (Erlang per km^2 and MHz bandwidth) is as high with TDMA as it is with CDMA has been widely discussed in the literature, with apples being compared with pears; there does not seem to be a fairer mathematical comparison.

2.6.4.2 CDMA in Cellular Systems

- All frequency bands in each cell can be used.
- Because of the resultant interference ratio C/I, two channels (one each per BS) are usually operated simultaneously in the neighbouring area of two cells.
- Power control must be performed quickly: a 1 kbit/s channel is required with IS-95; each `1` bit transmitted means 1 more level of power, each `0` bit means 1 power level less. There are many levels.

- Speech transmission with variable bit rate: 1 kbit/s up to approximately 6 kbit/s, depending on the speech activity.
- As the traffic load increases, the service quality decreases because of system limitations, e.g., *graceful degradation* of speech quality.

2.6.4.3 TDMA in Cellular Systems (GSM)

Per GSM network:

$$\begin{aligned} 61 \text{ channels} \cdot 200 \text{ kHz} &= 12.2 \text{ MHz} \\ N = 7 &\rightarrow 9 \text{ frequency channels/cell} \\ (N = 12 &\rightarrow 5 \text{ frequency channels/cell}) \end{aligned}$$

$$\begin{aligned} \text{With CDMA:} & \\ N = 1 \text{ with } B = 12.2 \text{ MHz} &\rightarrow 7\text{-fold spreading possible} \end{aligned}$$

On the basis of past experience with CDMA systems, a spreading factor of 7 is clearly too low. A factor of 100 to 200 is required instead, B-CDMA (B$\hat{=}$broadband). Therefore a different channel structure is needed, e.g., not 200 kHz per channel but 10 kHz instead, which will produce a spreading factor of 140 (corresponding to the IS-95 approach, see Section 4.3).

2.6.5 Space-Division Multiplexing, SDM

With space-division multiplexing the frequencies used for transmission are reused at appropriate geometric intervals so that theoretically a huge volume of traffic can be transported over an enormous extended area despite the limited frequency spectrum.

This method is facilitated by the circumstance that the field strength of a radio signal decreases as the distance from the transmitter increases. Therefore the space-division multiplexing method uses the propagation attenuation of electromagnetic waves. If the distance from the transmitter is great enough, the signal becomes so weak that the noise due to interference that occurs when this frequency is reused by another transmitter can be tolerated (cluster principle).

Space-division multiplexing describes the basic concept of cluster formation that helps to provide cellular systems with an almost unlimited amount of traffic capacity restricted only by cost considerations. Sectorization of the supply area of a base station is another related measure.

There has been recent discussion about using phased-array antennas to set up multiple antenna beams that swivel electronically and can focus sharply, thereby allowing a base station's various *spot beams*, which have been angled towards different directions, to use the same frequency channel at the same time. Space-division multiplexing would be occurring in the same cell, thereby

contributing to a dramatic increase in capacity. Spot beam antennas reduce the angle of incidence at the receiver for signal phasing, thereby reducing signal interference, i.e., dispersion, and allowing the use of simple (low-power) equalizers.

2.6.6 Hybrid Methods

In practice, hybrid multiplexing methods are usually applied because they offer efficient means of utilizing the frequency spectrum. Therefore a combination of two or several multiplexing techniques are applied in one system. By using a combination of technologies and incorporating the respective advantages of the different multiplexing techniques, it is possible to make good economic use of the radio channels (see Figure 2.43).

In the public mobile radio networks GSM, USDC and PDC, for example, frequency-division multiplexing as well as time-division multiplexing are used alongside the respective cellular structure. The systems DECT, TETRA, HIPERLAN and UMTS are other examples of the use of a combination of the two methods, FDM and TDM. All these systems are introduced later and described in detail.

2.7 Fundamentals of Error Protection

2.7.1 Error Protection in Radio Channels

Channel coding protects transmitted data from transmission errors. Because of fluctuations in signal level *(fading)* that characteristically occur in mobile radio transmission, the bit-error ratio can fluctuate considerably owing to distance, speed and the radio signal shadowing of stations communicating with each other.

2.7.1.1 Effects of Fading

A distinction is made between *long-term fading* and *short-term fading* of a received signal. Long-term fading is caused by shadowing, such as by buildings; therefore it has a greater chance of occurring in built-up urban areas than in flat countryside. In GSM networks this type of fading occurs at intervals of about 12–60 m; at a speed of 36 km/h this produces intervals in the range of 1.2–6 s.

Short-term fading is mainly caused by the multipath propagation of radio waves that are reflected or diffracted on obstacles such as buildings and hills (see Figure 2.8). The transmitted signal reaches the receiver with time delays and on paths with different lengths. Depending on the phase position of the signals of the individual paths, this results in interference (loss or boosts) of the received signal at the receiver. Fades are more or less at fixed locations in a given environment. The distances of the fading minima are frequency-

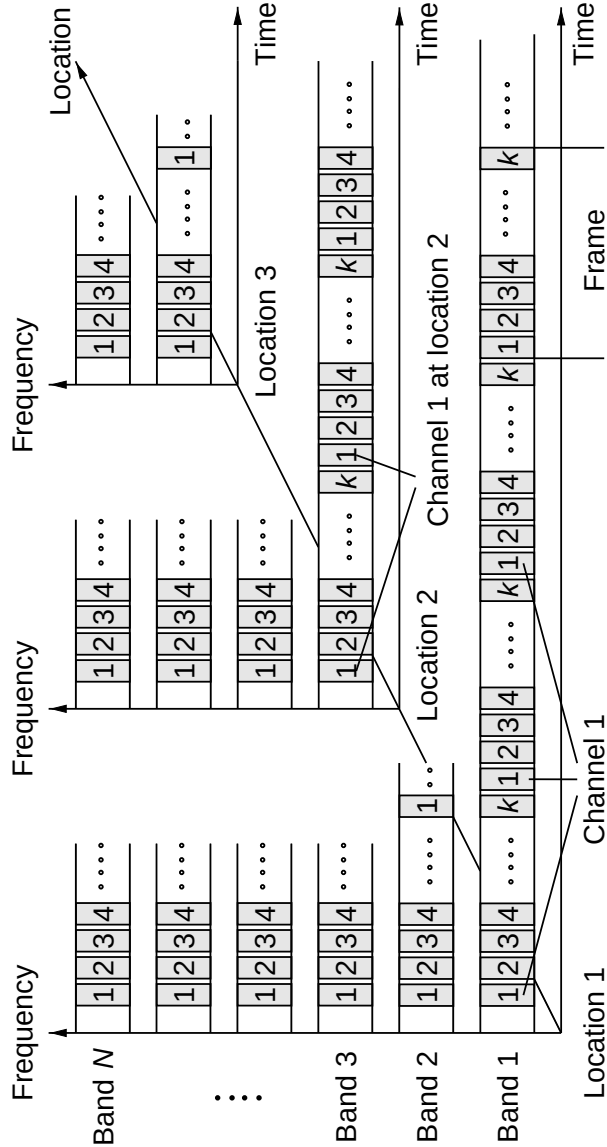


Figure 2.43: Using a hybrid multiplexing technique with a combination of FDM, TDM and SDM

dependent and about half a wavelength; thus in GSM at approximately 15 cm (in DCS 1800 at approximately 8 cm), which explains the term short-term fading seen from a mobile receiver.

2.7.1.2 Necessity of Error-Protection Methods

Error-protection methods reduce or eliminate the effects of interference to the receiving signal described above, and hence the bit-error ratio of the received data stream. Special codes are selected, which are used in conjunction with *interleaving* and *frequency hopping*. Short-term fading is much easier to combat than long-term fading.

Error-protection methods for data transmission have much higher requirements than those for speech. Whereas with speech a residual bit-error ratio of 10^{-2} is still considered tolerable, with data it should be lower than 10^{-7} if possible. This explains why different error-protection methods are used for speech and for data transmission in digital mobile radio networks.

A distinction is made between three different methods of error protection, namely error *detection*, error *correction* and error *handling*.

2.7.2 Error Detection

Through error detection it is possible to establish whether a received data word is a valid code word. However, data words that are recognized as being incorrect cannot be corrected.

In GSM checksums of cyclic codes (CRC $\hat{=}$ *Cyclic Redundancy Check*) are used for error detection. This usually involves BCH codes (*Bose-Chaudhuri-Hocquenhem-Codes*), which are described in detail in [32].

An (n, k)-BCH code is characterized by the following values:

k			block length *before* coding (data word length)
n	$=$	$2^m - 1$	block length *after* coding (code word length)
$m \cdot t$	$=$	$n - k$	number of redundant parity bits
d_{min}	$\geq$	$2t + 1$	Hamming (minimum code words) distance

In RLP, the data link control protocol for GSM (see Section 3.9.4), a (240,216) BCH code is used, i.e., a 24-bit long frame check sequence is attached to the original data word, which consists of $k = 216$ bits. The data word that is created after the coding has a length of $n = 240$ bits.

This code has a Hamming distance of $d_{min} = 7$, so that all defective code words with a maximum $d_{min} - 1 = 6$ errors (at random bit positions) are detected. More than 6 errors in a code word can result in the creation of another valid code word, which will be accepted as correct. If this happens, the code has failed in this particular case.

The residual bit-error ratio (RBER) P_e of an (n, k)-BCH code is

$$P_e = \sum_{i=d_{min}}^{n} r(i)p(i) \tag{2.39}$$

$$r(i) = \frac{A(i)}{\binom{i}{n}} \tag{2.40}$$

$$p(i) = \sum_{j=0}^{i} \binom{j}{n} BER^j (1 - BER)^{n-j} \tag{2.41}$$

The values represent:

- $r(i)$ probability (reduction function) with which an error pattern with the Hamming weight i corresponds to a valid code word $A(i)$ and therefore is not recognized as incorrect
- $A(i)$ weight distribution of a code
- $p(i)$ probability of the occurrence of an error based on weight i

The RBER for the range $d_{min} \leq i \leq n - d_{min}$ can then be estimated using the following formula:

$$P_e = \frac{\text{number of used code words} - 1}{\text{number of all possible code words}} = \frac{2^k - 1}{2^n - 1} \approx 2^{-(n-k)} \tag{2.42}$$

Accordingly, the residual bit-error ratio depends on the number of parity bits. Therefore it is irrelevant whether a data word with 50 bits or one with 3000 bits is protected by a 24-bit long CRC sequence; the residual error probability is equally high in both cases.

For the (240,216)-BCH code used in the RLP mentioned above the RBER approximates

$$P_e = 2^{-(240-216)} = 2^{-24} \approx 5.96 \cdot 10^{-8} \tag{2.43}$$

An RBER of this magnitude is sufficient for most applications in mobile data transmission.

2.7.3 Error Correction

Forward error correction (FEC) also occurs in the context of error-protection coding. With forward error correction the transmitter adds so much redundancy to a data word that the receiver is able to correct a certain number of errors. In contrast to the ARQ methods described in Section 2.7.4, a reverse channel from the receiver to the transmitter is not required.

A distinction is made between two code families that are suitable for forward errror correction, namely the linear block codes and the convolutional codes. Linear block codes are systematic codes, i.e., a certain number of redundant bits are calculated from the data word to be protected and are transmitted with it. The coded data word can thus be divided into a redundant and a non-redundant part. On the other hand, the convolutional codes described later are non-systematic codes.

2.7.3.1 Reed–Solomon Codes

Reed–Solomon (RS) codes constitute a special class of BCH codes that use q-ary instead of binary symbols. Because of the binary character of the data

Table 2.2: Block lengths of Reed–Solomon codes over GF(2^m)

GF(2^m)	Block length n in	
m	q-ary symbols	bit
3	7	21
4	15	60
5	31	155
6	63	378
7	127	889
8	255	2040

being transmitted, RS codes are usually selected over the Galois field GF(2^m), thus at $q = 2^m$.

According to [22], an RS code can be described based on the following values:

k					block length before coding
n	$=$	$2^m - 1$	with	n	block length *after* coding
$d_{min} - 1$	$=$	$n - k$		d_{min}	Hamming distance
$2t + e$	$=$	$n - k$		t	number of correctable errors
				e	number of losses

RS codes are particularly suitable for the correction of errors that occur in bursts (bursty errors). This makes them especially useful in mechanical storage systems (e.g., hard disks) as well as with compact disks (CD), because these frequently have large blocks that are garbled (e.g., due to scratches on the surface). Because strongly correlated errors likewise occur in radio transmission owing to signal fading, RS codes are also highly suitable for this area.

A particular feature of the RS code in comparison with the BCH code is that with given block length n the data word length k can be selected, and this selection always produces an optimal distance d_{min}.

Table 2.2 presents calculations for the block lengths of some RS codes. It can be seen that the block length with the factor m of the Galois field increases quickly, which results in very redundant codes. Abbreviated RS codes are used to counteract this problem (see [32]). This involves stuffing a given data block length with so many zero symbols that a new length is produced with the desired redundancy and consequently the required correction quality. The zero sequence itself is only used to form the check bits, but is not transmitted because before decoding the receiver can add these zero symbols to the same positions again.

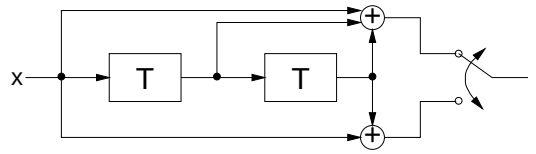


Figure 2.44: Structure of a (2,1,3) convolutional code

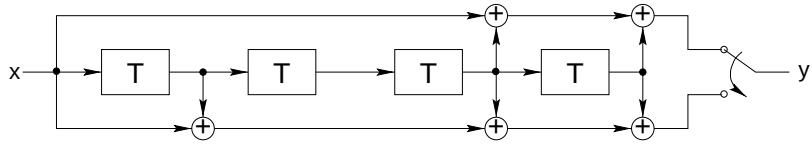


Figure 2.45: (2,1,5) convolutional coders in GSM

2.7.3.2 Convolutional codes

In comparison with block codes, convolutional codes have a memory, i.e., a code word position is not only dependent on the actual data bit but on several preceding data word positions. An (n, k, m) convolutional code has an influence length m, k input and n output bits per data word. The direct result is a code ratio of $R = k/n$. Convolutional codes are explained in detail in [22].

Figure 2.44 presents a schematic structure of a simple (2,1,3) convolutional code with code ratio equal to 1/2. This coder has two internal shift registers T and three exclusive-or functional units. The input bit x_i is coded into two output bits $y_i^{(1)}$ and $y_i^{(2)}$. Together these two output bits form the code word.

Because of the two shift registers, the convolutional code can assume $2^2 = 4$ different states. In the literature these states are labelled as a, b, c and d.

The convolutional coder changes its state depending on the bit occurring at the input (`0` or `1`). A number of $(m - 1)$ `0` bits, called tail bits, are added to the bit stream to ensure a return to the initial state a after the coding of any length of bit stream.

Viterbi algorithms and *Maximum Likelihood Decision* (MLD) are used in convolutional decoding. The path generating the bit sequence that comes closest to that of the received sequence is selected from the Trellis diagram [22, 32]. This path is then used for decoding the entire received bit sequence.

The residual bit-error ratio (RBER) remaining after convolutional decoding decreases in line with the number of internal shift registers of the coder. As an example, the RLP of GSM uses a (2,1,5) convolutional code. This convolutional coder referred to as GSM-96 has four registers (see Figure 2.45) and a lower RBER than the (2,1,3) convolutional coder shown in Figure 2.44.

The decoding complexity increases disproportionately to the number of memories for the convolutional coder. Convolutional coders with 2–10 shift

registers are common today in quasi-real-time operations with decoding in the millisecond range.

Convolutional decoders are very suitable for the correction of uncorrelated errors, but are extremely sensitive when it comes to bursty distributed errors such as those which frequently occur in radio channels. Consequently, in mobile radio systems convolutional coders are almost exclusively used in combination with *interleaving* (see Section 3.3.7.1), a technique that is able to let bursty errors appear as single-bit errors.

2.7.3.3 Soft Decision

With the decoding method called *soft decision* the convolutional decoder receives additional information from the *equalizer* about the bit just received. In contrast to the *hard decision* method, in addition to the actual value of the bit (`0` or `1`) a statement about the reliability of the decision designated as a soft decision value is provided based on the received signal level by the equalizer after quantization (e.g., coded with 2–8 bits in GSM).

The soft decision value is proportional to the local bit error ratio and can be used by the convolutional decoder for its maximum-likelihood decision. The quantized soft decision value can be assigned to the local bit error ratio using the following equations:

$$p_e = \frac{e^{-sd}}{1 + e^{-sd}}, \qquad sd = -\ln\left(\frac{p_e}{1 - p_e}\right) \tag{2.44}$$

A more accurate statement about the reliability of the decision on each bit decreases the residual bit-error ratio compared with the hard decision method. The gain amounts to about 3 dB, i.e., with the same residual bit error ratio a channel must show a 3 dB higher C/I value with a hard decision than with a soft decision.

2.7.3.4 Punctured Convolutional Codes

Puncturing is a method for shortening a convolutional code. This involves deleting one or more bits from the output stream of the convolutional coder according to a prescribed scheme called a puncture table.

A puncture table consists of the elements `0` and `1` and is periodically read-out. A `0` signifies that the corresponding bit is not being transmitted in the output bit stream; a `1` means that the bit is being sent. This shortens the coded sequence and weakens the convolutional code. So the puncturing removes part of the redundancy added with the convolution process.

The advantage of a punctured code is that different code ratios can be implemented. Based on a mother code with the ratio $1/n$, codes with higher code ratios can be developed through periodic puncturing.

Punctured codes can be decoded by the receiver using the same convolutional decoder as for the mother code. To do so, the receiver, also familiar

with the puncture table, adds bits with a random value (e.g., `0`) at the appropriate positions. This process is called repuncturing. To the Viterbi decoder this 'unknown bit' means that the probability of a `0` or a `1` is the same.

2.7.4 Error-Handling Methods by ARQ-Protocols

Unlike forward error correction, when *Automatic Repeat Request protocols* are employed, corrupted data words are not corrected by the receiver and then accepted, but are instead checked for correctness on the basis of an error-detection code, which may also be used for correcting them (see Section 2.7.2). If a data word is not recognized as being correct, the receiver requests the transmitter to transmit it again.

The prerequisite for this method is a reverse channel between transmitter and receiver over which the results of the error evaluation are transmitted to the transmitter as an *acknowledgement*. For this purpose all the transmitted data words are provided with a sequence number NS. A positive acknowledgement ACK[NS] is sent when the word received is correct. If it is defective then a negative acknowledgement NAK[NS] is sent along with a request that the data word should be resent. A data word has the structure of a protocol data unit of the ARQ-protocol used, and is mostly called a block, a frame or a packet.

The transmitter must maintain a copy of the packet sent until the acknowledgement indicates that it has been received error-free.

To ensure that a faulty transmission can be detected in the first place, the transmitter must transmit the checksum of an error-detecting code (CRC $\hat{=}$ *Cyclic Redundancy Check* or FCS $\hat{=}$ *Frame Check Sequence*) in the packet.

2.7.4.1 Frame Types and Flow Control

Flow control is implemented in order to regulate the data flow from the data source in such a way that the data sink is not overloaded. The mechanism is based on numbering in sequence all frames carrying data from higher layers. This numbering helps to monitor whether a frame is missing. Furthermore, it provides a means for confirming explicit receipt of frames and requesting that frames be resent when a layer 2 connection is in the acknowledged mode of transmission. Section 3.4.1 explains how this mode is reached. But first of all, here is an overview of the different types of frames that are usually sent between two peer entities in a data link layer in acknowledged mode of operation.

- **I-Frame, Information**
  This frame transports the information fields provided by layer 3. The frames are numbered in sequence (NS, *Send Sequence Number*), and *piggy-back* acknowledgement is possible through the inclusion of an NR (*Number Received*) number.

- **RR-Frame, Receive Ready**
  This indicates that a layer-2 entity is ready to receive I-frames. All frames that have already been received are acknowledged through the parameter NR contained in the RR frame.
- **RNR-Frame, Receive Not Ready**
  This frame occurs when the transmitter of this frame is unable to receive further frames. This frame also contains an NR.
- **REJ-Frame, Reject**
  This frame is used when a sequence error occurs while I-frames are being received. It indicates that the other side should repeat the I-frames. The repeat process begins with the I-frame in which the NS is equal to the parameter NR contained in the REJ-frame.
- **SREJ-Frame, Selective Reject**
  An SREJ is an alternative to the REJ-frame. The difference is that a single I-frame is requested on a selective basis. The side that receives an SREJ should continue with normal operations after this requested I-frame has been sent.

Sequential numbering All I-frames transmitted are provided with a continuous number NS modulo n. If an I-frame is lost then its number can be used to confirm this fact.

All the frame types listed above contain a Received Number (NR). The NR in a frame that has been received represents an acknowledgement of all I-frames that have already been sent with the sequence numbers including NR−1. NR is the sequence number of the next I-frame to be sent. Both sequence numbers, NS and NR, are normally coded in bit fields of length n.

Sliding window protocol Protocols of the data link layer must be able to deal with special cases in which frames have been damaged or lost. The sliding window technique helps to control the information flow in such a way that communication between entities is kept synchronous.

- **Transmit window**
  The transmitter maintains a list of successive sequence numbers (NS) that match the corresponding I-frames. These I-frames are in the transmit window. The lower edge of the window is marked by VA (*Acknowledge Variable*) and the upper edge by VS (*Send Variable*). The lower edge is updated through the NR frames. The upper edge is increased when a new I-frame is sent. I-frames remain in the window until receipt has been confirmed by the other side. If VA = VS applies then the window is open. If all the possible window places are occupied (VS = VA + k, Window k) then no further I-frames are allowed to be sent; the window is closed. I-frames must be sent by the transmitter in the sequence corresponding to the information supplied by layer 3.

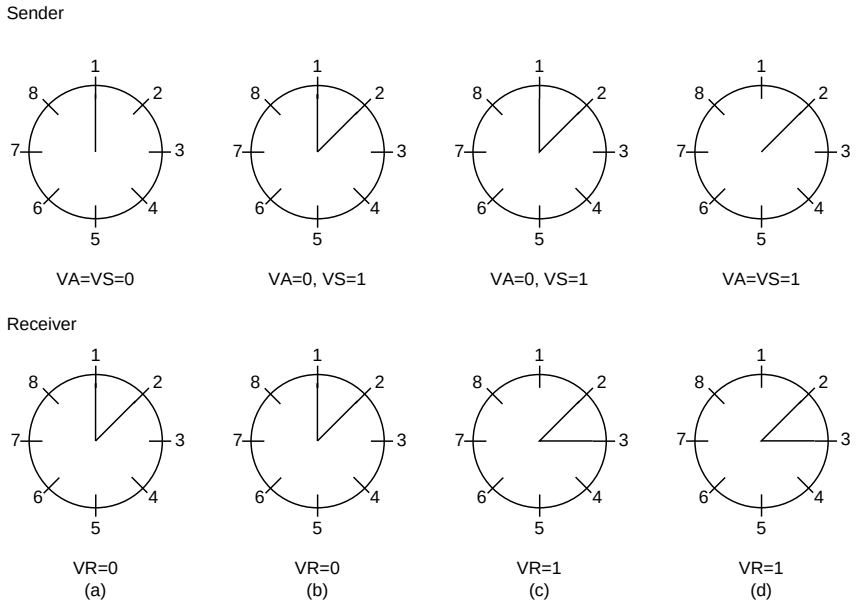


Figure 2.46: Size-1 sliding windows

- **Receive window**
  The receiver keeps a list of all the sequence numbers (NS) that correspond to the I-frames waiting to be received. VR (*Receive Variable*) indicates the lower edge of this window. The receive window has a fixed size k. I-frames can be accepted in this window in any sequence, as long as their sequence number (NS) fits in. It is important that the layer-2 entity passes on the information received from these I-frames to its layer 3 in the sequence in which it was sent.

Figure 2.46 presents an example of how the sliding window technique functions [34]. Each partner entity has a transmitting window as well as a receiving window. The transmitting and receiving windows can each be of a different size.

2.7.4.2 Time Monitoring

The confirmation of a transmitted frame that is part of a transmit window is monitored by a *timer*. If confirmation for the I-frame arrives on time, the timer is stopped. If no confirmation is received, the I-frame is resent after the timer runs out.

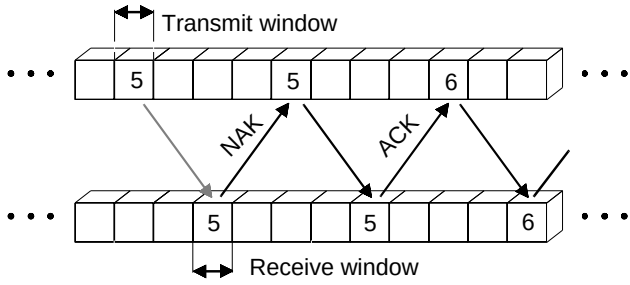


Figure 2.47: Send-and-wait ARQ

With a size-1 transmit window it is possible either to set up a separate timer for each window space or to pursue the principle of timer extension.

It is also possible to use the timer to influence the window size. On the assumption that a size-4 transmit window is planned, if the same premise applies that a timer runs out for an I-frame before the next possible transmitting cycle for the next I-frame then only one window space is used in the entire window. In effect the protocol runs its course with window size 1.

2.7.4.3 Send-and-Wait ARQ Protocol

Protocols with a window size of 1 are referred to as *send-and-wait* protocols. This means that a transmitter sends an I-frame and then is required to wait for (positive or negative) acknowledgement that this I-frame has been received before the next I-frame can be sent. The LAPD_m belongs to this group of data link control protocols (see Section 3.4.1).

The transmitter responds to the receipt of acknowledgement NAK by retransmitting the packet, and the subsequent packet is then sent after an ACK has been received. The throughput with this method is minimal, especially with long signal propagation delays and with short data packets.

The minimal waiting time for an acknowledgement corresponds to the length of t_{rd} (*round-trip propagation delay*). The period of time t_{rd} is equal to double the transfer time t_f of the transmission channel.

A sample protocol sequence for the send-and-wait ARQ method is found in Figure 2.47; the numbers refer to specific packets.

With

$$\begin{aligned} n &= \text{packet length in bits} \\ PER &= \text{packet error ratio} \\ c &= \text{delay } t_{rd} \text{ between end of transmission of last packet and start of next packet in seconds} \\ v &= \text{transmission rate in bit/s} \end{aligned} \tag{2.45}$$

the throughput D (more accurately the utilization of the channel) is

$$D = \frac{n(1 - PER)}{n + cv} \tag{2.46}$$

If an ideally fast and interference-free channel is used in a model, a window size of 1 will certainly be sufficient. If a delay of the confirmation occurs on the route between the transmitting layer-2 entity and the receiving partner entity, the pipelining principle can be advantageous. Delays occur when the transmission time over the medium being used is very high, interleaving is implemented in layer 1 or an ACK/NAK frame is lost. With pipelining, a window size >1 is required. Pipelining means that a limited number of other I-frames are being sent, even if confirmation is not provided.

Pipelining over a seriously unreliable channel presents some problems. What happens, for example, if an I-frame is lost in the middle of a long sequence? The subsequent I-frames will arrive at the receiver before the transmitter is able to establish that an error has occurred. There are two methods for mastering these situations effectively.

2.7.4.4 Go-Back-N-ARQ Protocol

The Go-Back-N-ARQ method is frequently called the REJ method or cumulative ARQ. With this ARQ method data flow between transmitter and receiver on a duplex channel is more or less continuous.

With a transmit window that is unlimited in size, data packets with consecutive packet numbers NS are transmitted until the transmitter receives a NAK[NS_e] for a packet that the receiver has recognized as being defective. When this occurs, the transmitter interrupts transmission and switches back to the defective packet with the number NS_e. The packet with packet number NS_e and all subsequent packets are then resent. The receiver ignores all other packets until packet NS_e is received.

With the REJ, the transmitter requires sufficient buffer in which to store all the unacknowledged data packets. This is why a transmit window of finite size is usually agreed upon. If the transmitter receives a positive acknowledgement ACK[NS] for a packet already sent then all the packets with sequence numbers less than or equal to NS are considered to have been transmitted correctly. The receive window selected is often larger than 1 to enable positive and negative acknowledgements to be sent (*piggy-back*) with an information packet from receiver to transmitter. Acknowledgements may also be sent before the receive window is completely filled up. A section of an REJ protocol process is shown in Figure 2.48.

The throughput with REJ-ARQ methods can be determined on the basis of the expressions introduced in Section 2.7.4.3:

$$D = \frac{n(1 - PER)}{n + PER \cdot cv} \tag{2.47}$$

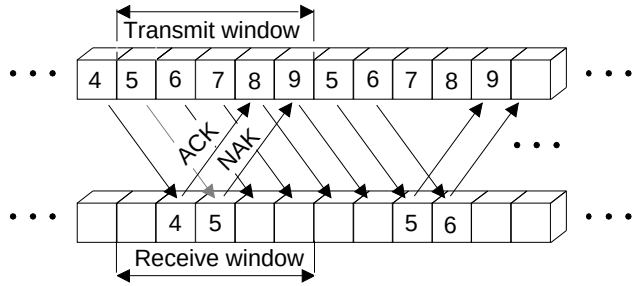


Figure 2.48: Go-Back-N-ARQ

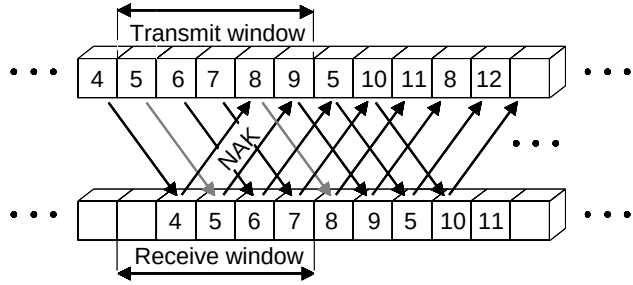


Figure 2.49: Selective-Reject-ARQ

2.7.4.5 Selective-Reject ARQ-Protocol

As with REJ, the objective of the SREJ method is to transmit packets between transmitters and receivers as continuously as possible. With this method, when the receipt of a defective packet is detected, only the packet that is defective is selectively requested. In contrast to the REJ method, the packets that arrive in the meantime are not rejected but are filed in a storage buffer. Therefore the transmitter only resends the data packets that were transmitted with errors.

At the same time the receiver must store a packet with the number NS_i until all packets with lower packet numbers have been correctly received. This is the only possible way to forward the packets in the correct sequence to the next highest ISO/OSI layer. For its part, the transmitter must keep the transmitted data packets in a storage buffer until a positive acknowledgement has been received for them, just as is the case with REJ. SREJ methods are not able to make full use of the transmitting window, because otherwise the uniqueness of the packet that has been requested again can be lost. Figure 2.49 shows the fundamentals of the SREJ process.

Assuming the availability of unlimited receiving storage capacity, the throughput with an SREJ-ARQ method amounts to

$$D = 1 - PER \tag{2.48}$$

Consequently the highest throughput of the three ARQ methods presented is achieved with SREJ.

2.7.4.6 Comparison Between FEC and ARQ Methods

The advantages and disadvantages of FEC and ARQ methods are summarized and compared here.

FEC methods

Advantages:
- No reverse channel required
- Constant throughput irrespective of channel quality
- Constant delay time between transmitter and receiver

Disadvantages:
- Low throughput due to high redundancy (low code ratio)
- Residual bit error ratio strongly dependent on channel quality
- Complicated coder and decoder algorithms required

ARQ methods

Advantages:
- High throughput when channel quality is good
- Guarantee of very low residual bit error ratio
- Less complex coder and decoder unit required

Disadvantages:
- Reverse channel required and additional signalling
- Throughput fluctuates depending on channel conditions
- Variable delay time, less suitable for real-time applications
- Additional storage needed by transmitter and receiver

2.7.4.7 Adaptive Coding

A mobile terminal experiences different channel qualities counted by the bit error ratio, depending on the current location, speed, etc., which is mainly characterized by the current receive signal level and C/I value at the mobile's receiver.

Since the channel coding defined for a mobile radio system has to take into account the worst possible situation in which the system should still be able to operate satisfactorily, the coding is overdimensioned in all the

situations, where a terminal is operating in an excellent, good or acceptable receive situation.

Clearly the channel coding should be adaptive. This requires that the code ratio be dependent on the current receive conditions so that a given quality of service (typically measured by parameters like bit-error ratio, throughput, delay, etc.) can be guaranteed.

The result of an adaptive coding is that the net bit rate of the channel is dependent on the receive situation of the mobile terminal, since the redundant parity bits would depend on the current needs.

With speech transmission where the bit rate is continuous in time, during time intervals where a small code ratio is adaptively chosen, the channel throughput would be too high. Silence periods could then be inserted for the transmission of speech codec data and the interference generated by use of the channel to other communicating terminals would be reduced. The use of intermittend speech transmission is already practised, e.g., when using packet-data-based voice, as is usual with CDMA systems, or using discontinous transmission, as is usual with the GSM system.

With data services a variable bit rate resulting from adaptive channel coding might be of advantage, too. This especially is the case with packet-oriented data services.

2.7.5 Hybrid ARQ/FEC Methods

ARQ methods provide a high level of transmission security, which can also be maintained when channels are severely disrupted albeit at the cost of throughput.

FEC methods provide continuous channel throughput, but the level of transmission security decreases the more error-prone the channel becomes. If the bit-error ratio of a channel is too high to achieve a required level of throughput using an ARQ method, and the code ratio of an FEC method is too small, then a combination of the two error-protection methods is applied.

This kind of combination in which an ARQ method is set up on an FEC system is called a hybrid ARQ/FEC method or an HARQ method. These methods combine the advantages of both techniques: the FEC system improves the residual bit error ratio of the channel; the ARQ technique eliminates any remaining errors. The principal structure of a hybrid ARQ/FEC system is presented in Figure 2.50.

The selection of the correct code allows the channel throughput, reduced because of the transmission of the redundancy bits of the FEC method, to be optimized despite the retransmission of disrupted packets.

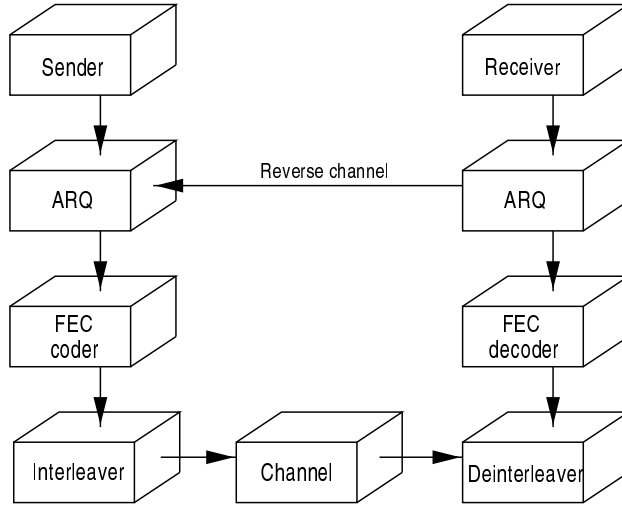


Figure 2.50: Hybrid ARQ/FEC system

2.8 Fundamentals of Random Access*

This section starts with a description of the random-access protocols that are essential for the initiation of contact between a mobile station and a radio network. First the Slotted-ALOHA access protocol will be examined in detail. This will be followed by an introduction to the methods described in the literature for control of random access in Slotted-ALOHA systems. These are important for the transmission of information (and not only for signalling) in radio networks that transmit using asynchronous time-division multiplexing.

2.8.1 Slotted-ALOHA Access Methods

The access method used for initial access by a mobile station in a mobile radio network is based on a Slotted-ALOHA protocol. The protocol has been covered in depth in numerous publications (e.g., [8] and [17]). Some of the important features of Slotted-ALOHA are explained here.

2.8.1.1 Fundamentals of ALOHA

The ALOHA access protocol is used in communications networks in which a large number of uncoordinated users are competing for the same channels. The basic idea behind an ALOHA system is simply that users are allowed to send data anytime they wish.

*With the collaboration of Dietmar Petras and Martin Steppler

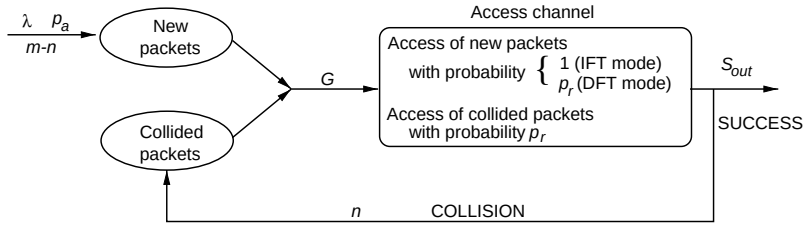


Figure 2.51: The Slotted-ALOHA access method (m and n are the number of not-collided and collided stations, respectively)

The protocol does not exclude the possibility of collisions, i.e., the fact that simultaneous or overlapping transmission by different stations can occur that cannot be decoded by the receiver. Collided packets must be resent. However, they cannot be resent until a randomly selected waiting time has passed; otherwise the same stations would collide again.

New packets are sent after they are created to ensure that there is as small a delay as possible with initial access (see Figure 2.51). If a collision occurs with other transmitted packets, the channel will be occupied but will have no throughput. Collided and successful packets together create a traffic load, which is measured by the arrival rate G (packets/slot). Two modes of transmission may be distinguished, IFT and DFT; see Section 2.8.3.2.

In contrast to *pure ALOHA*[1, 2], with the Slotted-ALOHA protocol transmission can only take place at the beginning of a *time slot* with a constant length. Therefore this is a time-discrete variation of the ALOHA protocol that offers double the amount of throughput S_{out}. If a situation with a large number of independent stations is assumed then the arrival process for new packets is a Poisson process (with an arrival rate λ). If collided packets are retransmitted after an independent random delay, the channel then experiences a Poisson arrival process at the rate G, and it can be shown [34] that the throughput at $G = 1$ reaches its maximum, which is $1/e = 0.368$. The probability that transmission will be successful is

$$S_{out} = Ge^{-G} \tag{2.49}$$

which is derived as follows and is presented graphically in Figure 2.52. Assuming an arrival rate $\lambda < e^{-1}$, the equilibrium point at G_1 initially becomes significant and the throughput is $S_{out} = \lambda$. However, the throughput at this point only constitutes an average value over a certain period of time. If the present traffic rate exceeds the value G_1 by a small amount, the throughput will be somewhat higher than λ. Consequently, data packets leave the system faster than they arrive, which causes the access rate to return to G_1. The system remains stable at point (G_1, λ) as long as the traffic data rate does not increase dramatically in the short term.

If the momentary arrival rate λ and consequently the traffic rate increase even further, e.g., to $G = 1$, then the system will become unstable (see the

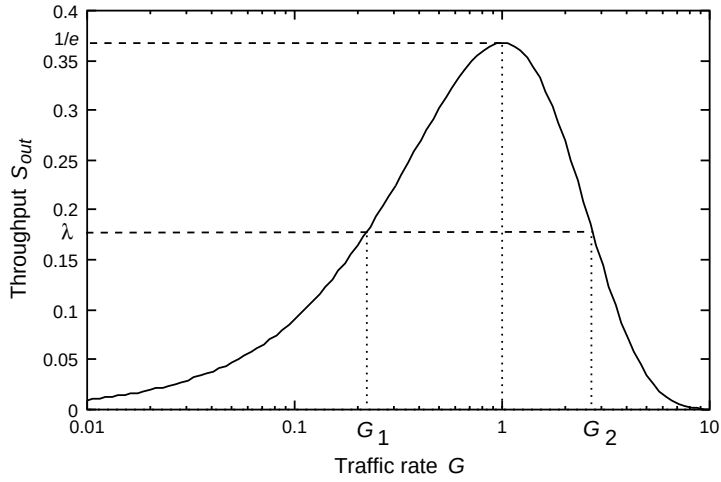


Figure 2.52: Throughput of Slotted-ALOHA

working point at G_2). But if the momentary arrival rate is somewhat lower than λ then the throughput S_{out} will also be lower than the arrival rate, and this will lead to an increase in the number of collided stations. When this happens, G continues to increase and the throughput S_{out} drops. This process continues: the throughput moves more and more towards 0. This is what makes the Slotted-ALOHA access protocol unstable.

It can also be shown [5] that the Slotted-ALOHA access method becomes unstable with fixed access probability per slot of $p < 1$ when the number of participating stations is very high. This instability can be avoided through the use of different approaches, as explained below.

2.8.1.2 Analysis of Slotted-ALOHA Access Methods

If the channel access rate G is normalized to the length of a time slot (G comprises new transmission attempts and retransmissions due to collisions), and a Poisson process is used to model all access, then the probability of a successful transmission attempt per slot can be described on the basis of Equation (2.49). The system is in a state of equilibrium when $S_{out} = \lambda$ (λ is also normalized to the slot duration). This formula offers no insight into the dynamic behaviour of a system, because feedback is produced owing to the number of collisions, which in turn alters the value of G. Generally there are too many unused slots if $G < 1$ and too many collisions if $G > 1$.

References [5, 10] describe a model for a Slotted-ALOHA system in which the stations directly waiting for the next transmission attempt are not accepting new data for transmission from their own application processes. The number m of stations is so large that the arrival process can be approximated as a Poisson process with rate λ. Furthermore, it is assumed that each sta-

tion is carrying out a renewed transmission attempt per slot for its collided packet with fixed probability p_r. Consequently the number i of slots between a collision and a renewed attempt is geometrically distributed with probability

$$P(I = i) = p_r(1 - p_r)^{i-1} \tag{2.50}$$

If n indicates the number of collided stations at the beginning of a particular slot then each of the n stations, independently of one another, is transmitting with probability p_r. Each of the other $m - n$ stations will send its packet in this slot if it has arrived at the station during the previous slot. Because the arrival processes for the new packets of each station likewise are Poisson processes with mean rate λ/m, the probability that no data is arriving is $e^{-\lambda/m}$; consequently the probability that a station that has not yet collided will send its packet in the slot in question is $p_a = 1 - e^{-\lambda/m}$.

If $P_a(i, n)$ is the probability that i packets that have not yet collided are being sent in a specific slot, and $P_r(i, n)$ is the probability that i packets that have already collided are being sent, then

$$P_a(i, n) = \binom{m-n}{i}(1 - p_a)^{m-n-i}{p_a}^i \tag{2.51}$$

$$P_r(i, n) = \binom{n}{i}(1 - p_r)^{n-i}{p_r}^i \tag{2.52}$$

From this assumption, it is possible to produce a discrete-time Markov chain in which the state variable stands for the number of collided stations and the slot duration is the unit of time. In each unit of time the value of the state variable increases by the number of collided stations in the respective slot, but remains the same if a new packet is successfully transmitted or the slot is unused, and drops by one if a collided packet is successfully transmitted.

A transmission attempt is successful if

- there is no packet that has already collided and a newly arrived packet is sent in the slot in question;
- no packet has arrived and a collidet packet is is sent in the slot in question.

Consequently, the transition probability with a transition from state n to state $n + i$ can be expressed through the following equations [5, 10].

- If two or more packets arrive during a slot then

$$P_{n,n+i} = P_a(i, n)\,, \qquad 2 \le i \le (m - n) \tag{2.53}$$

- With exactly one newly sent packet and at least one resent collided packet,

$$P_{n,n+1} = P_a(1, n)\,[1 - P_r(0, n)] \tag{2.54}$$

- With exactly one newly arrived packet and no attempt to resend packets that have already collided or with no newly arrived packet and no successful attempt to resend a packet that has already collided, the following applies:

$$P_{n,n} = P_a(1,n)P_r(0,n) + P_a(0,n)\left[1 - P_r(1,n)\right] \tag{2.55}$$

- With exactly one attempt to send a packet that has already collided,

$$P_{n,n-1} = P_a(0,n)P_r(1,n) \tag{2.56}$$

Let D_n be the expected difference in the number of collided stations for the next slot in state n from the updated number of collided stations. Therefore D_n in state n consists of the anticipated number of new transmitting stations, i.e., $(m-n)p_a$, minus the anticipated number of stations transmitting successfully together in a specific slot, which corresponds to the probability of a successful transmission attempt and is denoted by P_{succ}. Thus

$$D_n = (m-n)p_a - P_{succ} \tag{2.57}$$

with

$$P_{succ} = P_a(1,n)P_r(0,n) + P_a(0,n)P_r(1,n) \tag{2.58}$$

If G(n) is the number of transmission attempts in one slot in state n then

$$G(n) = (m-n)p_a + np_r \tag{2.59}$$

From an analysis of the Markov chain, P_{succ} (on the condition that p_a and p_r are low) can be approximated (see [5]) as

$$P_{succ} \approx G(n)e^{-G(n)} \tag{2.60}$$

with the probability of an empty slot being about $e^{-G(n)}$. This approximation therefore confirms the above assumption of G as the parameter for a Poisson distribution (see the result in Equation (2.49)).

The relationships that have just been explained are presented graphically in Figure 2.53 for $p_r < p_a$. In Equation (2.57) the deviation D_n is the distance between the curve that represents successfully transmitted data packets and the straight line that describes the arrival rate. Two stable states of equilibrium are evident.

Let us consider the effects of varying the parameter p_r. If p_r is increased then the delay until the new transmission of collided packets is reduced. However, because of the linear dependence between n and the rate $G(n) = (m-n)p_a + np_r = mp_a + n(p_r - p_a)$, $G(n)$ also increases by n if p_r is increased and $p_r > p_a$. If the horizontal axis in Figure 2.53 is considered for a fixed n, this change to G corresponds to a shortening of the horizontal scale for G and therefore a horizontal compression of the curve Ge^{-G}. This means

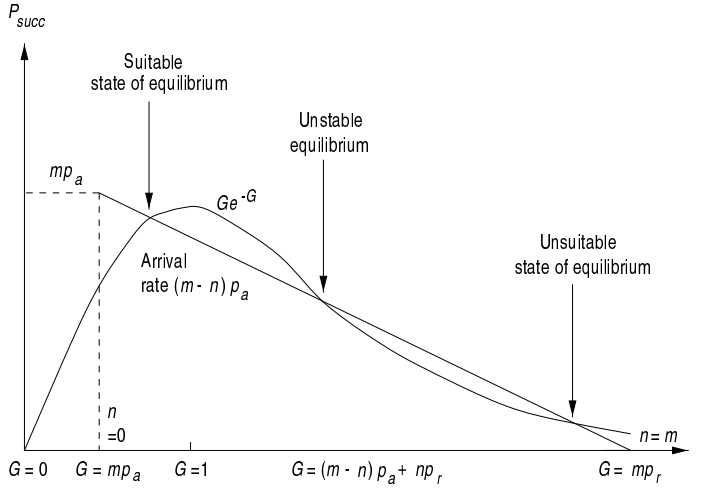


Figure 2.53: Stability in Slotted-ALOHA

that there is a decrease in the number of collided packets that are needed in order to achieve an unstable equilibrium.

If, on the other hand, p_r is decreased, the delay will be higher with the renewed attempt to transmit (see Section 2.8.3), although a state of unstable equilibrium will not be reached as quickly. With a reduction of p_r, the graph of Ge^{-G} widens. This means that a stable equilibrium point is now only located in the area of the straight lines. This area has a large number of the m stations in a collided state, which means that no new data can be sent from these stations. This has a negative effect on the delay of the newly generated packets there.

The proportion of successfully transmitted packets P_{succ} must be controlled if stability is to be achieved. As Equation (2.60) clearly shows, the maximum is $G(n) = 1$. Therefore it is desirable to change the rate $G(n)$ dynamically so that it is as close to 1 as possible. The difficulty in doing so is that n is not known by stations and can only be approximated by them if they receive feedback.

2.8.1.3 Stability Aspects in Slotted-ALOHA

The qualitative statements made in the previous section will be covered in more detail in this section.

First an analysis will be made of how probable it is that a transmitted data packet will be received successfully. Equations (2.51) and (2.52) will be used for this purpose. The probability of one of the m stations transmitting a new packet in a slot and not belonging to the n stations that have already transmitted a packet unsuccessfully and are in the collision state is $p_a = \lambda/m$.

The stations undertaking another attempt to send their packets transmit with the probability p_r.

If the transmission result in the kth slot is indicated as e_k then

$$e_k = \begin{cases} 0, & \text{if no station is transmitting in the } k\text{th slot} \\ 1, & \text{if exactly one station is transmitting in the } k\text{th slot} \\ c, & \text{if two or more stations are transmitting in the } k\text{th slot} \end{cases} \tag{2.61}$$

It is assumed that e_k in the kth slot is known and can be used to establish the transmission probability for the collided stations in the $(k+1)$th slot.

The throughput of n collided stations can be indicated by the probability that exactly one data packet will be sent in slot k. This is based on the probability that exactly one of the n previously collided stations is transmitting but no station is transmitting for the first time, and the probability that exactly one station is transmitting for the first time but none of the n collided stations is transmitting data packets. Therefore the probability of a successfully transmitted data packet is

$$\begin{aligned} P(e_k = 1 \mid n \geq 1) &= P_a(1,n)P_r(0,n) + P_a(0,n)P_r(1,n) \\ &= (m-n)\left(1-\frac{\lambda}{m}\right)^{m-n-1}\frac{\lambda}{m}(1-p_r)^n \\ &+ \left(1-\frac{\lambda}{m}\right)^{m-n} np_r(1-p_r)^{n-1} \end{aligned} \tag{2.62}$$

It is assumed here that there is at least one station in the system that has already collided. For a fixed arrival rate $\lambda = 0.2$ and a finite number of 25 stations, the probability of successful access can be shown graphically (see Figures 2.54 and 2.55). The probability of successful access can also be viewed as the momentary achievable throughput. From this diagram, it can be seen that for a certain number of collided stations (*backlog*) n the corresponding transmission probability is $p = p_r$, which, in the case being considered, possesses the greatest probability for successful access. So Figure 2.54 shows that with 10 collided stations the momentary throughput for $p = 0.1$ is approximately 0.38, whereas for $p = 0.25$ it is only about 0.17 and for $p = 0.5$ it is almost equal to 0.

Figure 2.55 shows a three-dimensional representation of expected throughput as a function of backlog n and the arrival rate λ for the different transmission probabilities $p = 0.5,\ 0.25$ and 0.1.

The relationships shown above are analyzed more precisely in [17] with respect to the stability of the access channel. The assumption there is that there are m users in the system and that non-collided stations are generating new packets with the probability of p_a. Consideration should be given to the (n, λ) level in which the straight line $\lambda = (m-n)p_a$ is indicated as a straight traffic line. For the fixed value p an equilibrium curve is described through the quantity of points for which the arrival data rate λ is exactly equal to the

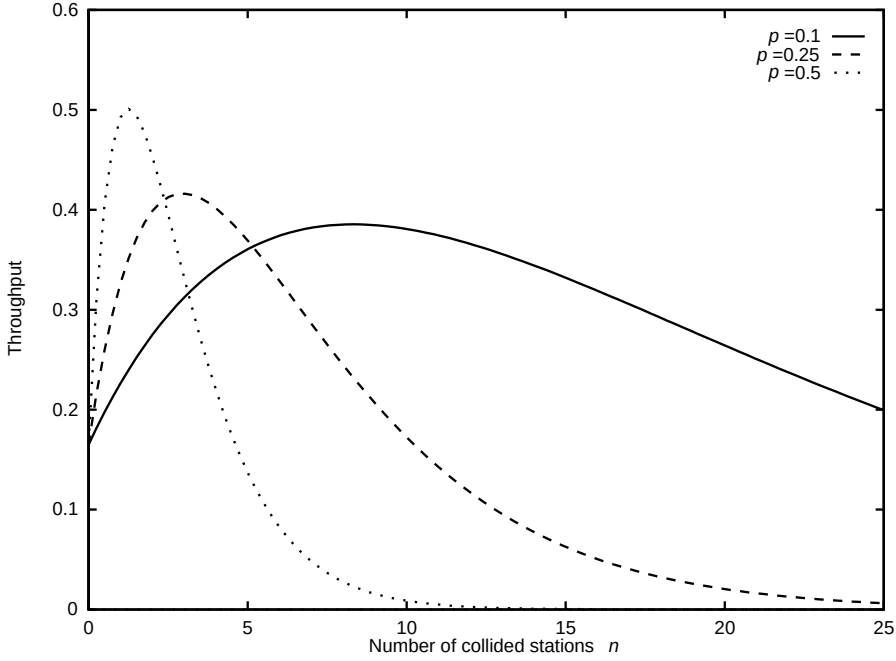


Figure 2.54: Throughput with $\lambda = 0.2; m + n = 25$

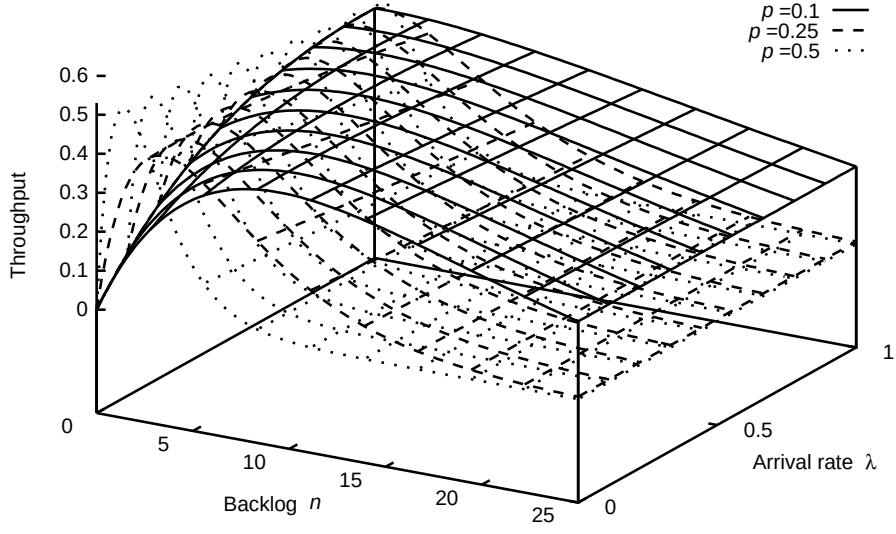


Figure 2.55: Representation of throughput Equation (2.62)

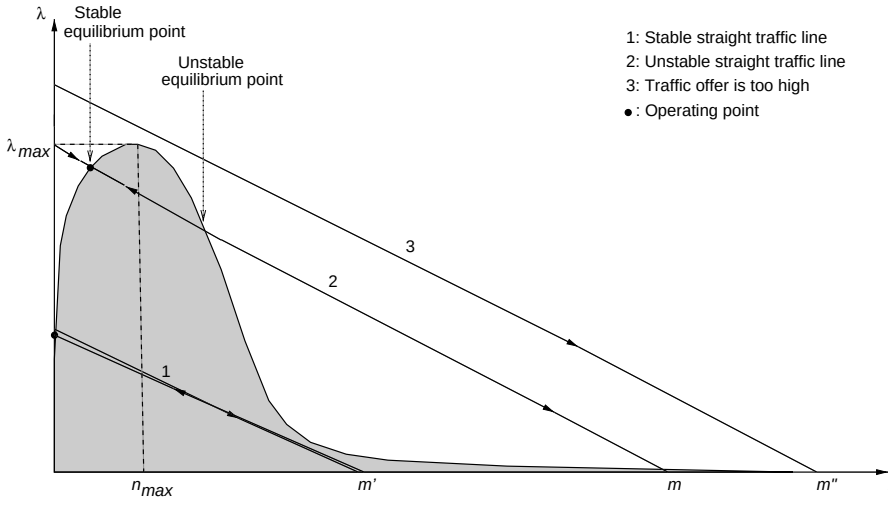


Figure 2.56: Stable and unstable data traffic

anticipated throughput rate $S_{out}(n, \lambda)$. An example of this curve is illustrated in Figure 2.56.

Within the shaded area in the figure, $S_{out}(n, \lambda)$ is greater than λ; outside this area, λ is greater than $S_{out}(n, \lambda)$. Three traffic lines corresponding to station numbers m, m' and m'' have been drawn. The arrows on the traffic lines indicate the direction (tendency) in which the number n develops. A traffic line can intersect with the equilibrium curve at several points. These points should be indicated as equilibrium points (n_e, λ_e). An equilibrium point is described as being *stable* if it can be regarded as a sink in respect to tendency n; or it can be regarded as *unstable* if the point considered has the characteristics of a source.

A stable equilibrium point is described as a working point if $n_e \leq n_{max}$; on the other hand, it is referred to as a saturation point if $n_e > n_{max}$. A traffic line is considered stable if it has exactly one stable equilibrium point; otherwise it is considered unstable. In a stable channel the point (n_e, λ_e) determines the throughput and access delay for a limited period of time. On the other hand, an unstable channel displays a *bistable* quality [8]; the throughput there is only achievable for a limited period of time before the working point moves in the direction of a saturation point. Line 3 in Figure 2.56 has a saturation point as its only stable equilibrium point. Therefore the traffic offered on the basis of m'' stations is too high with the given values for p_a and p.

Taking a specific traffic line, p_o can be considered as the optimal transmission probability (this will be explained in more detail in the following sections), where n is minimized in the working point and λ maximized. This value of p can cause the system to become unstable; optimal throughput is only achievable for a limited period of time. The following measures can be

undertaken to stabilize the access channel: (1) selection of a smaller value for p or (2) reduction in the number m of stations allowed to transmit. The first option produces a higher value for n in the working point, which could result in making the access delay unacceptably high; the second option implies that $\lambda \ll \lambda_{max}$ (and thus $p_a \ll 1$) in the working point. However, this would mean a waste of channel capacity. One way to prevent the system from moving into a range of unstable equilibrium is to have it instruct the stations to discard all packets that are in a colliding state.

A better approach is to use the dynamic control algorithms discussed below.

2.8.2 Slotted-ALOHA with Random Access Frames

In most mobile radio systems with FDM/TDM channels the random access channel is not continuously available during frame length FL but only at certain times. The individual mobile stations are able to transmit spontaneously on the access channel over an interval of the length of FL slots. We shall look more closely at how access frames are structured with different user numbers and traffic rates. Consideration will also be given to the fact that slots used for reservations and signalling data are not available for random access.

With the access methods analysed here a base station sends information over access parameters, including frame length FL, to all stations at regular intervals. For its first access attempt a mobile station transmits in the next available slot. It then waits a certain period of time for confirmation from the base station. If the mobile station does not receive confirmation during this period of time (because of a collision or transmission error), it waits until the beginning of the next access frame and then randomly accesses one of the FL slots for a renewed attempt. The base station is able to control the value n of collided stations in such a way that the access channel is not overloaded because of too many collisions.

In the model being considered, two types of data traffic coexist on the access channel. In addition to the slots for random access by the mobile stations, there are slots that are either reserved for data or are used for control information. The issue of how the configuration of the access channel with these different types of data affects system behaviour will also be discussed.

2.8.2.1 Scenario with Fixed Frame Length

Reference [4] examines Slotted-ALOHA access methods for fixed frame lengths. The analysis particularly considers system stability with an average number of collided stations. An optimal value for the access frame length FL (in slots) is deduced from the analysis.

The analysis is based on an infinite number of stations, so that the packet arrival rate λ follows a Poisson process. Therefore the probability that k stations will be transmitting in a slot is

$$p_a(k) = \frac{\lambda^k}{k!} e^{-\lambda} \tag{2.63}$$

At the beginning of an access frame n collided stations from the previous frame attempt to transmit equally distributed in one of the FL slots of the current frame. The probability that a station will transmit in a particular slot is therefore $p = 1/(FL)$, and the probability that i of the n stations will transmit in a slot is derived from the binomial distribution:

$$p_{in} = \binom{n}{i} \left(\frac{1}{FL}\right)^i \left(1 - \frac{1}{FL}\right)^{n-i} \tag{2.64}$$

If the average number of collided stations per access frame is specified as N then this value is made up of the three contributions N_1, N_2, N_3. The portion N_1 consists of the stations that were not successful in the previous access frame and collided again with the new frame. If i is the number of collided stations from the previous access frame that are transmitting again in the considered slot (and collide again) and k is the number of stations that want to transmit for the first time in the considered slot then the number of stations colliding in this slot is equal to

$$n_{c1} = \sum_{k=0}^{\infty} (k+i)\, p_a(k) \qquad (i \geq 2) \tag{2.65}$$

The average number N_1 of stations that are colliding again is then

$$N_1 = FL \cdot \sum_{i=2}^{n} p_{in} \sum_{k=0}^{\infty} (k+i)\, p_a(k) \tag{2.66}$$

The contribution N_2 includes the collided stations in the previous frame as well as the stations that are transmitting their packets for the first time in the considered slot:

$$N_2 = FL \cdot p_{1n} \sum_{k=1}^{\infty} (k+1)\, p_a(k) \tag{2.67}$$

The contribution N_3 describes a case in which two or more stations are transmitting for the first time in a particular slot and are colliding. Therefore an access frame that is FL slots in length gives

$$N_3 = FL \cdot p_{0n} \sum_{k=2}^{\infty} k p_a(k) \tag{2.68}$$

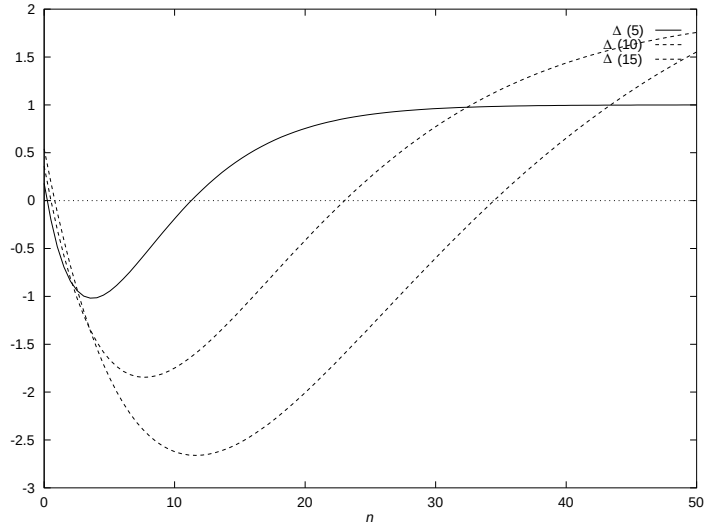


Figure 2.57: Difference $\Delta(FL)$ in the number of collided stations in successive frames when $\lambda = 0.2$

The total number of collided stations per access frame is $N_1 + N_2 + N_3$, and according to [4] can be described as follows:

$$N = FL \cdot \left\{ \lambda - \left(1 - \frac{1}{FL}\right)^{n-1} e^{-\lambda} \left[\left(1 - \frac{1}{FL}\right) \lambda + \frac{n}{FL} \right] \right\} + n \qquad (2.69)$$

The difference Δ of the number of collided stations from one access frame to the next one is

$$\begin{aligned} \Delta &= N - n \\ &= FL \cdot \left\{ \lambda - \left(1 - \frac{1}{FL}\right)^{n-1} e^{-\lambda} \left[\left(1 - \frac{1}{FL}\right) \lambda + \frac{n}{FL} \right] \right\} \end{aligned} \qquad (2.70)$$

The minimum of Δ works out as

$$n_{min} = -\frac{\lambda\,(FL-1)\ln\left[1 - 1/(FL)\right] + 1}{\ln\left[1 - 1/(FL)\right]} \qquad (2.71)$$

If the ln function in Equation (2.71) is expanded into a Taylor series (with the condition $FL \gg 1$), it follows that

$$n_{min} = FL \cdot (1 - \lambda) + \lambda \qquad (2.72)$$

Δ for an arrival rate $\lambda = 0.2$ and frame lengths FL of 5, 10 and 15 slots is illustrated in Figure 2.57.

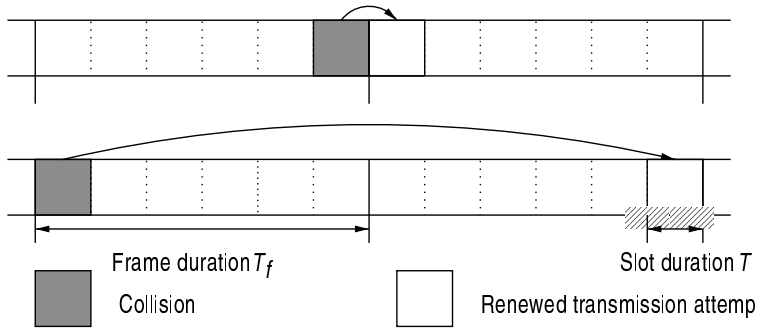


Figure 2.58: Distribution of the delay after a collision

This shows that $\Delta(n)$ has two zero values. Furthermore, it is obvious that the first zero value (Z_1) is a stable equilibrium position. Reference [4] has examined system behaviour using different arrival rates and frame lengths as parameters. What emerged was that a larger value for FL also increases the area in which Δ is negative, and, as expected, this contributes towards the stabilization of a system. It was proved that a system is unstable with a fixed frame length—something that is already known from frame-less systems.

In addition to stability, another important parameter of the access method that is of interest is the average access time T_m, which in [4] is derived from the mean waiting time T_c between collisions of the same packet and the number of all packets n_t sent per frame:

$$T_m = \frac{T_c}{n_t} \tag{2.73}$$

Until now the implication has been that access frames follow each other seamlessly and have the duration T_f. As Figure 2.58 shows, the delay after a collided slot is evenly distributed over an interval between 0 and $2T_f$ and therefore $T_c = nT_f$. So Equation (2.73) is expressed as

$$T_m = \frac{\sum_{i=1}^{N_f} n_i T_f}{GN_f \cdot FL} \tag{2.74}$$

with n_i standing for the number of collided packets in the ith slot, N_f the number of preceding frames and G the average number of packets generated during a time slot of length T. The average number of collided packets during a frame is

$$N = \frac{1}{N_f} \sum_{i=1}^{N_f} n_i \tag{2.75}$$

and therefore it follows that T_m is

$$T_m = \frac{T}{G}N = \frac{N}{\lambda} \tag{2.76}$$

with λ denoting the number of packets generated per unit of time.

It is clear from Equation (2.76) that T_m is not directly dependent on FL, but is indirectly dependent on the parameter N, which according to Equation (2.69) is dependent on FL.

2.8.2.2 Scenario with Variable Frame Length

When frame length FL is varied based on the number of collided stations n, it should be noted that although an increase of FL reduces the number n, it also simultaneously increases the access time.

In the previous section we learned how to minimize the difference Δ with a specific arrival rate λ through selection of the frame length FL. The optimal frame length that follows from Equation (2.71) is

$$FL = \frac{n - \lambda}{1 - \lambda} \tag{2.77}$$

This frame length can be considered an adaptively gained optimal frame length for the momentary state of a system. However, the values for λ and n cannot be determined directly by the base station. Within the framework of the different algorithms that optimize throughput in Slotted-ALOHA using the above optimal frame length (or transmission probability), the next section will describe how these values can be determined using different estimation algorithms based on a sequence of empty, collided or successfully transmitted slots.

From Equations (2.77) and (2.70) the difference would be, with a variable frame length,

$$\Delta_v = \frac{n - \lambda}{1 - \lambda}\left[\lambda - \left(\frac{n-1}{n-\lambda}\right)^{n-1} e^{-\lambda}\right] \tag{2.78}$$

Δ_v is negative for $\lambda < 1/e$ and decreases with increasing n; thus the system is stable. With higher values of λ, however, Δ_v is positive and increases with n; when this is the case, the system is unstable.

Since the analysis assumes an unlimited number of users (to enable the rate λ to be used as a constant), and in practice the frame length FL cannot be varied between 0 and ∞, the result shown in Equation (2.78) should be used with caution.

2.8.2.3 Configuration of an Access Channel

If $FL = A$ is the number of slots available for random access, and lA ($l > 1$) is the number of reserved slots in a frame, then the frame consists of $A+lA$ slots.

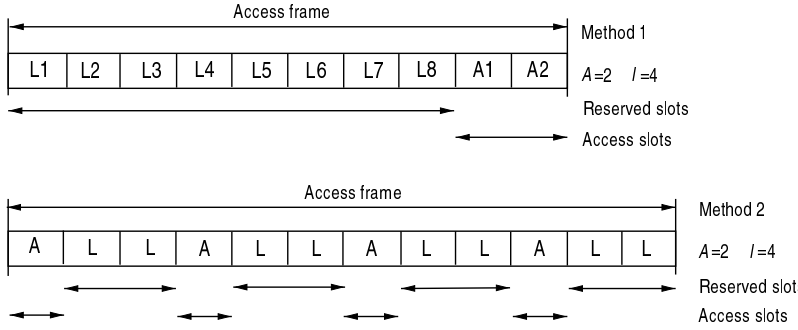


Figure 2.59: Different uses of an access frame

The reserved slots can be positioned in different ways in the frame. Reference [4] explores frames in which reserved slots are grouped at the beginning of each frame (method 1) as well as a configuration in which the slots that are reserved and appropriate for access are arranged in alternating order (method 2). This situation is clarified in Figure 2.59.

The difference Δ can also be established analytically and can be determined as a function of n with set parameters λ and FL. The function $A = f(n)$ looks similiar to the one in Figure 2.57, although it is shown that the curve for method 2 already has its first zero positioned at lower values for n and also has a more distinct minimum. Therefore it proves to be more appropriate. It is also possible to visualize this situation by considering that no random access is permitted during the phases of reserved slots and that all stations receiving data during this time will access the first slot released after the reserved period. So, after a long reserved period, there is a backlog of access requests and consequently an increase in the number of collisions in this time slot.

It can be shown [4] that the optimal frame length with method 2 is

$$A = FL_{M2} = \frac{n - \lambda(l+1)}{1 - \lambda(l+1)} \tag{2.79}$$

and produces a minimum number of collided stations. With this method the system only operates in a stable state if the arrival rate is

$$\lambda < e^{-(l+1)} \tag{2.80}$$

2.8.3 Access Delay with Slotted-ALOHA

An important parameter of access algorithms is the access delay. It plays an important role with regard to time-critical services. In some cases it is more important to control the system so that the access delay remains within a particular time requirement than to maximize throughput. As was shown in

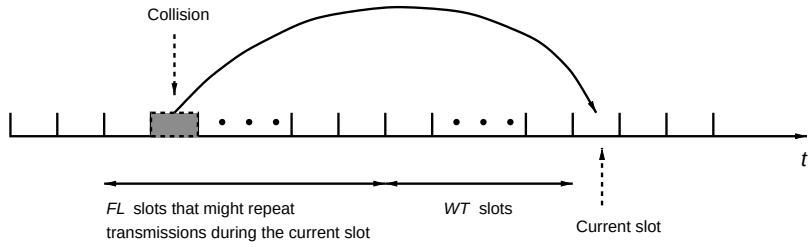


Figure 2.60: Repetition of a collided packet

the previous section, an algorithm that stabilizes the system to an optimum also minimizes the access delay at the same time.

Therefore the following discussion describes how access delay is dependent on the parameters waiting time WT and frame length FL. This is followed by a description of an algorithm used to determine the access delay from the calculated approximation for the arrival rate λ and the backlog of collided packets n.

2.8.3.1 Derivation of Access Delay

The access delay denotes the period of time from the first attempt to transmit to the completion of successful transmission including the duration of the transmission. A Slotted-ALOHA system with a finite number of stations is considered below. Data packets are produced at the total arrival rate λ. Stationary equilibrium is assumed. If a packet is not acknowledged after a fixed period of WT slots, it is resent k slots later, with k being randomly selected uniformly from the FL slots (see Figure 2.60).

The average delay time D of a packet is defined in [15] as the packet transmission duration (1 slot) plus the acknowledgement time WT when transmission has been successful after the first attempt. With R repeated transmissions, R times the average waiting time between two attempts plus the transmission duration and acknowledgement time are added. Therefore

$$D = WT + 1 + \frac{1}{2} + R\left(WT + 1 + \frac{FL - 1}{2}\right) \tag{2.81}$$

with half the slot taking into consideration the average waiting time of a new packet at the beginning of the slot. In Equation (2.81) the value of R is an unknown quantity. In the stationary equilibrium the channel traffic G is $R+1$ times larger than λ:

$$G = \lambda(1 + R) \tag{2.82}$$

R is established by defining p_n as the probability for the successful transmission of newly generated packets and p_r as the success probability of a previously collided packet. Thus the following applies for the probability $P\{i\}$

that exactly i repetitions are required until successful transmission has taken place:

$$P\{i\} = (1 - p_n)\, p_r\, (1 - p_r)^{i-1} \tag{2.83}$$

The average number of repetitions is then

$$R = \sum_{i=1}^{\infty} iP\{i\} = \frac{(1 - p_n)\, p_r}{\left[1 - (1 - p_r)\right]^2} = \frac{1 - p_n}{p_r} \tag{2.84}$$

Here p_n and p_r are unknown probabilities, which have been derived in [10]:

$$p_r = \frac{1}{1 - e^{-G}} \left(e^{-\frac{G}{FL}} - e^{-G}\right) \left(e^{-\frac{G}{FL}} + \frac{G}{FL} e^{-G}\right)^{FL-1} e^{-\lambda} \tag{2.85}$$

$$p_n = \left(e^{-\frac{G}{FL}} + \frac{G}{FL} e^{-G}\right)^{FL} e^{-\lambda} \tag{2.86}$$

The unknowns in Equations (2.81) and (2.82) are derived on this basis, and the delay can be deduced from the arrival rate and the momentary channel traffic in the following two equations:

$$D = WT + 1 + \frac{1}{2} + \frac{1 - p_n}{p_r} \left(WT + 1 + \frac{FL - 1}{2}\right) \tag{2.87}$$

$$\lambda = \frac{G}{1 + R} = \frac{G p_r}{1 + p_r - p_n} \tag{2.88}$$

Equations (2.87) and (2.88) have been evaluated numerically in [15] for a finite number of stations, and are represented for different frame lengths in Figure 2.61.

2.8.3.2 Algorithms for Determining Access Delay

This section presents a method adapted from [10] that enables access delay to be calculated from the system values for arrival rate and backlog.

Until now packet delay D has been defined as the time between the first attempt of transmission and completion of successful transmission. The following example includes the time from when the packet was generated up to the the first attempt to transmit. The Slotted-ALOHA access method makes a distinction between two different variants.

In IFT (*immediate first transmission*) mode a newly generated packet is directly transmitted in the next slot. In DFT (*delayed first transmission*) mode a newly generated packet is delayed by a specific waiting time. The waiting time can be useful, especially for preventing collisions of the new packets with data that is being resent after an interval of reserved slots. Packet delay in IFT mode will be discussed first.

Two distributions are required in order to establish packet delays: the distribution of the waiting time between two transmission attempts and the

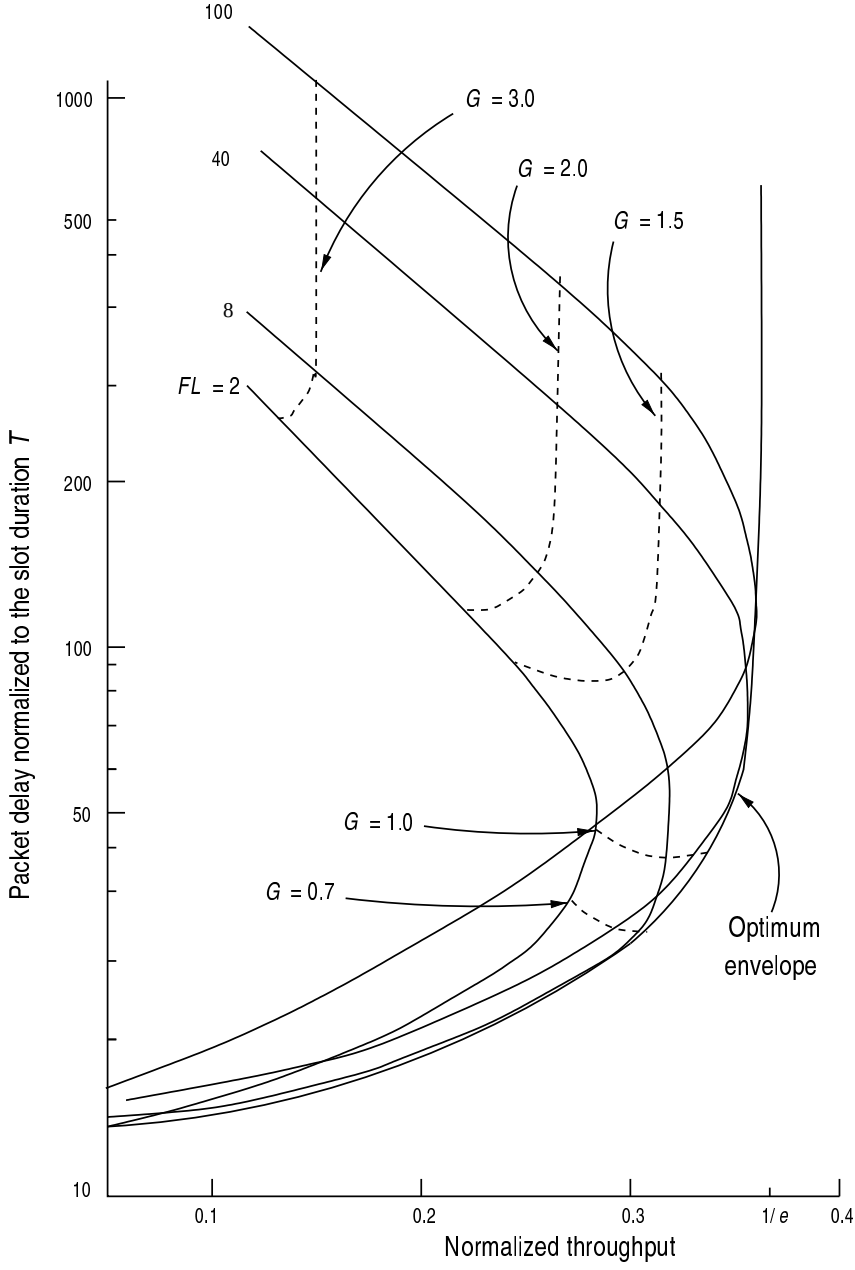


Figure 2.61: Packet delay versus throughput

distribution of the number of repeated transmissions required until successful transmission has taken place. If Z_i is the waiting time in slots after the ith attempt then $P\{Z_i = x\}$ is the waiting time distribution between the ith and $(i+1)$th attempts. If, furthermore, R indicates the random variable of the number of repeated attempts then the probability $P\{D = x\}$ is that the packet delay equates to x slots:

$$\begin{aligned} P\{D=x\} &= P\{R=1\}P\{Z_1=x\} \\ &+ P\{R=2\}P\{Z_1+Z_2=x\} \\ &+ P\{R=3\}P\{Z_1+Z_2+Z_3=x\} \\ &+ \ldots \\ &+ P\{R=y\}P\{Z_1+Z_2+Z_3+\ldots+Z_y=x\} \end{aligned} \quad (2.89)$$

with $P\{R = y\}$ representing the distribution of the number of transmission attempts of a packet until transmission was successful. The distribution $P\{Z_1 + Z_2 + Z_3 + \ldots + Z_y = x\}$ in Equation (2.89) describes the probability that the entire waiting time $Z_1 + Z_2 + Z_3 + \ldots + Z_y$ between all y repeated transmissions together equals x slots, with y specifying the maximum number of repeated attempts. In some mobile radio systems, e.g., in the TETRA standard (see Section 6.3), the parameter $N_u = y$ is available for this purpose. Both distributions are derived on the basis of access parameters y and WT.

The probability that y repeated transmissions of a packet will occur is equal to the probability that the packet being considered has collided y times with packets from other stations before successful transmission took place. The probability $P\{A\}$ for the successful transmission of a packet in a Slotted-ALOHA system with an unlimited number of stations of which n are momentarily in a collided state can be approximated by replacing the arrival probability assumed to be binomially distributed; see Equation (2.64) with a Poisson distribution according to Equation (2.63). The probability of success is then

$$P\{A\} = \lambda e^{-\lambda}\left(1-\frac{1}{FL}\right)^n + e^{-\lambda} n \frac{1}{FL}\left(1-\frac{1}{FL}\right)^{n-1} \quad (2.90)$$

The probability that transmission will not be successful is therefore $P\{\bar{A}\} = 1 - P\{A\}$. This defines the distribution of the number of repeated attempts as

$$P\{R=y\} = P\{\bar{A}\}^y P\{A\} = (1-P\{A\})^y P\{A\} \quad (2.91)$$

The final element that still needs to be determined is the waiting time distribution of a packet whereby the waiting time in y waiting intervals together equals x slots.

First the probability that the time between a collision and a renewed attempt to transmit equals x slots is deduced on the basis of WT. This involves

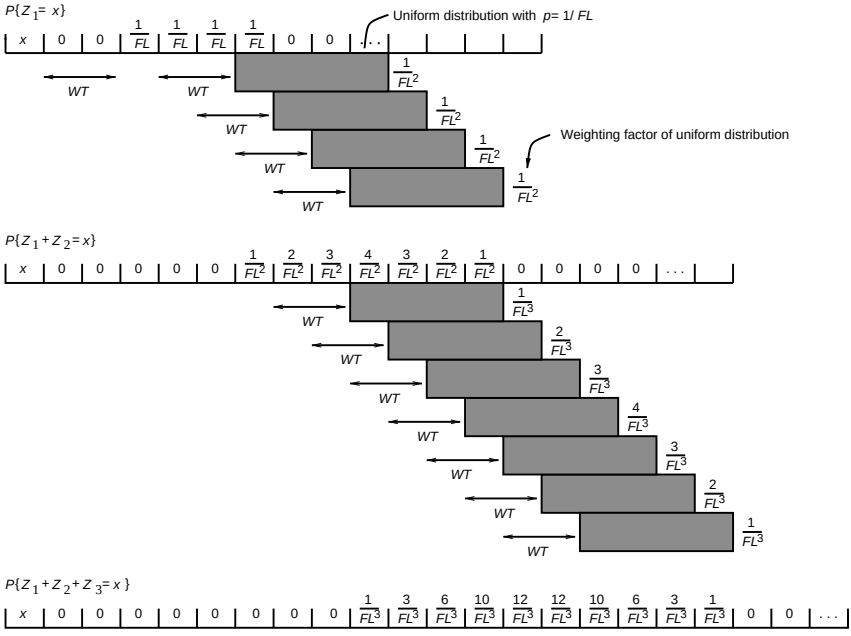


Figure 2.62: Waiting time distribution with $WT = 2$ and $FL = 4$

expressing the random distribution $P\{Z_1 = x\}$ independent of WT and FL as a discrete sequence of numbers. Thus

$$P\{Z_1 = x\} = \begin{cases} 0, & \text{for } 0 \le x \le WT \\ 1/(FL), & \text{for } WT < x \le WT + FL \\ 0, & \text{for } x > WT + FL \end{cases} \tag{2.92}$$

As the example in Figure 2.62 shows, the probability distribution with two collisions is produced from the discrete convolution of the probability distribution with one collision. Thus

$$P\{Z_1 + Z_2 = x\} = P\{Z_1 = x\} * P\{Z_1 = x\} \tag{2.93}$$

The probability distribution with three collisions is obtained in turn by discretely convoluting the probability distribution of two collisions with the probability distribution of one collision. The corresponding result is then

$$P\{Z_1 + Z_2 + Z_3 = x\} = P\{Z_1 + Z_2 = x\} * P\{Z_1 = x\} \tag{2.94}$$

The following is then obtained for y collisions:

$$P\{Z_1 + Z_2 + Z_3 + \ldots + Z_y = x\} = \underbrace{P\{Z_1 = x\} * P\{Z_1 = x\} * \cdots * P\{Z_1 = x\}}_{y \text{ times}} \tag{2.95}$$

This establishes the two unknown probability distributions in Equation (2.89). The mean packet delay is ultimately derived from the following equation:

$$E[D] = \sum_{x=1}^{\infty} xP\{D = x\} \tag{2.96}$$

2.8.4 Algorithms for Collision Resolution with S-ALOHA

Strategies that have been described in the literature and are used to stabilize Slotted-ALOHA access procedures while optimizing throughput are introduced below.

First we shall introduce some simple procedures that always involve evaluating the usage of the last slot, and on the basis of this evaluation we shall determine the new access probability for stations in a state ready to transmit. This is followed by a discussion of the more complicated algorithms that determine the transmission probability on the basis of a method for estimating the *backlog*, i.e., the number of collided packets.

The methods presented in this section can be used to control the access probability of stations centrally, thereby stabilizing a Slotted-ALOHA system, keeping access delays to a minimum and maximizing throughput.

2.8.4.1 Exponential-Backoff Algorithm

Backoff algorithms (for controlling renewed transmission of a collided packet) are very simple methods for achieving stability. They are used, for example, in standard IEEE 802.3-Ethernet-LANs. With this method the access channel is controlled in such a way that the access probability in each station is determined as $p_r = 2^{-i}$ on the basis of the number of collisions i of a packet, which means that access is distributed equally over the next 1 to 2^i slots. This algorithm extends the period of time in which a packet can be retransmitted by the number of collisions, thereby controlling the length of the access frame (which, however, ends here after successful access has taken place).

In the method suggested in [14], a station observes an access channel and determines whether the last slot has been accessed successfully, a collision has occurred or the slot has remained empty. If the number of stations transmitting in slot k is denoted by $z(k)$ then the result is $z(k) = 0$, $z(k) = 1$ or $z(k) \geq 2$ depending on whether the last slot was empty, transmission was successful or a collision occurred. With $p_r(k)$ as the transmission probability and I_A as the state function in slot k, where A is defined by $z(k)$,

$$p_r(k+1) = \min\left\{p_{max}\ ,\ p_r(k)\left(\frac{1}{q}I_{z(k)=0} + I_{z(k)=1} + qI_{z(k)\geq 2}\right)\right\} \tag{2.97}$$

in which $0 < p_{max} \leq 1$, $0 < q \leq 1$, and I_A has the value 1 if A occurs; otherwise I_A equals zero.

The above algorithm implies that the station serving as an example considers the momentary value of p to be too low (high) because of an empty

(collided) slot and therefore increases (decreases) the value of p. The value of p_{max} is 1, and the values $\frac{1}{4}$, $\frac{1}{2}$ and $\frac{1}{\sqrt{2}}$ are examined in [14] for the parameter q. Since this paper likewise assumes that newly generated packets are also being sent with the probability p, the approach used is the DFT mode of Slotted-ALOHA, and the throughput can immediately be determined using the fixed parameter p_r. If P_r in Equation (2.52) is taken for the value $i = 1$, since all stations are transmitting with the same probability, it follows that the throughput S_{DFT} of the Slotted-ALOHA protocol with fixed p_r is given by

$$S_{DFT}(m) = \binom{m}{1} p_r (1 - p_r)^{m-1} \tag{2.98}$$

For $m \geq 2$ the exponential backoff method can be described as a discrete Markov chain [14]. The system states $X_k (k = 0, 1, ...)$ in this description are selected so that $p_r = q^k$. If π_k is the state probability of the state X_k then it follows from the analysis of the Markov chain that the throughput of the exponential backoff algorithm is given by

$$S_{EB}(m) = \sum_{k=0}^{\infty} \binom{m}{1} p_r (1 - p_r)^{m-1} \pi_k \tag{2.99}$$

According to Equations (2.98) and (2.99) in [14], throughput is calculated as a function of n with different parameters p_r or q, with π_k being deduced from a recursion equation. The method with fixed p_r shows that with $m \to \infty$ the system becomes unstable and throughput is zero. In comparison, S_{EB} for a given parameter q converges with a specific positive value, even for very large values m. For $q = 0.5$ and with $m = 1000$ stations the result is $S_{EB} \simeq 0.311$; this corresponds to about 85 % of the maximum achievable throughput e^{-1} and in normal operations equates to a satisfactory result. The collision resolution method presented in [14] has therefore been proved to be stable.

2.8.4.2 The Pseudo-Bayesian Algorithm

This algorithm (see [28]) is based on a method that establishes the estimated value of a backlog in order to stabilize the Slotted-ALOHA access channel.

In contrast to the versions of Slotted-ALOHA access explored previously, with this method newly arriving packets are also sent with the probability p_r. Therefore the backlog n here is a combination of packets that have already collided and new ones. The access rate is consequently $G(n) = np_r$ and the probability of an access being successful is $np_r(1 - p_r)^{n-1}$. However, this modification has no major effect on access delay if the parameter p_r is around 1 when traffic volume is low.

The task of the algorithm is to determine an estimated value $\hat{n}$ for the number of collided stations n at the beginning of each slot. Each packet will then be transmitted with probability

$$p_r(\hat{n}) = \min\{1, 1/\hat{n}\} \tag{2.100}$$

The min operation establishes an upper limit for the transmission probability and causes the access rate $G = np_r$ to become 1. For a given slot k a new estimated value for n is determined for the next slot $k + 1$ based on the following rules.

If an empty or a collided slot is received, the estimated value for the backlog is calculated as

$$\hat{n}_{k+1} = \max\{\lambda, \hat{n}_k + \lambda - 1\} \tag{2.101}$$

with λ standing for the arrival rate normalized to the slot duration. The addition of λ to the previous backlog value applies to newly generated packets, whereas the max operation ensures that the estimated value is never lower than the contribution of the newly arrived packets. A successful transmission is considered by subtracting 1 from the previous estimated value of the backlog. A 1 is also subtracted for each empty slot, thereby causing a reduction to $\hat{n}$ if there are too many empty slots.

If a collision is detected, the new estimated value of the backlog is calculated as

$$\hat{n}_{k+1} = \hat{n}_k + \lambda + (e - 2)^{-1} \tag{2.102}$$

Here again λ is added to the previous backlog value to take into account the newly arrived packets. The addition of $(e - 2)^{-1}$ effects an increase to $\hat{n}$ if the number of collisions is too high. For high backlog values and an appropriate accurate estimated value for the backlog, the access rate $G(n)$ therefore becomes 1. If this rate is approximated using the Poisson distribution then empty slots occur with the probability $1/e$ and collisions with the probability $1 - 2/e$. Consequently the anticipated change $\delta(\hat{n})$ for the estimated value $\hat{n}$ because of empty and collided slots can be stated as

$$\delta(\hat{n}) = \frac{-1}{e} + \left(\frac{e-2}{e}\right)\left(\frac{1}{e-2}\right) = 0 \tag{2.103}$$

If this algorithm is used in practice, there is the problem that λ will not be known at the outset and furthermore that it varies. Therefore the method must either estimate the rate λ from the average values of successfully transmitted packets (this will be covered in more detail later) or the algorithm will determine a fixed value. It has been proved (in [5] and elsewhere) that the system remains stable at $\lambda < 1/e$.

The following approximation is given in [28] to estimate the arrival rate λ:

$$\hat{\lambda}_{k+1} = 0.995\hat{\lambda}_k + 0.005I_{z(k)=1} \tag{2.104}$$

2.8.4.3 Stabilization through MMSE Estimation

Another approach for stabilizing Slotted-ALOHA and reducing access delay times is offered by the MMSE (*minimum mean-squared error*) method, which estimates the number of collided stations. With this method transmission probability is determined dynamically on the basis of the estimated collision rate in order to maximize throughput. The technique presented in [35] is based on the same assumption as the one in the Slotted-ALOHA model that newly generated packets will be sent immediately in the next available slot. This is therefore the IFT mode of Slotted-ALOHA.

The control algorithm presented determines the transmission probability

$$p_r(k) = \min\left(\beta\ ,\ \frac{\alpha}{\hat{n}_k}\right) \tag{2.105}$$

with $\hat{n}_k$ as the MMSE estimation of the number of collided stations at the beginning of the kth time slot. The parameter β is to be selected so that $0 \leq p_r(k) \leq 1$. The study of the method in [35] was based on $\beta = 0.5$. The parameter α, which is supposed to minimize access delay times, will be covered in detail later. First the asymptotic MMSE estimation of the number of collided stations in the $(k+1)$th slot is given:

$$\hat{n}_{k+1} = \hat{n}_k + \begin{cases} -\alpha/(\lambda+\alpha), & \text{if } I_{z(k)=1} \\ 0, & \text{if } I_{z(k)=0} \\ \lambda/[1-(\lambda+\alpha)/(e^{\lambda+\alpha}-1)], & \text{if } I_{z(k)\geq 2} \end{cases} \tag{2.106}$$

and

$$\hat{n}_{k+1} \leftarrow \max(0\ ,\ \hat{n}_{k+1}) \tag{2.107}$$

I_A is the state function again (see Equation (2.97)). Since $n_t \geq 0$ in a real system, Equation (2.107) guarantees that the estimated value $\hat{n}_t \geq 0$ will be maintained.

The access method is stable for arrival rates that satisfy the condition

$$\lambda \leq (\lambda+\alpha)e^{-(\lambda+\alpha)} \tag{2.108}$$

The maximum throughput (on the basis of a theorem mentioned in [35]) is achieved for this α, which allows both sides in Equation (2.108) to have about the same value. This applies to

$$\alpha = 1-\lambda \tag{2.109}$$

which is synonymous with

$$\lambda_{max} = e^{-1} \tag{2.110}$$

The estimation algorithm is thereby simplified to

$$\hat{n}_{k+1} = \hat{n}_k + \begin{cases} -(1-\lambda), & \text{if } I_{z(k)=1} \\ 0, & \text{if } I_{z(k)=0} \\ 2.392\lambda, & \text{if } I_{z(k)\geq 2} \end{cases} \tag{2.111}$$

This recursive formula assumes knowledge of λ. The following algorithm for determining this arrival rate appears in [35]:

$$\hat{\lambda}_{k+1} = \frac{1}{k+1}\left[\hat{n}(k+1 \mid k,\, \hat{\lambda}_k) + \sum_{j=0}^{k} \epsilon_j + \lambda_0\right] \tag{2.112}$$

with

$$\epsilon_j = \begin{cases} 1, & \text{if } I_{z(k)=1} \\ 0, & \text{otherwise} \end{cases} \tag{2.113}$$

in which $\hat{n}(k+1 \mid k,\, \hat{\lambda}_k)$ represents the MMSE estimation at the time of the $(k+1)$th slot and takes into consideration all slots up to the kth slot and the rate $\hat{\lambda}_k$ as the value of the actual arrival data rate. The constant $\lambda_0 \in (0,\, e^{-1})$ is required in order to initialize the estimated algorithm. The parameter α given in Equation (2.106) can be adapted using $\hat{\lambda}_k$.

2.8.4.4 Stabilization Using Stochastic Approximation

Another technique used to optimize throughput and to keep it approximately constant is suggested in [11]. This algorithm does not take into account the number of collided stations, but instead presents a method, based on a traffic theoretic analysis, that each time produces a new transmission probability $p_r(k)$ at the point of the kth slot according to feedback on channel state. The technique determines $p_r(k)$ so that the throughput, which according to Equation (2.60) can be given as $S_{out}(k) \approx G(k)e^{-G(k)}$ and is not dependent on the momentary value of n, is always close to its optimal value; thus $G(k) = 1$.

The method is based on the transmission result in the last slot, and determines the new transmission probability depending on whether there is an empty slot, a successful transmission attempt or a collision:

$$p_r(k+1) = \min\left\{p_{max}\ ,\ p_r(k)\left(a_0 I_{z(k)=0} + a_1 I_{z(k)=1} + a_c I_{z(k)\geq 2}\right)^{\gamma}\right\} \tag{2.114}$$

with

$$(a_0, a_1, a_c) = \left(e^{\frac{1-2e^{-1}}{1-e^{-1}}}, 1, e^{\frac{-e^{-1}}{1-e^{-1}}}\right) \tag{2.115}$$

A concrete value for p_{max} is not given in [11]. However, an analysis is presented showing that when two stations are competing for a channel, throughput will be maximized if

$$p_{max} = \frac{1-\lambda}{2-\lambda} \tag{2.116}$$

p_{max} is dependent on the arrival rate, which has to be estimated and in some circumstances can vary. Therefore [9] gives a value for p_{max} that is independent of this rate. It uses the highest possible packet rate that keeps the system stable ($\lambda = 1/e$) in Equation (2.116) and therefore produces $p_{max} \approx 0.38$. For the parameters in Equation (2.115), $(a_0, a_1, a_c) \approx (1.52, 1, 0.56)$ applies. The behaviour of a system with the values of the parameters γ lying

between 0.1 and 4 is explored in [11]. It emerges that the average throughput of a system decreases with γ. The average number of collided stations is relatively high, however, with values of γ in the area $\gamma \leq 0.1$ and an arrival rate of $\lambda = 0.32$. In [9] $\gamma = 0.3$ is regarded as optimal.

2.8.4.5 The CONTEST Algorithm

Different methods, including the RCP (*retransmission control procedure*), are suggested in [16] for stabilizing the Slotted-ALOHA protocol. With this procedure transmission probability is determined as p_o or p_c, depending on traffic volume. The value p_o is selected so that throughput is optimized and p_c is low enough to stabilize the system.

The CONTEST (*Control-Estimation*) algorithm is based on this procedure. The decision whether to use transmission probability p_o or p_c depends on whether the number of collided stations n exceeds a particular limit value $\hat{n}$. With this value of n the traffic rate of the access channel is

$$\hat{G}_o = \hat{n}p_o + (m-n)p_a \quad (2.117)$$

$$\hat{G}_c = \hat{n}p_c + (m-n)p_a \quad (2.118)$$

with m as the total number of stations and p_a as the probability of new packets being sent. On the assumption that the traffic volume on the access channel is Poisson-distributed, the following critical values can be given because of the probability of an empty slot: $\hat{f}_o = e^{-\hat{G}_o}$ and $\hat{f}_c = e^{-\hat{G}_c}$. Furthermore, it is also assumed that the transmission result in earlier slots is observed over an interval of length W slots. If $\bar{f}$ is also the portion of empty slots in interval W up to time slot k then $\bar{f}$ approximates the probability of an empty slot in time slot $k+1$, on the assumption that there will be little change to the traffic rate during the period of time being considered. The following algorithm is given, in which $p(k)$ is the transmission probability to be determined for the new slot.

1. Step:
   - $k \leftarrow k+1$
   - $p(k) = p_o$
2. Step:
   - if $\bar{f} < \hat{f}_o$ GOTO step 4.
3. Step:
   - GOTO step 1.
4. Step:
   - $k \leftarrow k+1$
   - $p(k) = p_c$
5. Step:
   - if $\bar{f} < \hat{f}_o$ GOTO step 1.
6. Step:
   - GOTO step 4.

For p_o either a low transmission probability or the optimal one dependent on n and λ, derived in the next section, can be used. For p_c the value $p_c = K \cdot n$ with $K \geq 1$ is suitable for the satisfactory stabilization of a system.

Therefore the above algorithm can be expanded in all sorts of different ways. Similarly to a high-traffic-load situation, observation of the access channel can produce a limit for determining a low-traffic-load situation and provide an appropriate procedure for adapting the access parameters.

2.8.4.6 Adaptive Determination of Optimal Transmission Probability

The optimal frame length derived from Equation (2.79) corresponds to the optimal transmission probability $p_r = (1-\lambda)/(n-\lambda)$ stated in [7] and [23].

In [23] the mean value $\bar{n}$ is determined from the sequence of empty slots. It is assumed thereby that n as well as the arrival rate λ remain almost constant during an observed period of time of length x slots. The probability of an empty slot is then the product of the probability that new stations are not transmitting in slots being considered and the probability that none of the stations which has already collided is transmitting, and can be indicated as

$$P_0 \approx (1-p_r)^{\bar{n}} e^{-\lambda} \tag{2.119}$$

Furthermore, if an interval of x slots is given in which each one is empty with a probability P_0, and if e_0 is the number of empty slots in the period being observed, then e_0 is binomially distributed with mean value $E_0 = xP_0$. Then, from Equation (2.119), n becomes

$$n \approx \frac{\ln(E_0/x) + \lambda}{\ln(1-p_r)} \tag{2.120}$$

Reference [7] presents a different algorithm for deducing the value of n as well as the optimal transmission probability for collided stations p_r. Based on this algorithm, p_r is determined depending on n according to

$$p_r(\hat{n}) = \begin{cases} (1-\lambda)/(\hat{n}-\lambda), & \text{if} \quad \hat{n} \geq 1 \\ 1, & \text{if} \quad 0 \leq \hat{n} < 1 \end{cases} \tag{2.121}$$

The new number of collided stations in each slot is determined using the following estimation algorithm based on the feedback channel state information:

$$\hat{n}_{k+1} = \hat{n} + U(f_k, \hat{n}_k) \tag{2.122}$$

with

$$U(f_k, \hat{n}) = \max(\hat{n}_{min} - \hat{n}, u_f), \qquad \text{with } \hat{n} \geq \hat{n}_{min} \text{ and } f = 0, 1, c \tag{2.123}$$

In the algorithm, $\hat{n}_{min}$ represents the lower limit; u_0, u_1 and u_c are the real constants. Reference [7] gives the parameters $u_0 = 2 - e \approx -0.7183$, $u_1 = 0$, $u_c = 1$ and $\hat{n}_{min} = 2$ for this algorithm when the arrival rate is $\lambda = 0.32$. A simulation experiment produces the lowest average access delay when $\lambda = 0.32$ with the parameters $(u_0, u_1, u_c) = (-0.4, -0.4, 0.9)$.

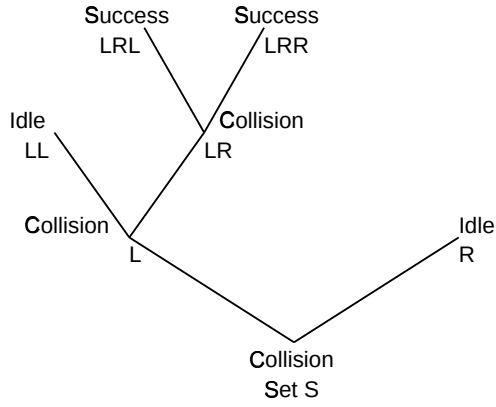


Figure 2.63: Example of a splitting algorithm

2.8.4.7 The Splitting Algorithm

The splitting algorithm is another method for resolving collisions. This entails assigning one of two possible sets to each station that wants to transmit using random access. All stations that are not participating in a current collision resolution are in a *waiting set*. If a new transmission request originates in a station, it takes its place in the waiting set. All the mobile stations that are participating in the current collision resolution phase are found in the *collision set*. At the beginning of a collision resolution phase, part of the waiting set is allocated to the previous empty collision set and therefore participates in the collision resolution. No other stations can be allocated subsequently to the collision set. Stations that have transmitted successfully are dropped from the collision set. This process is not completed until the collision set has been completely emptied. Then a new collision resolution phase begins with a new collision set.

The splitting algorithm is explained by means of an example in Figure 2.63. After a collision occurs, all mobile stations from set S that were involved in the collision are assigned to the collision set. With a binary splitting algorithm the mobile stations are assigned to the collision set of one of the two new subsets (sets L and R). Different criteria apply to the assignment and these will be discussed later. In the event that a collision occurs again in one of the two subsets, it will again be split. In Figure 2.63 a collision has occurred in set L, thereby resulting in subsets LL and LR. This algorithm is continued until all collisions have been resolved. The time between the start of a collision and the actual resolution of the collision is referred to as the collision resolution period. The stations finding themselves in the waiting set wait for the start of a new collision resolution period.

If stations are informed immediately of the status of their access attempt, this information can also be used in resolving the collision. These two im-

provements to the splitting algorithm help to increase throughput. These are introduced here.

There are different ways in which mobile stations can be allocated to the subsets:

- They are distributed evenly among the subsets.
- The distribution is according to arrival time.
- The mobile station calculates a priority for itself and uses it to determine the subset.
- Parts of the mobile stations's identification number are used in the allocation.

Optimizing the splitting algorithm If information about the success of the collision resolution of the other sets is available, there are two possibilities for optimizing the splitting algorithm to increase throughput.

In Figure 2.63 a collision has occurred in subset L and no mobile station has transmitted in subset LL (`Idle`). Therefore a collision would inevitably occur with subset LR, so this subset can immediately be split into LRL and LRR. After a collision has occurred in subset L, i.e., at least two mobile stations have allocated themselves to this subset, the probability that a third station will find itself in subset R is small. It is therefore more advantageous to assign subset R to the waiting set again. These two improvements will increase throughput to 43 % (without optimization) and to 48.7 % (with optimization) instead of the maximum throughput of 36.8 % without splitting.

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3
GSM System*

3.1 The GSM Recommendation

The early 1980s were marked by the development of a number of national and incompatible radio networks in Europe; see Table 1.2 and Figure 1.3. The seven different mobile radio networks made the prospect of the mobile telephone unattractive to many potential customers because of high tariffs and equipment costs.

For this reason, at its general meeting in Vienna in June 1982, CEPT (see Appendix B.2.2) decided to develop and standardize a Pan-European cellular mobile radio network. The aim was for the new system to operate in the 900 MHz frequency band allocated to land mobile radio.

A working group, called *Group Spécial Mobile* (GSM), was set up under the direction of CEPT. There were no guidelines on how the new mobile radio system was to transmit analogue or digital speech and data. The decision to develop a digital GSM network was not made until the development stage. But it was agreed from the beginning that the system being planned—called the GSM mobile radio system after the working group that developed it—should incorporate and consider new technology from the area of telecommunications, such as ITU-T Signalling System No. 7, ISDN and the ISO/OSI reference model.

Six working groups and three supporting groups were formed to cope with the enormity of the standardization work. The tasks of the different GSM working groups are listed in Table 3.1.

The GSM objectives for its *Public Land Mobile Network* (PLMN) were to offer [1]:

- A broad offering of speech and data services
- Compatibility with the wireline networks (ISDN, telephone networks, data networks) using standardized interfaces
- Cross-border system access for all mobile phone users
- Automatic roaming and handover
- Highly efficient use of frequency spectrum

*With the collaboration of Peter Decker and Christian Wietfeld

Table 3.1: Tasks of the GSM working groups

GSM working groups	Tasks
Working Party 1	Definition of services and service quality
Working Party 2	Definition of access, modulation and coding procedures
Working Party 3	Definition of protocols for signalling between mobile stations, mobile functions and fixed communications networks
Working Party 4	Specification of data services
Working Party 5	Development of UMTS
Working Party 6	Specification of network management features
Speech Coder Experts Group (SCEG)	Definition of technique for digitization of speech at a low bit rate
Security Experts Group (SEG)	Responsibility for all aspects of security (access, coding, authentication)
Satellite Earth Systems (SES)	Support of GSM through satellite systems

Table 3.2: Original timetable for introducing the GSM system

Date	Phase
February 1987	Invitation for tenders
Mid 1988	Letters of Intent
End 1988	Validation of interfaces
Mid 1990	System validation
March 1991	Start of equipment deliveries
June 1991	Operation of first base station
1993	Coverage to metropolitan areas and major roads
since 1995	Area-wide operation

- Support of different types of mobile terminal equipment (e.g., car, portable and hand-held telephones)
- Digital transmission of signalling as well as of user information
- Supplier-independence
- Low costs for infrastructure and terminal equipment

The GSM group tested a number of prototypes for digital cellular radio systems, and in 1987 decided on a standard that combined the best characteristics of different systems. A timetable drawn up at the same time for the implementation of the plan gained the full support of the European Union (EU) (see Table 3.2).

Table 3.3: The series of the GSM recommendation

Series	Content
00	Preamble
01	General aspects, terminology and service introduction phases of the GSM *Public Land Mobile Network* (PLMN)
02	Definition of telecommunications services, technical aspects concerning tariffs and international billing procedures
03	Definition of network functions such as traffic routing, handover, security issues relating to network access, network planning
04	Description and definition of protocols and interfaces between mobile station (MS) and base station (BS)
05	Radio path functions such as multiplexing, channel coding, synchronization and interleaving
06	Speech processing and speech coding functions
07	Adaptation of terminal equipment and transmission rates
08	Description of interface functions between *base station system* (BSS) and *mobile services switching centre* (MSC)
09	Definition of *interworking functions* (IWF) between one or more GSM networks and different fixed networks
11	Equipment specifications and type approval guidelines
12	Operation and maintenance of a GSM network

By 1987, comprehensive guidelines for the new digital mobile radio system had already been established by the GSM group. By signing the *Memorandum of Understanding on the Introduction of the Pan-European Digital Mobile Communication Service* (MoU) on 7 September 1987, the 13 participating countries confirmed their commitment to introducing mobile radio based on the recommendations of the GSM.

Later, in March 1989, the GSM working party was taken over by ETSI (see Appendix B.2.3), and since 1991 has been called the *Special Mobile Group* (SMG). Today the abbreviation GSM stands for *Global System for Mobile Communications*, thereby underlining its claim as a worldwide standard.

In the meantime all the European countries as well as a large number of other countries in the world have signed the GSM-MoU agreement and have developed or will be developing mobile radio systems in their countries based on the GSM recommendations (see Table 3.39).

The planned official start to the GSM system was delayed by one year. Only five countries were in a position to undertake test operations on 1st July 1991. The reason for the delay was the level of complexity of the digital network and its components, which is reflected in the voluminous specifications which today total around 8000 pages. In 1990 alone another 500 GSM *change requests* were passed. The entire set of GSM recommendations is divided into 13 series, which cover different aspects of the GSM system, as shown in Table 3.3 (see also Appendix E).

The GSM recommendations contain detailed specifications for the radio interface which in part are borrowed from the concepts for the analogue national cellular standard and ITU-T Rec. X.25. However, large parts of the radio interface are specific to the GSM system. Some of the important features of GSM include:

Frequency band The frequency range between 935 and 960 MHz is used as the base station transmitting frequency (*downlink*) and the frequencies between 890 and 915 MHz are used as the base station receiving frequency (*uplink*). The carrier frequencies of the FDM radio channels have 200 kHz channel spacing in each band, thus providing 124 FDM channels. With time-division multiplexing (TDM), eight communications channels (time slots) are supported per FDM channel.

Handover Handover from one base station to another is a mechanism that allows the connection quality of calls between users to be maintained, interference to be minimized and traffic distribution to be controlled. In addition, procedures are defined for the re-establishment of a connection if a handover fails.

Power control In the area over 30 dB the equipment of the mobile user and of the base station controls power in 2 dB steps in order to minimize interference.

Discontinuous transmission (DTX) GSM offers the option of discontinuous transmission of speech using voice activity detectors. With DTX, transmitter battery power is only used when speech or data is being transmitted, which minimizes interference and improves the utilization of frequency spectrum.

Synchronization Depending on the system, all frequencies and times are synchronized with a highly stable (0.005 ppm) reference, which can be coupled with a frequency normal.

The following features distinquish GSM from other European mobile radio systems:

- Europe-wide coverage
- Europe-wide standardization
- digital radio transmission
- extensive ISDN compatibility
- protection against eavesdropping
- support of data services

GSM is regarded as an important advance compared with predecessor systems and is considered to be representative of so-called 2nd-generation systems. Along with important technological advances (particularly the introduction of digital transmission technology), the standardization of the interfaces between subsystems in GSM has provided manufacturers and network operators flexibility in their development work and configurations.

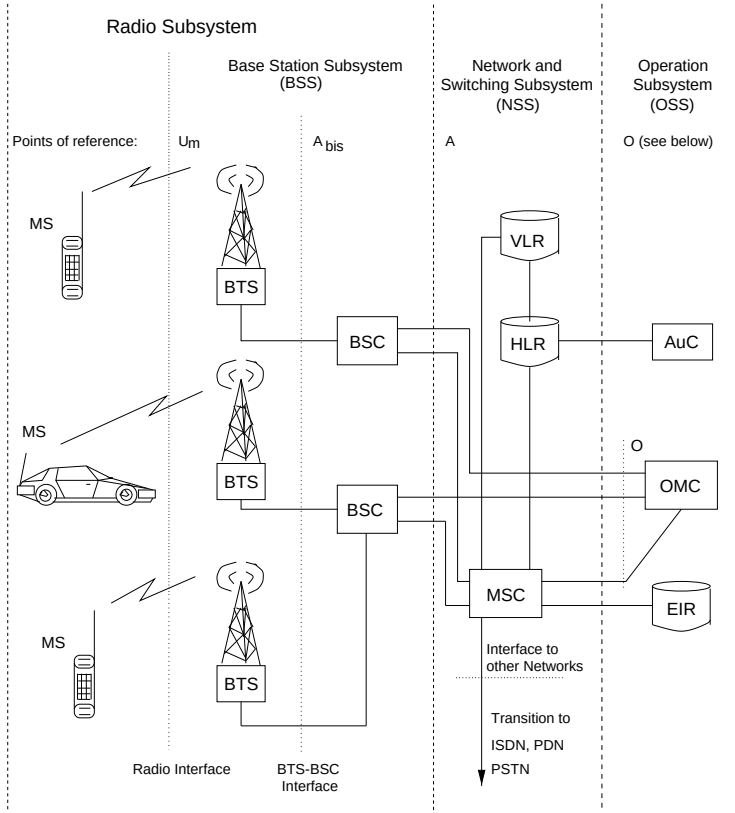


Figure 3.1: Functional architecture of the GSM mobile radio network

3.2 The Architecture of the GSM System

3.2.1 Functional Structure of the GSM System

In GSM specification 1.02 the GSM system is divided into the following subsystems [20]:

- *Radio subsystem* (RSS)
- *Network and switching subsystem* (NSS) and
- *Operation subsystem* (OSS)

These subsystems and their components are represented in the simplified version of the functional architecture in Figure 3.1.

3.2.1.1 Radio Subsystem

The *radio subsystem* is made up of the *mobile stations* (MS) and the *Base station subsystem* (BSS).

Mobile station The term mobile radio station (MS) refers to all the physical equipment of a PLMN user. It includes the mobile terminal and the user interface that the subscriber needs in order to access PLMN services.

A GSM mobile station consists of two parts. The first part contains all the hardware and software components relating to the radio interface; the second part, known as the *subscriber identity module* (SIM), stores all the subscriber's personal data. The SIM is either installed into the terminal or provided as a *smart card*, which is about the size of a credit card and has the function of a key. Once it has been removed from a device, it can only be used for emergency calls, if the network so allows. A mobile subscriber can use the SIM to identify himself over any mobile station in the network, and accordingly a mobile phone can be personalized using the SIM. In addition, each mobile station has its mobile *equipment identity* (EI).

The following numbers and identities are assigned for the administration of each mobile station within a GSM network; see Figure 3.57:

- *International Mobile Subscriber Identity* (IMSI)
- *Temporary Mobile Subscriber Identity* (TMSI)
- *Mobile Station International ISDN Number* (MSISDN)
- *Mobile Station Roaming Number* (MSRN)

Mobile stations can be installed in automobiles or provided as portable/hand-portable devices and, according to GSM Rec. 2.06, are divided into five different classes depending on the allowable transmitter power; see Table 3.4.

These classifications also characterize different types of devices: mounted, portable and hand-portable devices. Equipment for the GSM-900 class 1 (8–20 W) has not yet been developed. Instead, portable and mounted equipment is typically found in class 2 (5–8 W). Hand-portable equipment mostly conforms with class 4 (0.8–2 W). Class 5 (up to 0.8 W) is also being planned for hand-portable equipment, but places a considerable strain on cellular radio signal supply. This is one of the reasons why it is more suitable for urban environments with small cells, but it is hardly being used anywhere yet. An MS can have facilities for both voice as well as data transmission.

In addition to the network-dependent radio and protocol functions that enable access to operation in the network, a mobile station outwardly has at least one other *interface* to the mobile subscriber (see Section 3.2.2). It is intended either for a human user (man–machine interface) or for coupling the *terminal adapter* of another terminal, such as a computer or a fax machine or

Table 3.4: Power classes of mobile stations according to GSM or DCS 1800

Class	GSM 900		DCS 1800	
	Max. transmit. power [W]	Type of device	Max. transmit. power [W]	Type of device
1	20	Mounted and portable	1	Hand-portable
2	8	Portable and mounted	0.25	Hand-portable
3	5	Hand-portable	–	–
4	2	Hand-portable	–	–
5	0.8	Hand-portable	–	–

a combination of the two. The GSM specifications leave the conversion and extent of the interface technique up to the manufacturer.

A user interface usually consists of the following components:

- microphone
- speaker
- LCD display field
- alphanumeric keyboard
- so-called soft keys

Soft keys are function keys used to switch a terminal to different operating states. They are not assigned a specific function, as is the case with hard keys, e.g., on a drinks dispenser. Consequently the user must be informed of the respective function before using the keys.

Soft keys are extremely useful with hand-held mobile phones. The subscriber can use his mobile device with one hand because of the soft key menu functions that are displayed on the mobile, without having to press key combinations at the same time, as is required with the hard key version of a control panel.

Unlike the conventional telephone, where the user is identified through the fixed network connection, radio connections form an anonymous network. Therefore subscriber identification is a prerequisite in a mobile radio network alone for operational reasons. The stored subscriber-related data in a SIM module identifies the subscriber when he checks in, and his location area is derived from the serving base station—an automatic procedure when the terminal is used.

In older devices the SIM is installed into the equipment, but the new approach is to plug it in as a card; there are two versions of this:

- smart card, also called standard SIM card
- plug-in SIM card

The only difference between the two cards is their size. The standard SIM card is the size of a credit card based on standard ISO 7816, whereas the plug-in module is smaller in size and based on the GSM Rec. 02.17 [6]. In addition to their size, the cards are also used differently. Whereas the standard SIM card can be activitated simply by being inserted into the card slot provided in the mobile telephone, the smaller module slides into the equipment mounted on a cut-down card, which involves first removing the battery. The smaller plug-in SIM card has been successful with hand-held mobile telephones.

The subscriber-related data is stored in the non-volatile memory of the SIM. It can be changed statistically as well as temporarily. The permanent data includes the following elements [6]:

- SIM card type
- IC card identification: serial number of the SIM; identifies card holder at the same time
- SIM *service table*: list of additional services subscribed
- IMSI (*International mobile subscriber identity*)
- PIN (*Personal identity number*)
- PUK (*PIN unblocking key*)
- Authentification key K_i

Before a SIM card is assigned to a subscriber, it is first initialized with this data, and only then can the subscriber use the card to check into the network. On the other hand, the dynamic data, which is permanently updated when the terminal is switched on, accelerates the checking-in process because relevant information is already stored centrally and there is no need for it to be requested from the network. This includes the following data items [6]:

- Location information: consists of a `TMSI`, a LAI, a periodically changed location updating timer, and update status
- Ciphering key K_c for encoding, and its sequence number
- BCCH information: list of carrier frequencies for cell selection during handover and call setup
- List of blocked PLMNs
- HPLMN search: period of time in which an MS roams the home network before it tries to check into another network

Other optional data items can be found in [6]. All SIM data is copied in the memory of the MS only for the duration of the active operating state and then deleted. Manufacturers of mobile terminals have the option of additionally providing intermediate storage of less important data, such as short messages and the last-called telephone number. However, this data can only be called up if the equipment is turned on again with the same SIM card that was used for its previous deactivation [6].

PIN Except for emergency calls, mobile equipment can only be operated if the SIM card has first been activated. This is done by the subscriber punching in a `PIN` code, which can be between four and eight digits long, after switching on the equipment. When the SIM card is provided by the service provider, the `PIN` is generally preset with a four-digit number, which the subscriber can change as often as he likes. After the `PIN` has been correctly entered, the network responds and the mobile is automatically checked in.

A `PIN` can be but should not be disabled, because the subscriber will run the risk of potential thieves using the mobile free of charge until use of the card is suspended. Anyone who steals an activated mobile phone can only use the SIM card fraudulently until the first time the equipment is switched off or the battery runs out. If an incorrect `PIN` is inserted three times in a row, the card will be suspended. The subscriber then needs an unlocking key `PUK`.

Some cards are available with a second `PIN` to protect some of the numbers stored in the card. This specifically protects personal telephone numbers and names entered on the card from unauthorized access. The security mechanisms and maximum allowable code length of the `PIN2` are identical to those of the `PIN` [6].

PUK A blocked SIM card can only be released through the use of an *`PIN` unblocking key* PUK. The subscriber is allowed 10 attempts in which to enter the correct `PUK` code or else the card will be blocked permanently and can only be unblocked by the service provider. The `PUK` is an eight-digit permanent number that is divulged to the subscriber when he receives the card [6].

3.2.1.2 Base Station Subsystem (BSS)

The BSS comprises all the radio-related functions of the GSM network. Depending on the radio transmitting and receiving capabilities of the *base transceiver system*, which because of limited transmitter power only supplies coverage to a specific geographical area within the network, radio cells are created in which the mobile subscriber is free to roam or communicate. The size of the individual cells depends on a number of parameters, including characteristics of radio wave propagation, local morphology, and expected user density in the region.

A BSS uses transceivers and the following hardware and software to enable it to connect a mobile subscriber to a number in the public telephone network (PSTN) and allow it to communicate:

- signalling protocols for connection control
- speech codecs (coders/decoders) as well as data-rate adaptation (*transcoder/rate adapter unit*, TRAU) for access to the network
- digital signal transmission for coded data.

These functions already give an indication of some of the other important tasks of the BSS. Various interfaces have been specified between the BSS and GSM network elements and other networks for the exchange of information between subscribers and the GSM network or other networks; see Figure 3.1. The interface to the mobile subscriber is called the U_m-interface. It contains specific parameters for digital radio transmission, such as GMSK modulation, data rate, status of carrier frequencies in the 900 MHz band and channel grid. The BSS is connected to the GSM fixed network over the A-interface (familiar from ISDN) with MSCs, the NSS switching centres that provide the subscriber connectivity to each other and to the external network. The A-interface likewise contains specific digital transmission parameters, including PCM (*pulse code modulation*), a 64 kbit/s data rate and a 4 kHz voice bandwidth.

Network availability and quality is established by the network operations and maintenance centre (OMC) of the GSM operator over an O-interface, which provides direct access to BSS units.

The elements making up the BSS include:

- *Base transceiver station* (BTS)
- *Base station controller* (BSC)

BTS The BTS comprises the transmitting and receiving facilities, including antennas and all the signalling related to the radio interface. Depending on the type of antenna used, the BTS supplies one or several cells, so, for example, sectorized antennas can supply three cells arranged at 120° to each other (see Chapter 2.4).

In a standardized GSM structure the *transcoding and rate adaptation unit* TRAU is part of the BTS. It contains GSM-specific speech coding and decoding as well as rate adaptation for data transmission.

BSC The BSC is responsible for the management of the radio interface through the BTS, namely for the reservation and the release of radio channels as well as handover management. Its other tasks include *paging* and transmitting connection-related signalling data adapted to the A-interface from/to the MSC.

A BSC generally manages several BTSs, and is linked to the NSS via an MSC.

3.2.1.3 Network and Switching Subsystem (NSS)

Switching and network-oriented functions are carried out in a *Network and switching subsystem* (NSS). It forms the gateway network between the radio network and the public partner networks (e.g., *Public Switched Telephone Network* (PSTN), *Integrated Services Digital Network* (ISDN), *Public Switched Data Network* (PSDN)). In their entirety not only are the elements of an NSS purely physical components but, more importantly, the switching subsystem provides a large number of functions that are the responsiblity of the manufacturer and network operator to implement appropriately.

The NSS components include the *Mobile Services Switching Centre* (MSC), the *Home Location Register* (HLR) and the *Visitor Location Register* (VLR).

Mobile services switching centre (MSC) The MSC is a high-performance digital switching centre that carries out normal switching tasks and manages the network. Each MSC is usually allocated several base station controllers, and in the geographical area assigned to it carries out the switching between mobile radio users and other PLMNs and also forms the link between the mobile radio network and the wireline networks (PSTN, ISDN, PDN). The MSC is responsible for all the signalling required for setting up, terminating and maintaining connections, carried out in accordance with *Common Channel Signalling System No. 7*, and mobile radio functions such as *call rerouting* when there is strong interference, as part of a *handover* and the allocation and deallocation of radio channels.

Transmission functions for data services are supported through the use of specific *interworking functions* (IWF) that are integrated into each MSC. The respective communications channel functions are carried out by facilities called *data service units* (DSU). The DSU contains functions such as rate adaptation, modem and codec of layer 1, and protocol functions of layer 2.

The other tasks of the MSC include the supplementary services familiar from ISDN, such as call forwarding, call barring, conference calling and call charging to the user called. The MSC can be envisaged as an ISDN switching centre that has been expanded to include the necessary mobility-related switching functions.

Home location register (HLR) All important information (quasi-permanent static data) relating to each mobile subscriber, including telephone number, MS identification number, equipment type, subscription basis and supplementary services, access priorities and authentication key, is stored in the database referred to as the home location register. Temporary (dynamic) subscriber data (e.g., current *location area* (LA) of the mobile station and *mobile station roaming number* (MSRN)) that are necessary for setting up a connection are also stored. When a mobile user leaves his momentary location area (LA), the temporary data held in the HLR is immediately updated. The home location register usually falls under the responsibility of a mobile switching centre

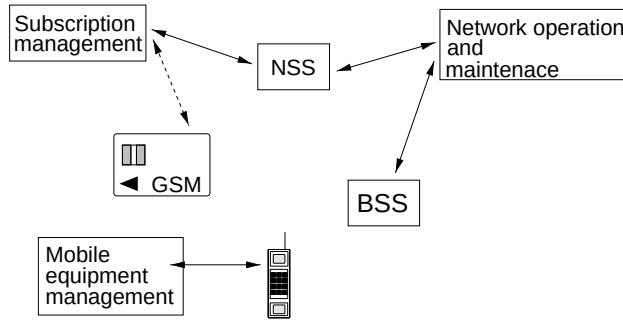


Figure 3.2: Structure of an OSS

(MSC). Each mobile subscriber and his related data are registered in only one home location register in which all the billing and administrative tasks are carried out. In many existent GSM networks there is only one HLR being implemented.

Visitor location register (VLR) The visitor location register is under the control of an MSC and is used to manage the subscribers who are currently roaming in the area under the control of the MSC or, more precisely, in one of possibly several location areas of the MSC. It stores information (e.g., authentication data, international mobile subscriber identity (IMSI), telephone number, agreed services) transmitted by the responsible HLR for the mobile users operating in the area under its control, thereby allowing the MSC to make a connection. The VLR also controls the allocation of roaming numbers (MSRN) to the mobile stations as well as of the TMSI. A special dialogue updates the VLR if a mobile user moves through several of the MSC's location areas. The same procedure applies when there is a change of MSC. The VLR avoids frequent interrogation of the HLR.

The functions *location area update* and *call setup* and the roles played by the HLR and the VLR in these functions are described in Sections 3.7 and 3.8.

3.2.1.4 Operation Subsystem (OSS)

The operation subsystem in GSM comprises all the important functions for operation and maintenance. The user is only indirectly aware of these functions through his experience with a smoothly functioning mobile radio network.

The functions of an OSS are allocated to three areas of responsibility (see Figure 3.2):

- Subscription management
- Network operation and maintenance
- Mobile equipment management

The following network elements are part of the OSS:

- Operation and maintenance centre (OMC)
- Authentication centre (AuC)
- Equipment identity register (EIR)

Subscription management Subscription management is able to authenticate a GSM user from the personal data stored in the HLR (see Section 3.13.1) and provide him with the agreed services (*subscriber data management*). This data provides the network operator and the service provider with a call-charging basis.

Subscriber data management The subscriber data is stored and managed in the HLR; information relating to data security is in the AuC. The HLR can provide restricted access to elements from other networks, e.g., in order to allow service providers access to tariff and services data and to ensure the consistency of data stored in different locations. As has already been mentioned, the SIM card is a dynamically changeable data storage unit during the active operation of a mobile station.

Call charging Similarly to ISDN, the mobile radio user is charged for services used on the basis of so-called *call tickets*. These call tickets are used for billing irrespective of where a call is made in the network. The billing location can be the MSC in which the mobile subscriber is currently active or a gateway MSC (GMSC) where a communication is connected to an external network.

The HLR only stores call-related data. Call billing is handled by the responsible OSS subscriber management. At the same time tariff data is also transmitted between the MSCs or GMSCs and the HLR over the *common channel signalling system no. 7 (SS 7)*.

Network operation and maintenance The control of network operation and maintenance tasks uses a separate switching network to connect operating personnel network elements. The network is based on the concept of TMN (*Telecommunications Management Network*) developed by the ITU-T. The TMN forms an integrated network with its own databases that offer the operator options for monitoring, control and intervention.

The TMN functions are divided into individual layers similar to the network element functions in the ISO/OSI reference model:

Business management Controls the interaction between network and services and provides information about other service and network developments.

Service management Used for the execution of all contractual aspects of a service between supplier and customer.

Network management Supports all network elements and helps to activate functions with similar elements of a network.

Network element management Facilitates access to individual network elements.

GSM uses standardized concepts for network management, thereby facilitating the integration of the network elements of different suppliers.

The TMN has links with defined interfaces to the network elements of the active network and to the workstation computers of operating personnel. OSS network elements that are connected to several BSS or MSS units are referred to as OMCs. A radio OMC, for example, is responsible for several BSCs and their BTSs.

Mobile equipment management The management of mobile equipment by the OSS only concerns information about owner and equipment identity, whereas the MSS coordinates the movements of the equipment, including *roaming*, *handover* and *paging*. For example, an OSS can search for stolen or defective equipment using its own database, an EIR, for storing data about equipment and its ownership (some operators have not established the EIR).

Operation and maintenance centre (OMC) The OMC centrally monitors and controls the other network elements and guarantees the best possible service quality for a network. It relies on services of the network management and control functions allocated to the network elements by the hierarchical network management system (TMN). Operator commands are used for intervention into the network elements, while the network management is alerted of any unexpected occurrences in the network. The OMC is connected to all network elements over the standardized O-interface (an X.25-interface). The management functions of the OMC include administration of subscribers and equipment, billing, and generation of statistical data on the state and the capacity utilization of network elements.

Authentication centre (AuC) The AuC contains all the information required to protect a subscriber's identity, and his mobile communication against eavesdropping, and his right to use the radio interface. Because the radio interface is generally susceptible to unauthorized access, special measures (e.g., authentication key assigned to each subscriber and coding of transmitted information) were undertaken in order to prevent the fraudulent use of GSM–PLMN connections. Authentication algorithms and encryption codes are stored in the AuC, and strict rules apply for access to this information (see Section 3.13).

Equipment identity register (EIR) The EIR is a central database in which subscriber and equipment numbers (*International Mobile Equipment Identity*,

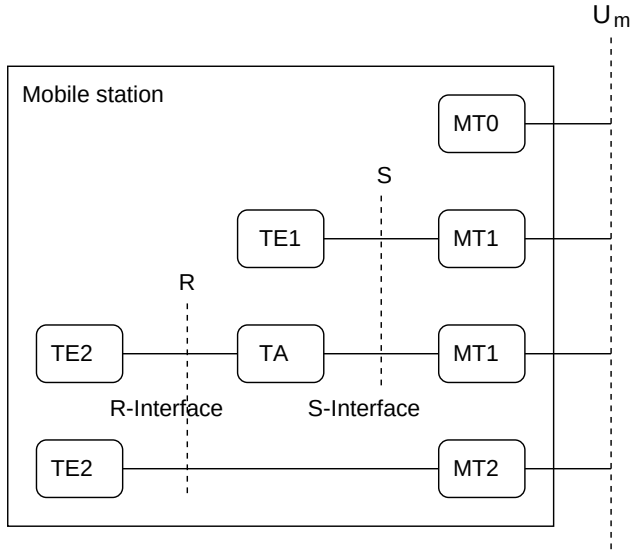


Figure 3.3: Mobile station network terminations with the reference points R, S, U_m

IMEI) are stored, and is connected over an interface to the NSS network elements and the OSS. The database contains a *white*, a *black* and a *grey* list. The white list contains the `IMEI` list of valid mobile radio stations; the black list contains all the `IMEIs` of stolen or suspended mobile radio stations. The grey list includes a list of `IMEIs` for malfunctioning equipment that is not receiving any services.

3.2.2 Interfaces of the GSM System

3.2.2.1 User Interface of the Mobile Station

A GSM mobile station consists of the *terminal equipment* (TE) to which the subscriber has direct access, a *terminal adapter* (TA) (if required) and a part that contains the functions shared by all the services and referred to as *mobile termination* (MT) in the GSM specifications. The subscriber interface on the terminal (TE) contains the network termination and the different equipment functions (see Figure 3.3).

The following mobile network terminations are used:

MT0 (Mobile Termination Type 0) A network termination for the transmission of speech and data integrating the terminal equipment, the terminal equipment functions and sometimes a TA.

MT1 (Mobile Termination Type 1) A network termination with an external ISDN S-interface to which an ISDN terminal (TE1) can be connected.

Conventional terminal equipment (TE2) corresponding to the ITU-T, V or X-series can be connected to an MT1 through the use of an ISDN terminal adapter (TA).

MT2 (Mobile Termination Type 2) This is a network termination with an external R-interface to which conventional terminal equipment corresponding to the ITU-T, V or X-series can be connected.

TE1, TE2 and TA correspond to comparable functional groups of the ISDN concept. The radio interface that supports ISDN-compatible access over traffic and signalling channels is located at reference point U_m.

3.2.2.2 Radio Interface

This is an important interface in the GSM system, and is therefore covered in detail in Section 3.3.

3.2.2.3 BTS–BSC Interface at Reference Point A_{bis}

Transmission over the A_{bis}-interface (see Figure 3.1) is based on PCM-30 and 64 kbit/s interfaces.

Because PLMN network operators frequently are not also the operators of the telecommunications networks, a submultiplex technique that transmits four 16 kbit/s channels over a 64 kbit/s channel was standardized to save on line costs.

3.2.2.4 BSS–MSC Interface at Reference Point A

Speech and data are transmitted digitally over the A-interface (see Figure 3.1), over PCM-30 systems based on the ISDN standard (ITU-T-Series G.732). A PCM-30 system has 30 full-duplex channels at 64 kbit/s, with a transmission rate of 2.048 Mbit/s full-duplex. Two channels each with 64 kbit/s are required for synchronization and signalling (D_2-channel).

3.2.2.5 BSC/MSC–OMC Interface at Reference Point O

The O-interface is based on ITU-T recommendation X.25, which was specified for the attachment of data terminal equipment to packet-switched networks. Physically this interface can be implemented over a 64 kbit/s channel. The option exists to use interfaces of line-switched networks, e.g., V.24bis or X.21.

3.3 The Interface at Reference Point U_m

This radio interface is located between the mobile station (MS) and the rest of the GSM network. Physically the information flow takes place between the mobile station and the base transceiver station (BTS). But, viewed logically,

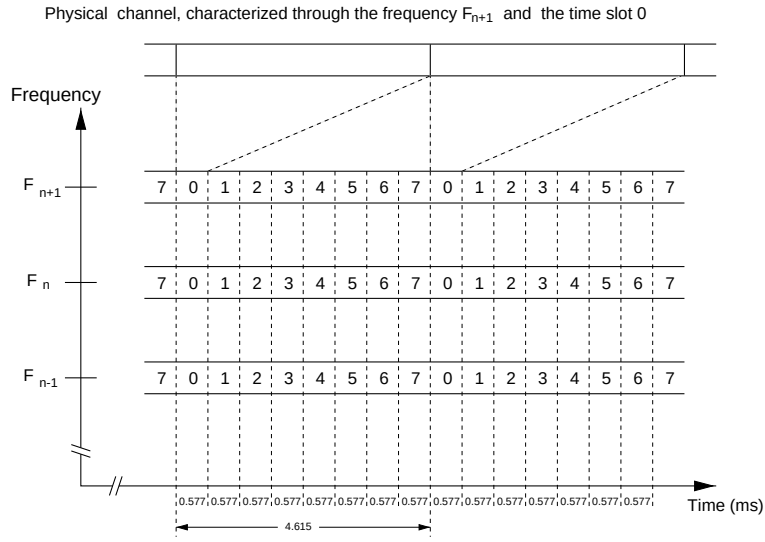


Figure 3.4: Realization of physical channels using FDM and TDM

the mobile stations are communicating with the base station controller (BSC) and the mobile switching centre (MSC). The gross transmission rate over the radio interface is 270.833 kbit/s.

3.3.1 Multiplex Structure

Along with voice coding and modulation, multiplexing is also very important. In the GSM recommendations a combination of *frequency-division multiplexing* (FDM) and *time-division multiplexing* (TDM) has been standardized, providing multiple access by mobile stations to these systems (FDMA, TDMA).

Figure 3.4 shows how a physical channel is produced through a combination of FDM and TDM (see channel 0 on frequency F_{n+1} and Sections 3.3.1.1 and 3.3.1.2).

GSM utilizes the cellular concept, already proven successful in analogue mobile radio networks, in which a geographical area is divided into planned radio cells (in the simplest case hexagons), with one BTS per cell with which the mobile stations can make contact. The radio cells, each having the exclusive use of specific FDM channels, are combined into groups (*clusters*). The same frequencies are only reused after a sufficiently long distance in neighbouring clusters (see Section 2.3).

The cell radius can vary according to user density. The likelihood that a mobile user will leave a cell during a call, thereby necessitating a *handover*, is less in large radio cells than in small cells. Small cells, on the other hand, make more efficient use of a frequency band because they operate with a lower transmitter power, the cluster is less spread out and consequently the available

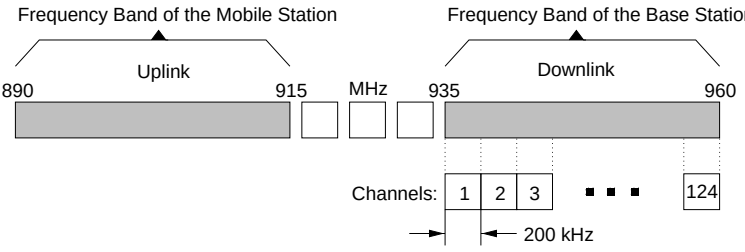


Figure 3.5: Frequency bands used by GSM

frequencies can be reused at smaller physical intervals. In practice, the size of cells is determined by traffic volume, the maximum transmitter power of the BTS of the frequencies allocated to a cell and morphological conditions.

Thus cells in rural areas can have a radius of up to 35 km. Larger cell radii would cause a higher round-trip propagation delay; the maximum delay is 0.233 ms, much larger than specified in the standard. In metropolitan areas the radius might only be at 300 m, which allows a traffic volume of up to 200 Erl./km^2. Cells are divided into sectors in order to increase capacity (see Section 2.4).

3.3.1.1 Frequency-Multiplexing Structure

One of the most important criteria in designing a radio interface was efficient utilization of the available frequency band. In Europe two 25 MHz wide frequency bands in the 900 MHz band were reserved for GSM. Transmission from the mobile unit to the base station (*uplink*) takes place in the 890 MHz to 915 MHz range; in the reverse direction (*downlink*) the 935–960 MHz frequency band is used in a *frequency-division duplex* (FDD) mode of operation. 15 MHz at the lower band limit and 1 MHz at the upper band limit will not be available until 2001. After current use is discontinued, an additional 10 MHz between 880 and 890 MHz and between 925 and 935 MHz will be available as a GSM extension band (see Appendix C). A duplex interval of 45 MHz exists between the transmit and receive frequencies.

The frequency bands are divided into 200 kHz bandwidth channels, therefore providing a total of 124 FDM channels each for transmitting and receiving operations (see Figure 3.5).

Each mobile station can occupy all 124 carrier frequency pairs, although according to the GSM specifications use of channels 1 and 124 should be avoided if possible. The respective 200 kHz bandwidth is kept as a guard band for the neighbouring systems in the frequency band. If the carrier frequencies on the uplink are denoted by F_u and those on the downlink as F_d then the GSM band can be defined as

$$F_u(n) = 890.2 \text{ MHz} + 0.2(n-1) \text{ MHz} \quad (1 \leq n \leq 124) \tag{3.1}$$

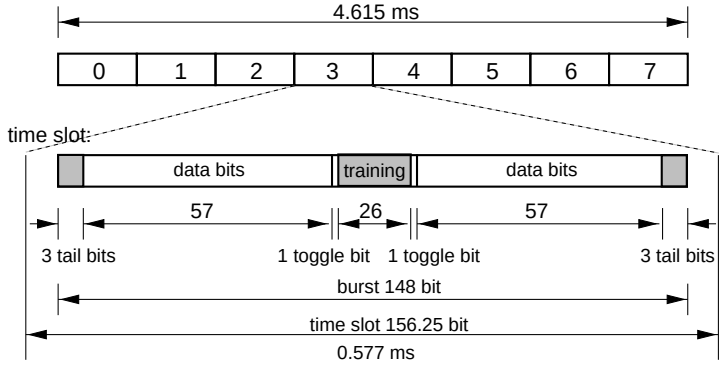


Figure 3.6: Structure of a TDMA frame

$$F_d(n) = 935.2 \text{ MHz} + 0.2(n-1) \text{ MHz} \quad (1 \leq n \leq 124) \tag{3.2}$$

and the extension band as

$$F_u(n) = 880.2 \text{ MHz} + 0.2(n-1) \text{ MHz} \quad (1 \leq n \leq 50) \tag{3.3}$$

$$F_d(n) = 925.2 \text{ MHz} + 0.2(n-1) \text{ MHz} \quad (1 \leq n \leq 50) \tag{3.4}$$

3.3.1.2 Time-Multiplexing Structure

With the TDM method a carrier frequency is divided into eight physical TDM channels in which the time axis is divided into eight *periodic time slots* of 0.577 ms duration. Eight time slots are combined into a TDM frame of 4.615 ms duration (see Figure 3.6). Because these time channels are used in multiple access, the frame is referred to as TDMA frame in the GSM recommendations.

A physical channel is characterized by its carrier frequency and the time slot available to it, which recurs every 4.615 ms. Each time slot has a length corresponding to the duration of 156.25 bits or 0.577 ms (15/26 ms). This length is produced from the transmission rate of the modulation method (1625/6 kbit/s) and the number of bits to be transmitted in a slot. A slot is used by a *burst* with a length of 148 bits, which, corresponding to the guard time, is 8.25 bits shorter in duration than the slots to avoid overlapping with other bursts. Data is transmitted in bursts. If messages are longer than a burst, they are split up among several bursts and then transmitted.

Overall there are five types of bursts (see Figure 3.7 [14]) which differ from one another in function and content. The tail bits that occur in all bursts are defined as modulation bits and always have the same value as specified in the standard. The bursts are sent so that the bits with the lowest value are transmitted first.

Normal burst For transmitting messages in traffic and control channels.

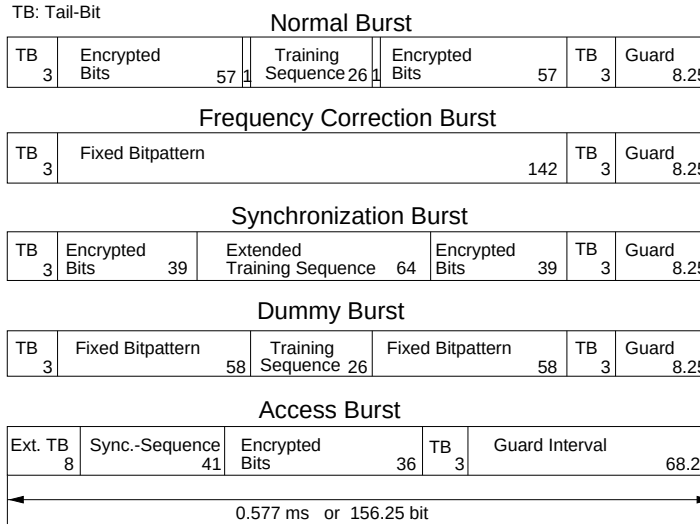


Figure 3.7: Bursts used in GSM

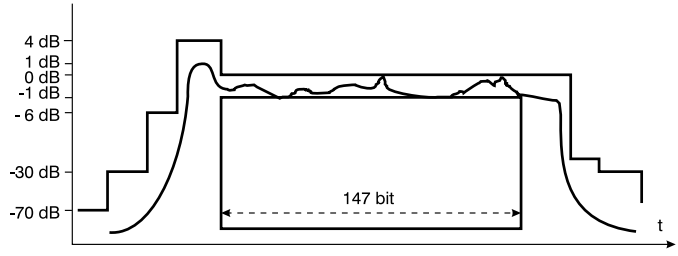


Figure 3.8: Envelope of the radio signal of a burst

Access burst Used for call setup. This burst is shorter than the others because it does not require the MS to be fully synchronous with the BTS.

Synchronization burst Sent by the base station and used for synchronization.

Frequency correction burst Sent by the base station and used for frequency correction at the mobile station to prevent possible interference from neighbouring frequencies.

Dummy burst Placed in an empty slot if no data is being sent.

The signalling characteristics of a burst over time are not allowed to exceed the area of a prescribed mask (see Figure 3.8). In the area of the tail bits and the guard space the signal can deviate considerably from the standard 0 dB. It is clear that neighbouring bursts only minimally overlap in the same TDMA frame.

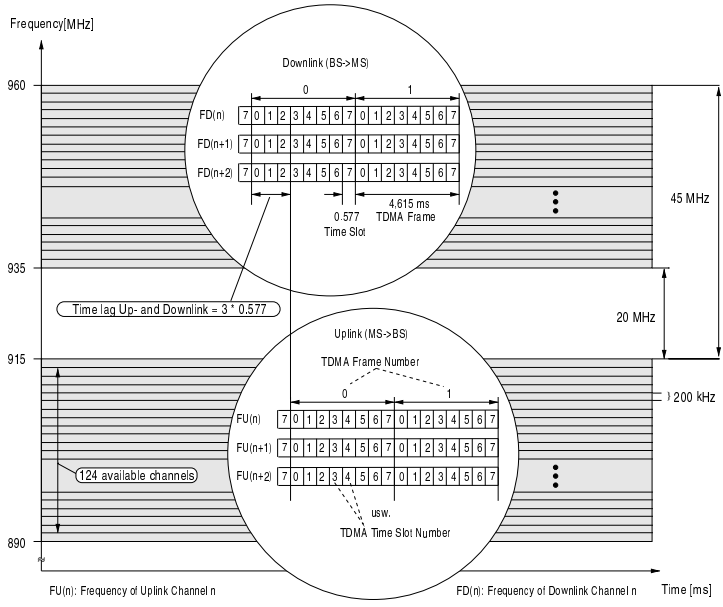


Figure 3.9: Time delay between uplink and downlink

The time-division multiplexing technique is applied to the uplink and to the downlink channel. So that the mobile stations do not have to transmit and receive at the same time, the TDMA frames from the uplink are transmitted with a delay of 3 time slots (see Figure 3.9). The parameter *timing advance* (TA) is used by the BTS to compensate for the round-trip signal propagation delay BTS-MS-BTS. The value of the 6 bit of TA indicates to a receiving mobile how many bit durations (3.7 μs each) it must transmit its burst earlier than as derived from the received slot tact signal to reach synchronization with the slot tact defined by the BTS.

3.3.2 Frequency Hopping (FH)

Since multipath reception and co-channel interference can affect the quality of certain FDM channels, an optional method called *frequency hopping* is applied. With this method the frequency is changed after each transmitted frame of a channel (see Figure 3.10). The frequency change, which can last approximately 1 ms, takes place between the receiving or the transmitting time slots.

The sequence of frequencies in a hopping cycle through which a mobile station passes is calculated with an algorithm implemented in each MS. The advantage of this procedure is that all mobile subscribers are guaranteed transmission channels with nearly the same quality. During data transmission, interference from co-channels in the cycle is limited for each frequency

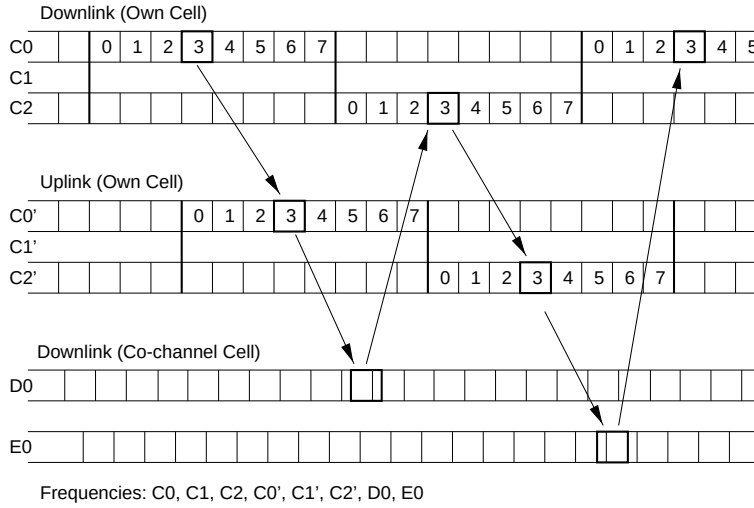


Figure 3.10: Frequency hopping method

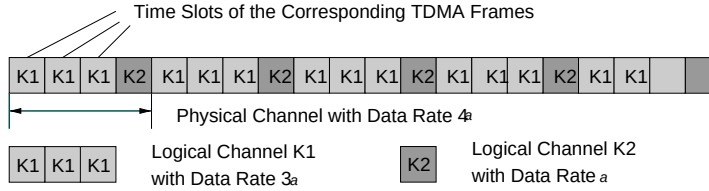


Figure 3.11: Relationship between logical and physical channels

to the duration of one burst only and can be eliminated through error handling; effective error-correction procedures are standardized for voice and data transmission.

3.3.3 Logical Channels

Logical channels occur through the allocation of time slots by physical channels. Consequently the data of a logical channel is transmitted in the corresponding time slots of the physical channel. During this process, logical channels can occupy a part of the physical channel or even the entire channel. For instance, if a physical channel has a transmission rate of $4a$, then a logical channel K1 with a data rate of $3a$ and a second logical channel K2 with a data rate a can transmit on the same physical channel (see Figure 3.11).

The GSM recommendations define several logical channels for signalling on the basis of this principle, dividing them into two main groups: traffic channels and control channels.

Table 3.5: Traffic channels in the GSM recommendation

Traffic channel	Abbreviation
Full-rate TCH for speech	TCH/FS
Half-rate TCH for speech	TCH/HS
9.6 kbit/s full-rate TCH for data	TCH/F9.6
4.8 kbit/s full-rate TCH for data	TCH/F4.8
4.8 kbit/s half-rate TCH for data	TCH/H4.8
$\leq$ 2.4 kbit/s full-rate TCH for data	TCH/F2.4
$\leq$ 2.4 kbit/s half-rate TCH for data	TCH/H2.4
Cell broadcast channel	CBCH

3.3.3.1 Traffic Channels

Traffic channels (TCH) are logical channels over which user information are exchanged between mobile users during a connection. Speech and data are digitally transmitted on these channels using different coding methods.

Different transmission capacities are required depending on the type of service used (e.g., voice transmission, short-message service, data transfer, facsimile). A distinction is therefore made between the following traffic channels:

B_m-channel Transmission over a B_m-channel (*m=mobile*), which is also called a full-rate traffic channel (*full-rate TCH*), is carried out at a gross data rate of 22.8 kbit/s. Digitalized and coded speech only require 13 kbit/s for transmitting voice information. The remaining capacity in voice transmission is used for error correction. It is possible to transmit data at 12, 6 or 3.6 kbit/s over a B_m-channel.

L_m-channel The half-rate traffic channel (*half-rate TCH*) transmits at a gross rate of 11.4 kbit/s. The number of channels in GSM can be doubled in a given frequency band because of the speech codecs available for half-rate channels. Efficient speech coding algorithms were developed in 1995; they were introduced commercially in 1997/98. Half-rate TCHs allow data to be transmitted at bit rates of 6 or 3.6 kbit/s.

Table 3.5 lists the traffic channels specified in the GSM recommendation.

3.3.3.2 Control Channels

Control information is used for signalling and for system control and is not passed down to the subscribers. Typical signalling tasks include the signalling for establishing, maintaining and releasing traffic channels, for mobility management and access control to radio channels.

Control information is transmitted over so-called *control channels* (CCH), which, following ISDN, are also referred to as D_m-channels. The control channels offer the mobile stations a packet-oriented continuous signalling service

Table 3.6: Control channels in GSM

Direction	Group	Channel	Channel identification
MS ← BS	BCCH	BCCH	Broadcast Control Channel
MS ← BS		FCCH	Frequency Correction Channel
MS ← BS		SCH	Synchronization Channel
MS ← BS	CCCH	PCH	Paging Channel
MS → BS		RACH	Random Access Channel
MS ← BS		AGCH	Access Grant Channel
MS ↔ BS	DCCH	SDCCH	Stand-Alone Dedicated Control Channel
MS ↔ BS		SACCH	Slow Associated Control Channel
MS ↔ BS		FACCH	Fast Associated Control Channel

enabling them within the PLMN to receive messages from the base stations and to send messages to the base stations at any time.

Because the control and management of a mobile radio network is far more complex from the standpoint of signalling than a fixed network, three groups of control channels were defined in GSM:

- *Broadcast control channel* (BCCH)
- *Common control channel* (CCCH)
- *Dedicated control channel* (DCCH)

Table 3.6 contains a list of all the control channels defined in the GSM recommendations, and in the directional column indicates the directions possible on each channel (uplink, downlink or both).

Broadcast Control Channel (BCCH) This channel is used to transmit information about the PLMN from the base station to the mobile stations in the radio cell through a point-to-multipoint connection. The kind of information conveyed over a BCCH includes identification of the network, availability of certain options such as *frequency hopping* and *voice activity detection* and identification of the frequencies being used by the base station and neighbouring base stations.

One of the subchannels of the BCCH is the *frequency correction channel* (FCCH), used for transmitting a frequency correction burst to the mobile station for possibe correction of the transmitting frequency.

Another subchannel of the BCCH is the *synchronization channel* (SCH), used for transmitting synchronization bursts to a mobile station to allow it to time-synchronize.

Messages transmitted over the BCCH and its subchannels are transmitted exclusively in simplex mode by the base station to the terminal equipment.

Common control channel (CCCH) This designation is an umbrella term for control channels that handle the communication between the network and the mobile phone. Included among the CCCH channels are:

Paging channel (PCH) This channel exists only on the downlink, and is activated for the selective addressing of a called mobile terminal during a connect request from the network (incoming call).

Random access channel (RACH) This access channel only occurs on the uplink, and allows the mobile station, using an S-ALOHA access protocol, to request channel capacity from the base station to establish a connection.

Access grant channel (AGCH) The base station uses this logical channel to respond to a message received over the RACH from a mobile station. In accordance with the call setup mechanism selected by the network operator, the mobile station is allocated an SDCCH or a TCH over the AGCH that only exists on the downlink; see Section 3.5.1.

Dedicated control channel (DCCH) This designation is an umbrella term for three bidirectional point-to-point control channels that are used to transmit signalling messages for call control at different bit rates. The three DCCH channels are:

Stand-alone dedicated control channel (SDCCH) This channel is always used when a traffic channel has not been assigned, and is allocated to a mobile station only as long as control information is being transmitted. The channel capacity available from an SDCCH is 782 bit/s, which is much lower than that of a TCH. Control information transmitted on the SDCCH includes registration, authentication, location area updating and data for call setup.

Slow associated dedicated control channel (SACCH) This channel is always allocated parallel to a TCH or an SDCCH. It is used to transmit at a data rate of 383 bit/s system information from the network to the mobile station and measurement data on signal strength and receive quality from the MS to the network.

Fast associated dedicated control channel (FACCH) This channel is set up in the short term only when a traffic channel exists and then it uses its time slots. As an example, an FACCH is set up for an impending handover and the necessary control data is transmitted over the FACCH. This channel can handle bit rates of 4600 bit/s or 9200 bit/s.

3.3.4 Hierarchy of Frame Structures

The division of the GSM frequency spectrum into 124 FDM channels and its TDMA channel structure, the tasks of the logical channels and the use of the different bursts have been explained in previous sections.

In GSM the TDMA frames, which contain eight time slots for transmitting bursts, are combined into *multiframes*. A distinction is made between multiframes of two different lengths:

- 26-frame multiframe
- 51-frame multiframe

The bursts of the traffic channels (TCH) and the SACCHs and FACCHs assigned to them are transmitted in 26-frame multiframes. Each traffic channel is allocated 1 of 8 (with half-rate transmission, 16) time slots of a TDMA frame. The associated 8 SACCHs are transmitted in the 12th TDMA frame. The last TDMA frame (25) of the 26-frame multiframe is only used if another 8 SACCHs are required for half-rate transmission. Time slots are stolen from the TCHs for the transmission of the FACCHs.

The data of the FCCH, SCH, BCCH, RACH, AGCH, PCH, SDCCH, SACCH/C and CBCH channels is sent in 51-frame multiframes. It was generally specified that speech and data would be transmitted in 26-frame multiframes and signalling data (except for SACCH/T and FACCH) in 51-frame multiframes. However, there was some deviation from this rule when the packet-data service was introduced (see Section 3.11.3).

Fifty-one of the 26-frame multiframes and 26 of the 51-frame multiframes are combined into a *superframe*, and 2048 superframes produce a *hyperframe* (see Figure 3.12). It takes almost 3.5 hours to transmit a hyperframe.

Figure 3.13 shows the relationship between 51-frame and 26-frame multiframes. The BCCH frames in a 51-frame multiframe are decoded every 240 ms during the unused 26th frame of the 26-frame multiframe.

3.3.5 Combinations of Logical Channels

Logical channels are mapped onto a physical channel through a process of cyclically recurring multiframe patterns. The correct positioning timewise of the cycles is achieved through synchronization of BTS and MS. The BSS provides a reference counter in each cell that is used to number the time slots and serves as the mapping scheme for the multiframes. In GSM each time slot is given a number. This number along with the TDMA frame number allows each time slot to be clearly identified (see Figure 3.9). Because of a frame structure encompassing several levels, the counter for the TDMA frame numbers can go as high as 2 715 648. This high a number was necessary for cryptographic reasons, because the enciphering algorithm uses the TDMA frame number as an input parameter.

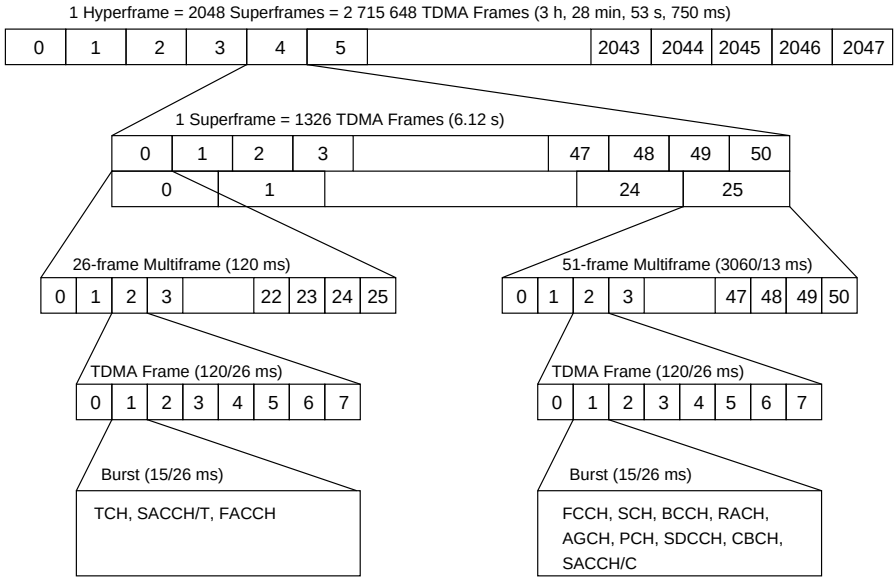


Figure 3.12: The structure of a transmission medium with TDMA frames, multiframes, superframes and hyperframes

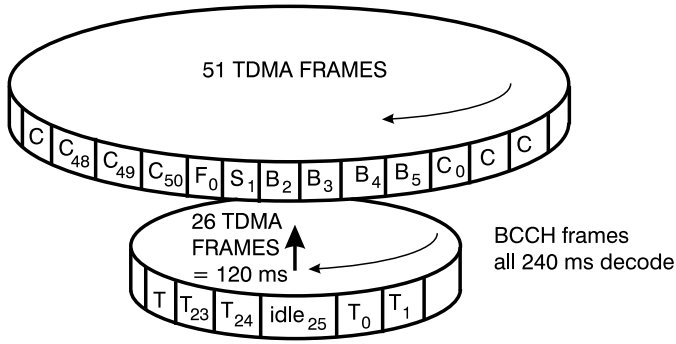


Figure 3.13: The relationship between 51-frame and 26-frame multiframes

Figure 3.14 shows how traffic and signalling channels are mapped onto a physical (frequency) channel. For each TDMA frame of the corresponding multiple frame, a slice of the 'cylinder' shown is provided that carries eight time slots (the TDMA frame) [33].

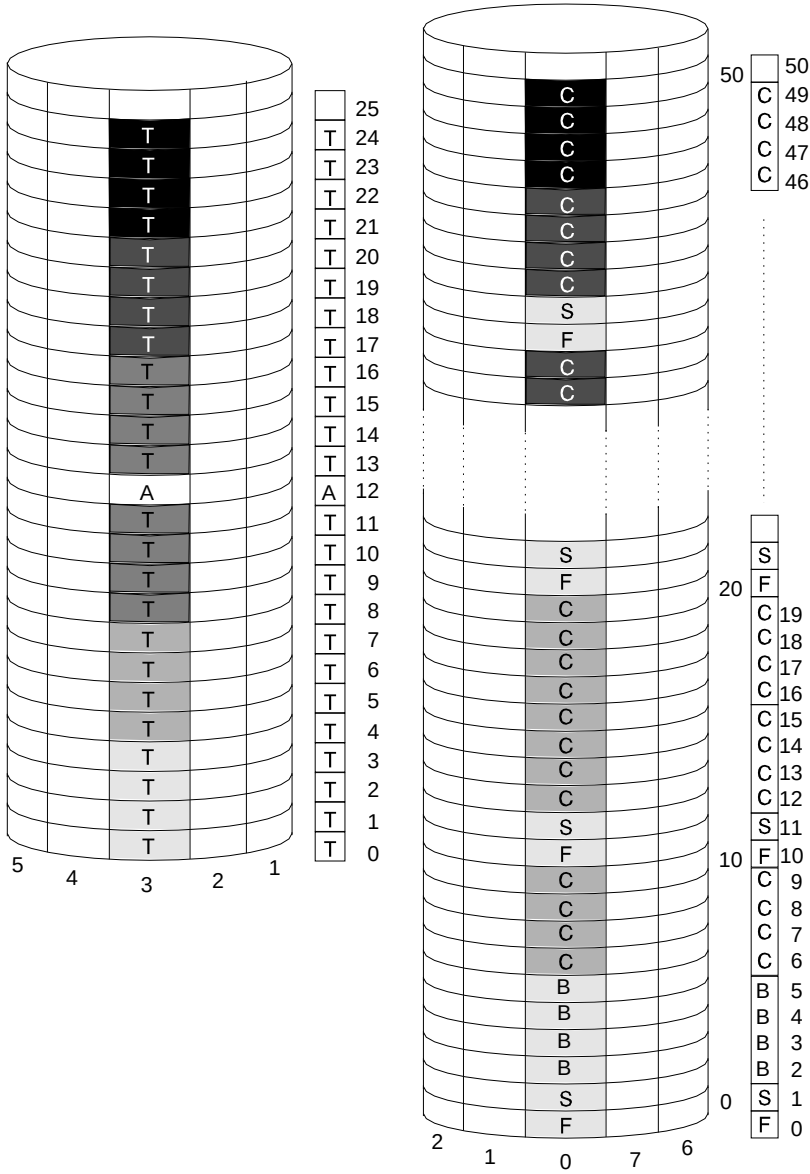


Figure 3.14: Traffic and control channel with 'wind up' time axis

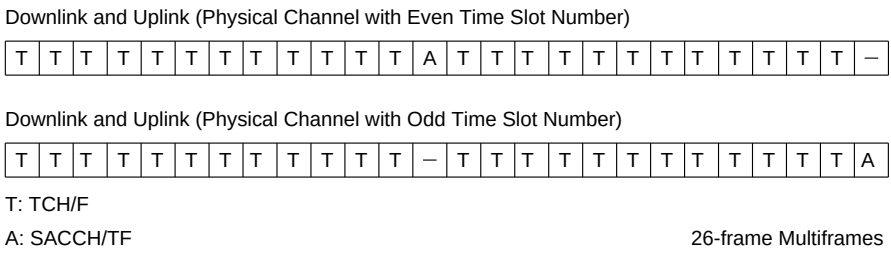


Figure 3.15: Channel combination 1: TCH/F + FACCH/F + SACCH/TF

3.3.5.1 Approved Channel Combinations

Logical channels are not simply distributed over the two multiframes, but follow a particular pattern. The following is a set of approved channel combinations (CC) [14], represented by their multiframes:

CC1:	TCH/F + FACCH/F + SACCH/TF
CC2:	TCH/H(0,1) + FACCH/H(0,1) + SACCH/TH(0,1)
CC3:	TCH/H(0) + FACCH/H(0) + SACCH/TH(0) +TCH/H(1)
CC4:	FCCH + SCH + BCCH + CCCH
CC5:	FCCH + SCH + BCCH + CCCH + SDCCH/4(0,1,2,3) + SACCH/C4(0,1,2,3)
CC6:	BCCH + CCCH
CC7:	SDCCH/8 + SACCH/8

A physical channel exactly contains one of these combinations. The group CCCH divides into the channels PCH, AGCH and RACH. The numbers in brackets after the logical channels give the numbers of the logical subchannels. This means that in the case of SDCCH/4 there are four SDCCHs on a physical channel, each of which is used by one mobile station.

The shared use of a physical channel by several logical channel types does not mean that all of these channels can be used at the same time. Although the mapping of combinations of logical channels onto a physical channel may mean that several logical channels can appear on one physical channel, they occur sequentially at intervals of at least one TDMA frame length. The sequential numbering of the TDMA frames ensures that the physical layers of both communicating partners use the currently appropriate logical channel for transmitting and receiving.

The multiframes in the approved channel combinations are shown in Figures 3.15–3.21. The mapped multiframe combinations are derived from tables found in Series 05.02 of the GSM standard [14]. The tables describe which time slots and which frequencies the respective logical channel is allowed to use.

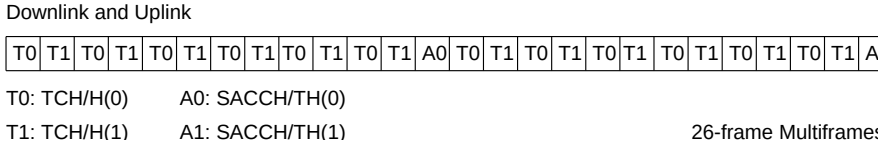


Figure 3.16: Channel combination 2: TCH/H(0,1) + FACCH/H(0,1) + SACCH/TH(0,1)

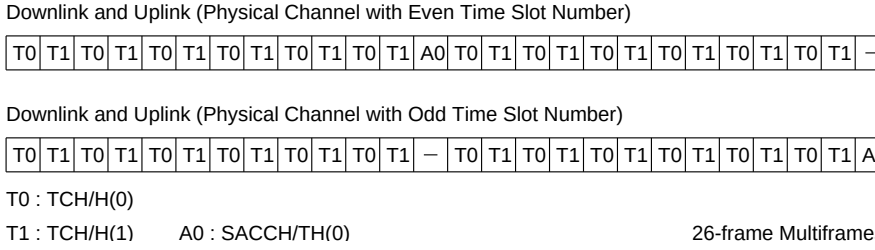


Figure 3.17: Channel combination 3: TCH/H(0) + FACCH/H(0) + SACCH/TH(0) + TCH/H(1)

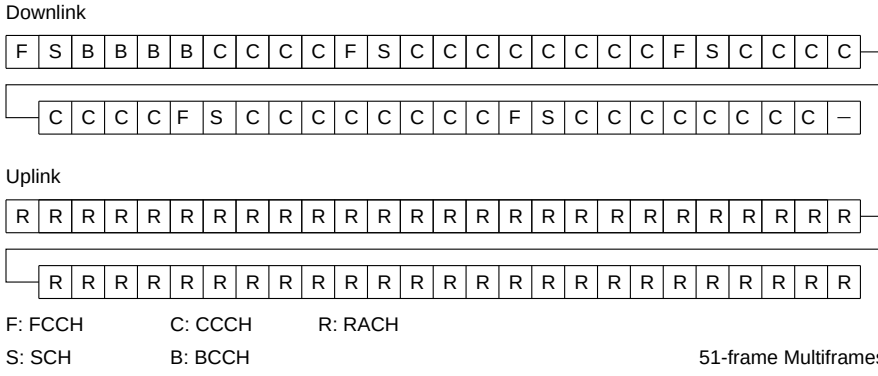


Figure 3.18: Channel combination 4: FCCH + SCH + BCCH + CCCH

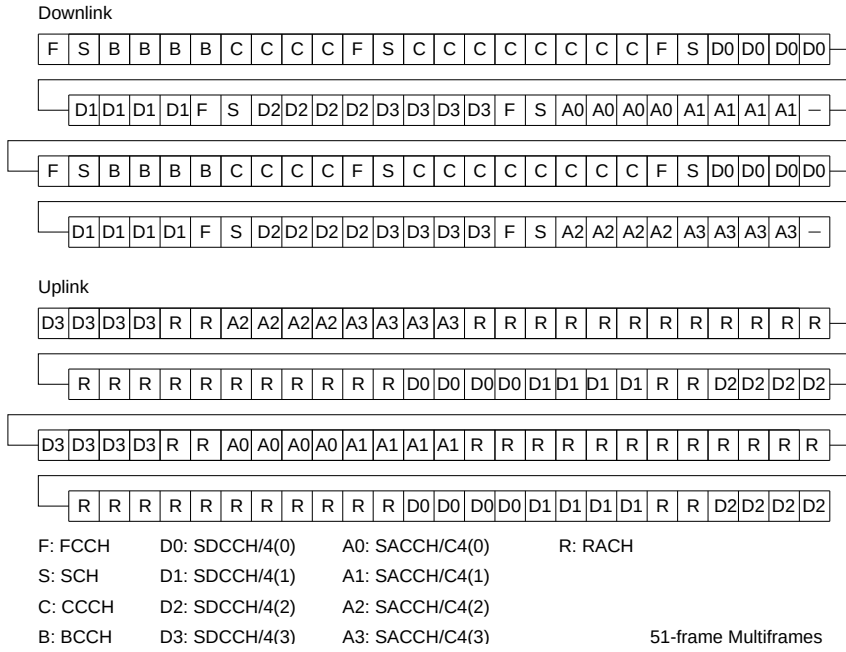


Figure 3.19: Channel combination 5: FCCH + SCH + BCCH + CCCH + SDCCH/4 (0, 1, 2, 3)

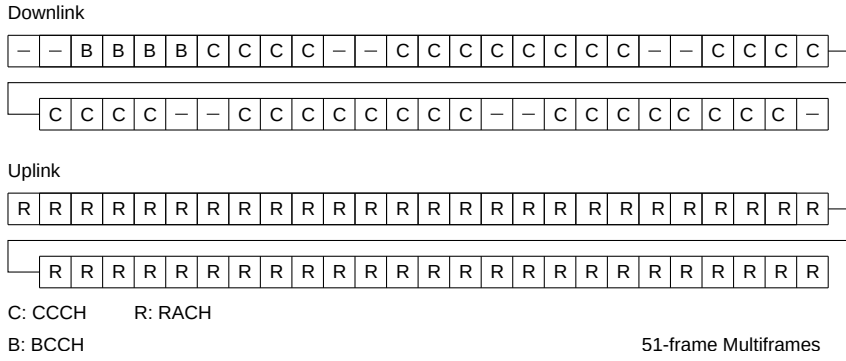


Figure 3.20: Channel combination 6: BCCH + CCCH

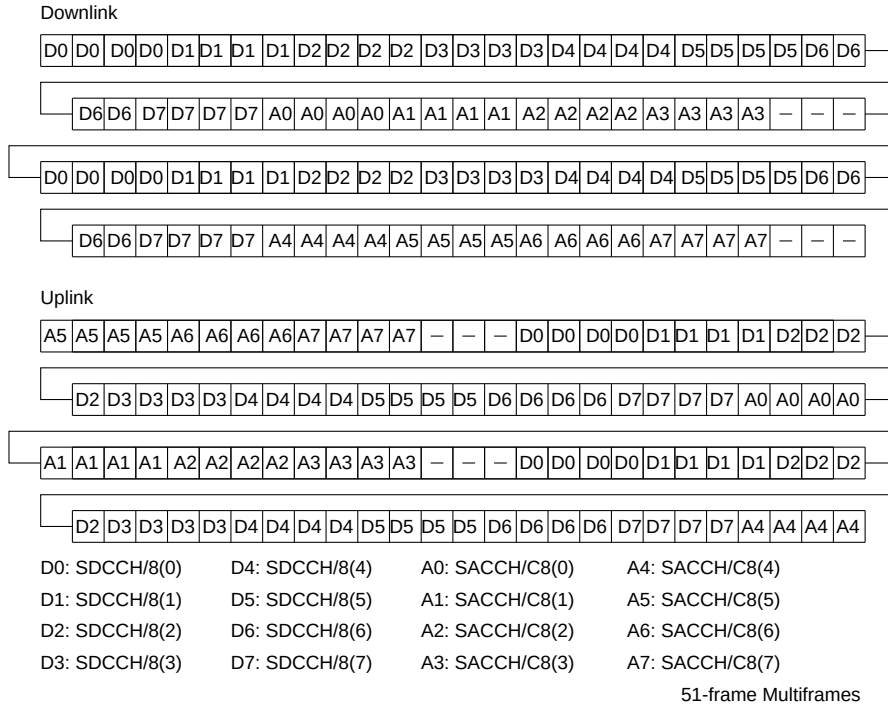


Figure 3.21: Channel combination 7: SDCCH/8 + SACCH/8

A cell in a GSM-PLMN has a set of $n + 1$ frequency channels available for its use. These frequency channels are denoted by $C_0, C_1, \ldots, C_n$. The indices have no relationship to the frequency. The described channel combinations are not allowed to occupy just any random time slots or frequencies.

For example, the combination 4 shown in Figure 3.18 is only allowed to be transmitted in time slot 0 of the carrier frequency C_0. On the downlink the BTS must transmit in each time slot number 0 the logical channel of combination 4 to enable the mobile station to carry out performance measurements. If no information is available for sending, a dummy burst is transmitted (see Section 3.3.1.1).

The combination 5 shown in Figure 3.19 is only allowed to occupy time slot 0 in frequency C_0. None of the other combinations, except for combinations 4 and 5, are allowed to use this physical channel.

The channel combination 6 (see Figure 3.20) occupies time slots 2, 4 and 6 of the carrier frequency C_0. Except for the restriction mentioned above, channel combinations 1, 2, 3 and 7 are allowed to use any time slots in any frequencies.

3.3.6 Channel Combinations of a Cell Depending on Anticipated Cell Utilization

The BSS provides a cell with a set of logical channels occupying several physical channels. On the basis of the anticipated traffic load of a cell, the network operator establishes a channel configuration that must adhere to the rules mentioned in the last section. Each individual *transceiver* (TRX) can offer eight channel combinations in each time slot. The time slot is identified through a *time slot number* (TN). Three common combinations are briefly presented here:

- Low-capacity cell with one TRX:
  - TN 0: FCCH + SCH + BCCH + CCCH + SDCCH/4(0,1,2,3) + SACCH/C4(0,1,2,3)
  - TN 1 to 7: TCH/F + FACCH/F + SACCH/TF
- Medium-capacity cell with four TRXs:
  - Once on TN 0: FCCH + SCH + BCCH + CCCH
  - Twice (on TN 2 and TN 4): SDCCH/8 + SACCH/8
  - 29 times: TCH/F + FACCH/F + SACCH/TF
- High-capacity cell with 12 TRXs:
  - Once on TN 0: FCCH + SCH + BCCH + CCCH
  - Once on TN 2: BCCH + CCCH
  - Once on TN 4: BCCH + CCCH
  - Once on TN 6: BCCH + CCCH
  - 5 times: SDCCH/8 + SACCH/8
  - 87 times: TCH/F + FACCH/F + SACCH/TF

After successful synchronization, the mobile station is informed through the system information of the BCCH which channel combinations on which physical channels are being offered to it by the BSS. Depending on its current operating state (idle state or dedicated state), it uses a particular subset from this offering of channels.

What is noticeable is that a BCCH always appears together with the logical channels SCH und FCCH and that it can always be found in time slot 0. This condition helps to facilitate synchronization when an MS makes its first contact with a BTS. Additional combinations 6 (see Figure 3.20) are added when traffic is expected to be heavy.

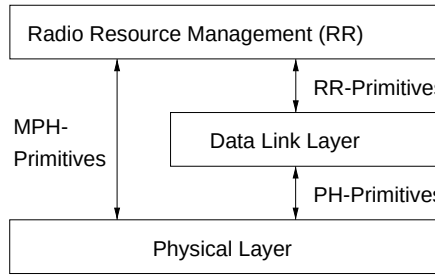


Figure 3.22: Interfaces of layer 1 (*physical layer*)

3.3.7 Layer 1: Physical Transmission

GSM layer 1 is essentially comparable to the physical transmission layer (*physical layer*) in the ISO/OSI reference model (see Section 2.5). Unlike the ISO/OSI model, an entity of layer 3, *radio resource management* (RR), directly accesses this lowest layer (see Figure 3.22).

This access allows channel allocations to be controlled and layer-3 information to be queried on the state of the *physical layer* and the radio connection, such as channel values, error ratios and received signal strength. This information is required by layer 3 in order for it to fulfil the typical cellular networks tasks such as *handover* and *roaming*.

Layer 1 of a GSM implementation is responsible for the radio transmission of traffic and signalling information. Its main tasks include:

- Producing bursts, multiplexing the bursts in TDMA frames, and transmitting the frames on the available physical channels over dedicated control and traffic channels.
- Seeking and seizing a broadcast control channel (BCCH) by the MS.
- Judging channel quality and received signal strength.
- Error detection and correction mechanisms (defective blocks are not forwarded to layer 2).
- Synchronization with frame transmission.
- Encoding of data stream.

The GSM system uses a digital 0.3 GMSK modulation format. GMSK stands for *Gaussian Minimum-Shift Keying*, in which *minimum shift* means that there is no discontinuity between the phase of the carrier and the time. The modulation format 0.3 GMSK means that the modulated signal passes through a Gauss filter with the product of the 3 dB bandwidth B and the bit period T producing the value $BT = 0.3$.

Table 3.8: Coding methods for logical channels

Channel type	Bit/Block Data+Parity+Tail	Convolutional coding rate	Bit/ Block	Interleaving depth
TCH/FS			456	8
Class I	$182 + 3 + 4$	1/2	(378)	
Class II	$78 + 0 + 0$	-	(78)	
TCH/F9.6	$4 \cdot 60 + 0 + 4$	244/456	456	19
TCH/F4.8	$60 + 0 + 4$	1/3	228	19
TCH/H4.8	$4 \cdot 60 + 0 + 4$	244/456	456	19
TCH/F2.4	$72 + 0 + 4$	1/6	456	8
TCH/H2.4	$72 + 0 + 4$	1/3	228	19
FACCHs	$184 + 40 + 4$	1/2	456	8
SDCCHs, SACCHs	$184 + 40 + 4$	1/2	456	4
BCCH, AGCH, PCH	$184 + 40 + 4$	1/2	456	4
RACH	$8 + 6 + 4$	1/2	36	1
SCH	$25 + 10 + 4$	1/2	78	1

3.3.7.1 Channel Coding and Interleaving

This section describes how GSM layer-2 messages are conveyed by means of bursts. The two dedicated signalling channels SDCCH and FACCH are especially important. In [12] they are referred to as the *main signalling link*. Table 3.8 presents a general overview of the coding procedures used for logical channels. The following information is provided for each type of channel:

- The number of bits per 456 bit-block of user information which are systematic redundancy and which are filler bits
- The coding rate of the convolutional code used
- The depth of bit interleaving

Different channel coding methods are applied on the various channels for the different services (see Section 2.7.1).

A Fire code—a linear, binary block code family derived from the cyclic block code—is used with most signalling channels. This code has a generator polynomial which offers good error detection and/or error correction when errors occur in groups (*burst errors*):

$$(X^{23} + 1)(X^{17} + X^3 + 1)$$

Forty Fire code parity bits are added to the 184 data bits of a layer-2 protocol data unit (see Figures 3.23 and 3.24). This allows groups of errors of up to 11 bits to be detected and corrected. When layer 1 recognizes a defective block and can no longer correct the errors, it does not forward this block to layer 2.

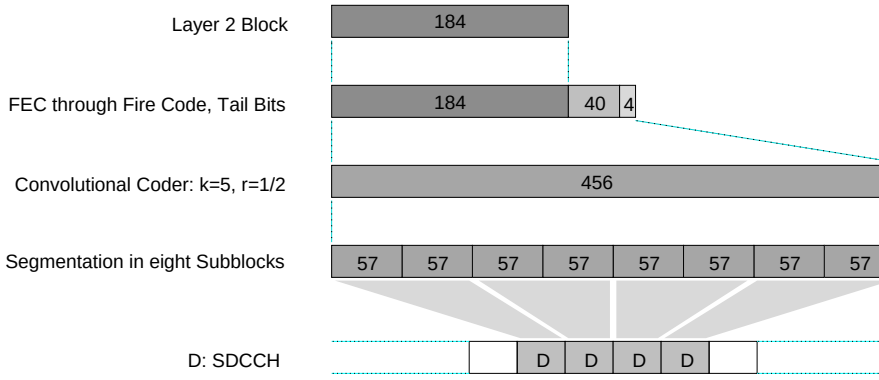


Figure 3.23: Coding and interleaving in an SDCCH

All channels have forward error correction (FEC) protection through convolutional coding. The respective code ratios are given in Table 3.8. A convolutional code with the ratio $r = 1/2$ and the influence length $k = 5$ (see Section 2.7.3.2) is used on the control channels.

Interleaving allows correlated errors caused by a short signal fading to appear as single errors at the decoder of the receiver (see Figure 3.25).

Two types of interleaving are in use, bit interleaving and block interleaving. The bit interleaving with a depth of n is based on an algorithm, where the 456 information bits of a convolutional coded layer-2 block (see Figure 3.23) are arranged into rows mod n into a table (see left-hand upper side of Figure 3.25).

The bits are read out for transmission from the table row by row (see right-hand upper side and lower side of Figure 3.25).

During this bit interleaving procedure, a block interleaving is introduced in addition whereby the first four rows of 57 times 8 bit (right-hand upper side of Figure 3.25), which each fill half a normal burst of a TCH, are inserted into bit positions in this burst with even bit numbers. The remaining four rows are transmitted on odd bit positions in the normal burst. Each of these rows corresponds to one subblock shown in Figure 3.23.

The de-interleaver at the receiver rearranges the interleaved bits into the original sequence. It is clear from the algorithm that a group of errors of maximum length $n - 1$ applied to the interleaved bit stream on the radio medium is resolved into single bit errors after de-interleaving.

The block interleaving procedure further contributes to spread originally neighbouring bits during the radio transmission to improve the resistance against the so-called error bursts resulting from signal fading.

Figure 3.26 (b) shows the error protection mechanisms for the *Random access channel* (RACH); see also Figure 3.32. Because a mobile station's call request should be recognized quickly by the base station and collisions can sometimes occur that must quickly be resolved, the 8-bit message of the accessing mobile station (code for type of access and random number along with

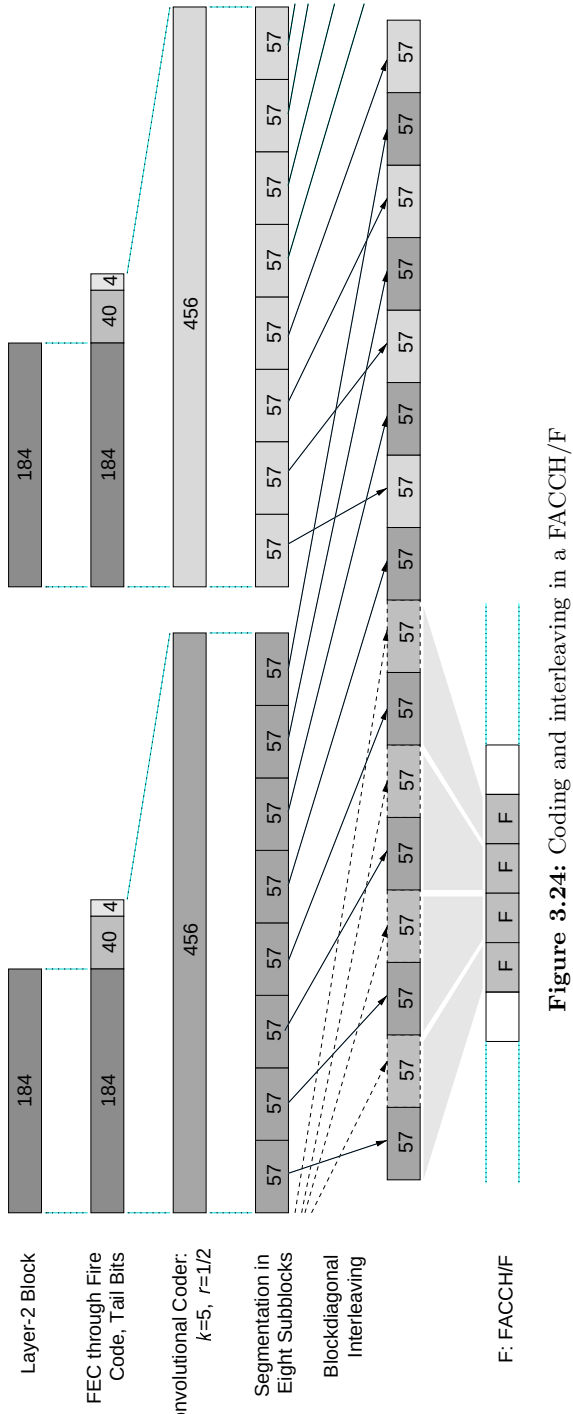


Figure 3.24: Coding and interleaving in a FACCH/F

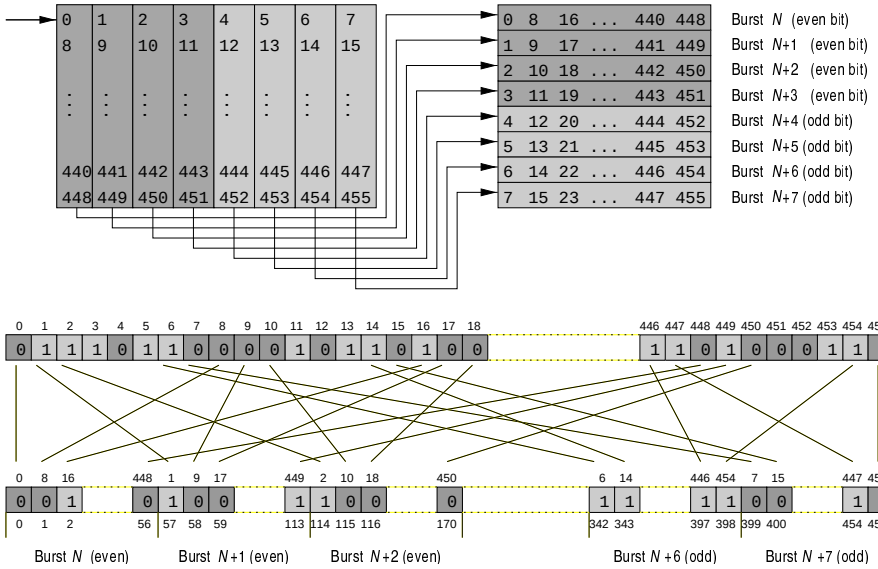


Figure 3.25: Example of interleaving

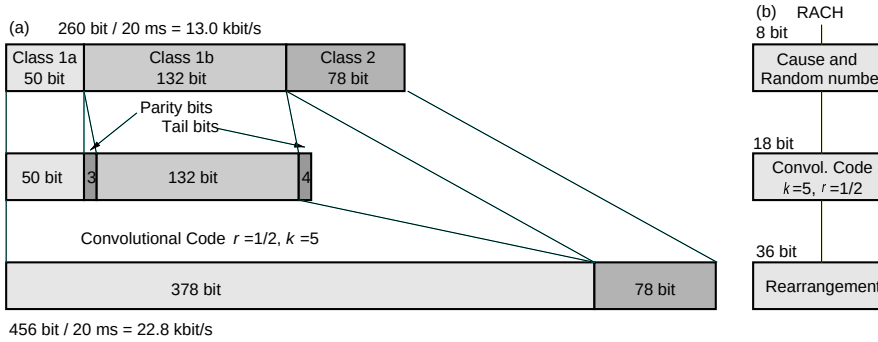


Figure 3.26: Channel coding for (a) speech transmission and (b) signalling

check sum) is modified through the *base station identification code* (BSIC), then convolutionally coded to 36 bits and transmitted without interleaving.

On the FACCHs block diagonal interleaving occurs in eight half bursts; on all other control channels (with the exception of RACH) interleaving is in four whole bursts. This is illustrated on the basis of an SDCCH (see Figure 3.23) and an FACCH/F (see Figure 3.24).

For example, at a transmission rate of 9.2 kbit/s, every 20 ms 184 bits are transmitted on an FACCH. For the detection of non-correctable errors 40 parity check bits calculated according to the Fire code are added by the physical layer. All bits are then protected by a convolutional code of ratio

$r = 1/2$. This provides sufficient error correction protection on the receiver side. The resulting 456 bits ($= 4 \cdot 117$) are then distributed over the four time slots of a logical channel. An interleaving of 4 is continuously applied to the signalling channels because messages are too short and otherwise too many slots would be occupied.

3.3.7.2 Speech Coding for Radio Transmission

In line with GSM goals on frequency economy, the transmission rate was not to exceed a value of approx. 16 kbit/s. The speech codec RPE-LTP was selected. It is based on the LPC technique (*linear predictive coding*) and incorporates a combination of *long-term prediction* (LTP) and *regular pulse excitation* (RPE). The codec has a net bit rate of 13.0 kbit/s. Together with the redundancy for error protection, 22.8 kbit/s (gross bit rate) are transmitted on the radio channel. Speech is produced in packets of 20 ms each from the speech codec. With consideration given to the channel characteristics and the possible bit error ratio of the speech codec parameters, the following three-stage channel coding method was selected (see Figure 3.26 (a)):

Ordering of the bits according to importance The bits of the speech codec parameters are arranged in descending order of importance, with the most important bit, No. 1, first and the least important bit, No. 260, last.

Cyclic redundancy check (CRC) for the most important 50 bits (Class 1a) These are extended by three parity bits as a *cyclic redundancy check* (CRC). The CRC bits are inserted after bit No. 50, and the bits that follow are renumbered. The associated generator polynomial is $g(x) = x^3 + x + 1$ and is implemented through a cyclic shift register with appropriate taps. On the receiving side these CRC bits are used for error detection and can sometimes indicate one or more errors within the most important bits.

Convolutional coding of the most important 185 bits (Class 1a+1b) for error correction Four tail bits (0) are attached to the first 185 bits, and these 189 bits are then recoded to 378 bits using a convolutional coding with ratio r =1/2. Since this protects the first and last bits better than it does the bits in the middle of a sequence, the first 185 bits are rearranged further so that the most important bit ends up first, the second most important bit is last (185), the third most important bit is second, and so forth. The remaining 78 bits of the speech codec parameters (Class 2) are transmitted without protection so that altogether 456 bits are produced per speech coding frame and transmitted with the interleaving factor 8.

With interleaving (distribution of data over several time slots of a logical channel), transmission errors occurring in bursts are distributed more

evenly over the speech data received, and transmission errors occur as single bit errors and therefore can be corrected.

Normal bursts (see Figure 3.7) with speech data are divided into two separate blocks, each with 57 bits and one control bit, which transport data of different (consecutive) speech coder frames so that eight sequential time slots contain the data from a speech coder frame of 20 ms duration and in addition the same amount of data from a second speech coder frame. Four consecutive time slots carry 456 user bits in 18.5 ms, namely the data output generated every 20 ms by the speech codec. The control bit indicates whether a block is carrying user data or data from the fast-associated control channel (FACCH).

3.3.7.3 Enhanced full rate (EFR) speech coder

The EFR speech coder is an Algebraic Codebook Excited Linear Predictive (ACELP) coder that has been introduced into products since 1998.

It produces 244 code bits for every 20 ms long speech frame, thereby corresponding to a bit rate of 12.2 kbit/s. These code bits are divided into three main categories upon their relative subjective importance. The most important 65 bits are protected by a half-rate convolutional encoder and an eight-bit cyclic redundancy checksum. If any of these bits are in error, the decoder will reject the speech frame as being in error. The following 117 bits are protected by the same half-rate convolutional code, whereas the remaining 78 bits are not offered any protection at all.

3.3.7.4 Discontinuous Voice Transmission

As an option, GSM permits the use of *discontinuous transmission* (DTX) for the transmission of voice signals, thereby reducing the amount of power handsets need in order to transmit and co-channel interference. Voice pauses are recognized by a function referred to as *voice activity detection* (VAD) and transmission is halted until data is again available. The traffic channel remains assigned to the connection.

To give the receiver the impression that a connection still exists during transmission pauses, the receiver side produces something called *comfort noise*, which is provided by the measurement results of the previous voice transmission phase and is activated by the transmission of updated measurement values from the transmitter during long voice pauses.

3.3.7.5 Adaptive Equalization of the Radio Signal

The 26 bits in the middle of each time slot (training sequence) are used by the receiver in order to adjust the parameters of its matched filter and to be able to select the path with the strongest signal in multipath propagation of the radio waves in its equalizer.

In the GSM frequency band, shadowing may happen to the radio signals of mobile and base stations; this is partly compensated by the fact that radio

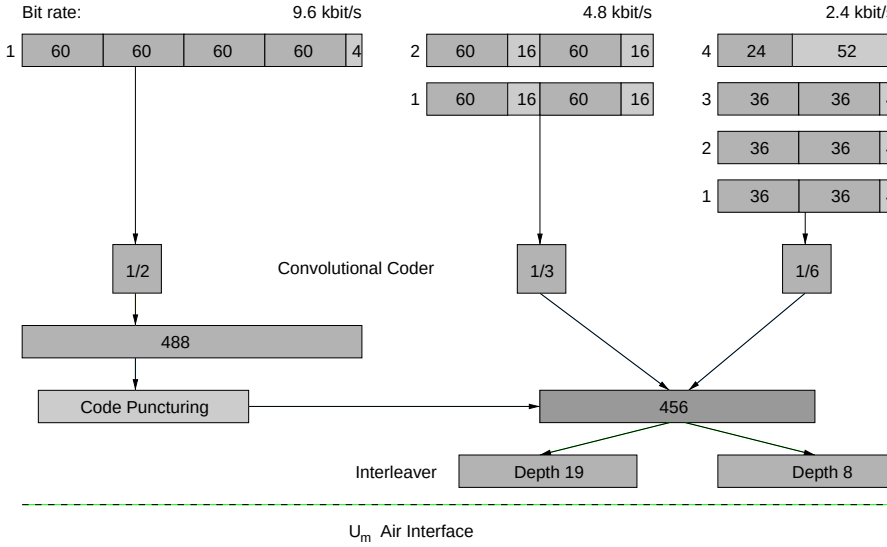


Figure 3.27: Forward-error correction of subscriber data with the different GSM bearer services

waves are reflected by buildings, mountains, inshore waters, etc., and fill in shadowed areas. The radio path of reflected waves is longer than the direct path. With the bit rates selected for GSM this difference in path lengths can amount to several bit durations (3.7 µs). Equalization of superimposed multipath signals by the receiver can compensate for delay differences of up to 16 µs (3 km difference in path length).

The communications signal is recovered when an estimation of the impulse response of the radio channel is carried out per time slot based on the training sequence contained in the received burst, and the receive signal is passed through an appropriate parameterized matched filter. The training sequence with the known information it contains is also used for determining the current quality of a radio channel through counting the bit errors.

3.3.7.6 Coding for Signalling and Data Transmission

Because longer transmission times are acceptable, interleaving factors of up to 19 are used for bearer services in data transmission (see Figure 3.27).

3.3.7.7 Throughput and Delay

Figure 3.28 shows the transmission rates (throughput) and delay times of all the logical channels (see Table 3.12). The *delay* results from the respective periodic recurrence (*recurrence interval*) of the time channel plus a period

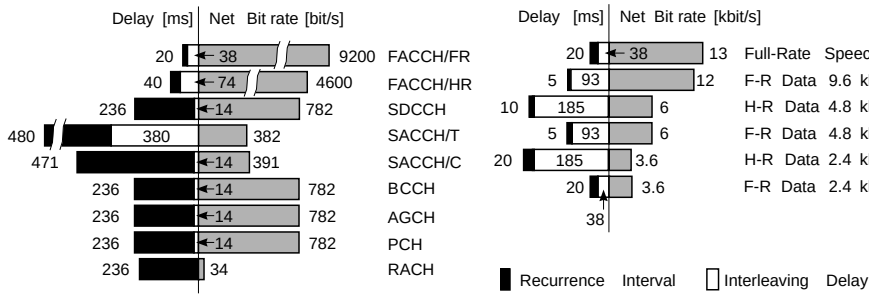


Figure 3.28: Logical channel throughput and delay times

corresponding to the selected interleaving depth (e.g., 14 ms with a large number of signalling channels when there is multiple interleaving).

3.3.7.8 Synchronization

If a mobile station wishes to make contact with the network through a base station, it must first detect the control channel (BCCH). The time slot and TDMA frame synchronization will then be established. The frequency of the mobile station's oscillator is automatically matched to the carrier frequency of the receive signal.

Recognition of the control channel (BCCH) The BCCH reserves one time slot (TN=0) per TDMA frame on a fixed carrier frequency. An important feature is that, on the frequency on which the BCCH is being sent, transmission is continuous with constant power (see also [16]).

In contrast, on other frequencies it is possible to transmit in each time slot with power attenuated to the individual connection. On other frequencies, in particular, the sender in unoccupied time slots is switched off. Frequency hopping is not used on the BCCH because this method must first be set up by the system information, which is sent on the BCCH.

If a mobile station is switched on, it conducts measurements of the mean signal level on all the frequencies known to it. A preliminary selection is made on the basis of the strongest mean receiving field strength. The BCCH can only be identified with satisfactory certainty if the *frequency correction burst* (FCB) and the *synchronization burst* (SCB), which must be on the same physical channel as the BCCH, are detected. The FCB is used for frequency synchronization, and the SCB is needed for establishing the time slot and TDMA frame synchronizations.

Time Synchronization Time synchronization means that the mobile station is intercepting the frequency channel at the appropriate time and is sending a burst at the appropriate moment. This requires that all logical channel

Table 3.9: Time synchronization using counters

Counter	Abbreviation	Value range
Quarter-bit number	QN	0–624
Bit number	BN	0–156
Time slot number	TN	0–7
TDMA frame number	FN	0–2 715 647

Table 3.10: System parameters `BS_CCCH_SDCCH_COMB`, `BS_CC_CHANS` and `CCCH_CONF`

`CCCH_CONF`	`BS_CC_CHANS`	`BS_CCCH_SDCCH_COMB`
`000`	1	`FALSE`
`001`	1	`TRUE`
`010`	2	`FALSE`
`100`	3	`FALSE`
`110`	4	`FALSE`

combinations of all eight physical channels of a frequency as well as the uplink and the downlink be time-synchronized with respect to the communicating stations (BTS and MSs).

This synchronization is based on the frame hierarchy of the different frame types presented in Section 3.3.4. Each mobile station has a set of counters that must run synchronously with the reference counters of the BSS. For each TDMA frame these counters produce a number that increases as time goes on. As long as a GSM-PLMN is in operation, the frame numbers continue to run; after a cycle of 2 715 648 the numbers start to repeat themselves. The counters for time synchronization are given in Table 3.9.

The quarter-bit number is derived from the training sequence. Because the SCH is always located on a physical channel of TN = 0, the mobile station sets the value for TN at 0 when it is receiving an SCB. The value of the frame number (FN) appears in coded form in the information bits of the SCB.

After the initial setting of these counters, the mobile station can use its own clock to update them. It increases the value of QN every 12/13 µs. The value of the bit number (BN) is produced from the integral part of QN/4. TN is incremented when QN changes from 624 to 0. FN goes up by one when TN increases from 7 to 0.

Recognition of the channel configuration of a cell If the BCCH is located and the mobile station's counter has been updated, the mobile station reads the system information in order to decode the channel structure available to this cell. The system information contains the parameter `CCCH_CONF` that establishes the number of physical channels (`BS_CC_CHANS`) supporting CC-

CHs. CCCH_CONF contains information on whether CCCHs and SDCCHs are combined (BS_CCCH_SDCCH_COMB = TRUE) or not combined (BS_CCCH_SDCCH-_COMB = FALSE) on a physical channel (see Table 3.10).

If CCCH_CONF takes on the value of 000 then combination 4 (see Figure 3.18) is supported and there is no combination 6 (see Figure 3.20). The value 001 means that only combination 5 (see Figure 3.19) is used in the cell. As far as the remaining three values are concerned, combination 4 is again available (see Figure 3.18), and, depending on the utilization level of the cell, is expanded by one, two or three combination 6s (see Figure 3.20).

The number of physical channels carrying CCHs is at the most four. Each CCCH is allocated its own group of mobile stations (CCCH_GROUP), which, when in idle mode, listen in on the PCH; there is a maximum of four radio paging groups [14].

The mobile station is now in a position to respond to a call or to initiate its own call. The logical channels that it needs, i.e., the CCCHs and the BCCH, are known to it.

3.3.7.9 Synchronization of Base Stations and Mobile Stations

The TDMA method requires that the signals of all mobile stations using the same carrier frequency reach the base station within the time slots assigned. They are not allowed to overlap. Because of the propagation speed of the signal at approximately 300 000 km/s, the synchronization reference signal of the base station reaches the mobile stations closest to it sooner than those further away. According to the GSM recommendation, a cell is allowed to be within a 35 km radius of the base station. The maximum loop duration from the base station to the edge of the cell and back is 0.23 ms for a 70 km stretch. If only the reference signal were being used for time synchronization, the time slots would have to contain the appropriate guard time; 40 % of the time slot would be wasted. In the GSM system the base station measures the loop duration and informs the mobile station by how much bit time before the reference signal it should be transmitting. This allows the guard time of each slot to be reduced to 30 µs (8.25 bits). This guard time is needed to allow the receiver to select the best out of a number of available radio paths of an incoming signal and still receive within the limits of a time slot.

3.3.8 GSM Layer 2: Data Link

In the ISO/OSI model this layer is responsible for the secure transmission of calls over individual subnets between two directly connected systems as well as for the error handling of data packets (see Section 2.7.4). The GSM specifications in regard to the *data link layer* are oriented on the ISDN standards, such as ISDN/LAPD based on X.200/Q.920, X.25/LAPB and HDLC/ISO 3309/4335. Some individual adaptations were made because, for example, unlike the LAPD protocol (*Link Access Procedure on the D-Channel*), no lim-

iting flags are required. The synchronization is already provided by layer 1, and because of the existence of several logical channels a special data link protocol had to be specified. Analogously to ISDN, this protocol is called $LAPD_m$. It is used between MS and BS, whereas between BTS and BSC over the A_{bis} interface the LAPD protocol is used, and between BSC and MSC over the A-interface the MTP protocol familiar from SS 7 is employed (see Figure 3.29).

3.4 Signalling Protocols in the GSM Data Link Layer

The communication protocols in the GSM system in some cases deviate from the ISO/OSI reference model (ITU-T recommendations X.200 to X.219). Although the protocols used in GSM can be structured on the ISO/OSI model, other protocol functions come into play because of the special characteristics of a cellular radio network. Therefore certain wide-ranging tasks, such as the evaluation and allocation of required capacity in a radio path and management-related services, do not necessarily only affect one particular ISO/OSI layer.

Because of the physical qualities of the transmission medium radio and the characteristics of GSM, suitable communication protocols are required. The signalling between the network elements (BSS, MSC, HLR, VLR, AuC, EIR and OMC) of the GSM system are based on *Common Channel Signalling System No. 7* (SS 7), ITU-T Series Q.700-795. A *mobile application part* (MAP) was developed specifically in order to accommodate the radio signalling in a GSM-PLMN. Figure 3.29 shows the architecture of the signalling protocols up to layer 3 and how they are distributed among the nodes of the GSM-PLMN.

The MAP part is implemented in all the switching centres directly linked to the mobile network. It consists of several *application service elements* (ASE), which are required for transactions involving registration and database inquiry and for determining a mobile subscriber's current location.

In addition to the MAP, the layered functions of CCS 7 also include the following components [32]:

- The *Transaction Capabilities Application Part* (TCAP) consists of two sublayers: a transaction sublayer and a component sublayer. The transaction sublayer is responsible for controlling the transactions or dialogues in a point-to-point connection. Each message is given a counter that is used to identify all messages at the other end of a connection. The component sublayer controls the different operations involved, such as a response or a notification that an operation has been completed successfully (or unsuccessfully). The TCAP supports the MAP; the services it provides correspond to those of ISO/OSI layer 7.

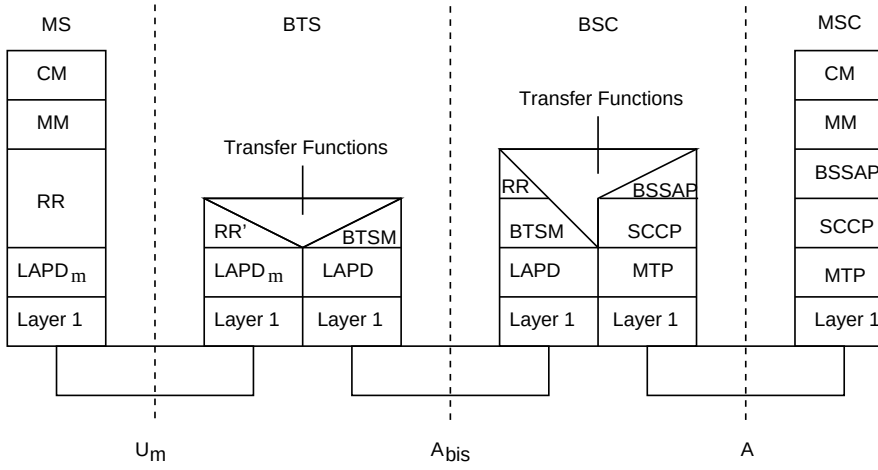


Figure 3.29: Architecture of the signalling protocols and distribution among GSM nodes

- The *Intermediate Service Part* (ISP) corresponds to ISO/OSI layers 4 to 6, and for the time being is still empty because these layers have no significance in a message-switched environment.
- The *Signalling Connection Control Part* (SCCP) represents a part of layer 3, and is used to set up a point-to-point connection for the transmission of individual messages.
- The *Message Transfer Part* (MTP) corresponds to layers 1 and 2 and part of layer 3, and is used to transmit messages between two nodes connected over a subnet. The MTP is divided into three sublayers, with the two lower sublayers fulfilling protective functions for message transfer and the third having responsibility for operating and maintenance functions in the network.

Radio resource management (RR) is supported to a limited degree in the base station and is implemented through the functions of *BTS management* (BTSM). In the BSC these functions correspond to the base station subsystem application part (BSSAP). As is shown, the entities of *mobility management* (MM) communicate directly between MS and MSC over an appropriate protocol. This establishes the service of *call management* (CM). The services CM, MM, BSSAP and SCCP fall under layer 3 (see Section 3.5).

The protocols of layers 1–3 defined in Series 04 and 08 of the GSM recommendations are required for signalling. The following sections will delve into the important characteristics of these three layers. Special emphasis will be given to two features of the LAPD_m protocol that are dictated by channel structure (*change of dedicated channel*) and the air interface (*contention resolution*).

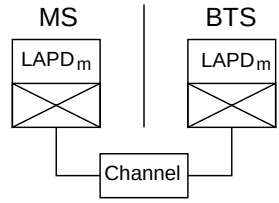


Figure 3.30: Position of signalling protocol LAPD_m

3.4.1 The LAPD_m Protocol

This section presents an overview of the functions and services of LAPD_m. Two special procedures, *contention resolution* and *change of dedicated channel*, are described in detail. Extensive information on the service primitives and procedures used can be found in [10, 11].

The LAPD_m protocol protects the data of the signalling protocols of the network layer and the short-message service on the radio channel. The participating partner entities are the MS and the BTS (or BSS) (see Figure 3.30). As is clear from the similarity in the names, the LAPD of ISDN was used as the basis for the development of LAPD_m.

3.4.1.1 Identification of Connection Endpoints

The data link layer provides its services to the network layer at a service access point (SAP) (see Section 2.5). An SAP has one or more *connection endpoints* (CE). Together an SAP and a CE form the *Data Link Connection Identifier* (`DLCI`).

The `DLCI` consists of:

Service access point identifier The *Service Access Point Identifier* (`SAPI`) appears binary-coded in the address field of each frame. The layer-3 entity determines which `SAPI` will be used by a layer-3 message.

Channel identification The `SAPI` determines which logical channel will be used for sending a message. The underlying channel structure with its existing time synchronization makes it possible for the channel identification to be managed locally in the terminal systems. It is transmitted as vertical control information between layer 3 and layer 2. This helps to keep the layer-2 protocol header short.

As can be seen in Figure 3.31, a message in layer 3 is clearly allocated to a logical channel. So far two different values for `SAPI` have been defined in the corresponding GSM recommendation 04.06 [11]:

- For the subprotocols *call control* (CC), *mobility management* (MM) and *radio resource management* (RR) of layer 3, `SAPI = 0` is applied.

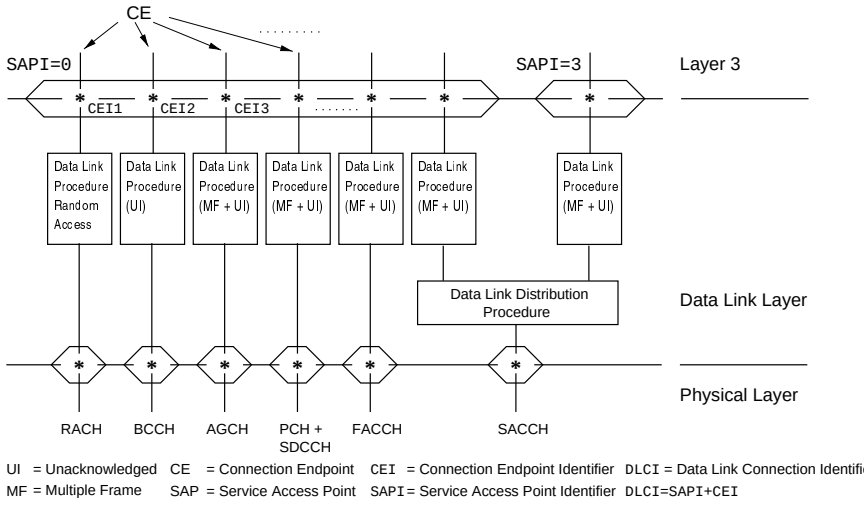


Figure 3.31: Allocation of layer-3 messages

- With messages of the *Short-Message Service* (SMS) an `SAPI = 3` is used. Consideration is being given to providing additional `SAPIs` for this service in the future.

`SAPI = 0` has a higher priority than `SAPI = 3`. Should other functions be added to the higher layers in the future, other `SAPIs` will be defined [10]. Coding possibilities are available for eight `SAPI` values.

3.4.1.2 Functions and Procedures

LAPD_m is used for transmitting signalling messages over the air interface between two layer-3 partner entities. LAPD_m procedures affect:

- several layer-3 entities
- several layer-1 entities
- signalling over the broadcast control channel (BCCH)
- signalling over the paging channel (PCH)
- signalling over the access grant channel (AGCH)
- signalling over dedicated control channels (DCCHs)

There are no procedures in LAPD_m to support the messages of the RACH. These messages are transported without any involvement of layer 2 between layers 3 and 1 of a subsystem (MS or BSS).

LAPD_m includes the following functions for implementing the services to layer 3:

- Operation of several layer-2 connections on different logical channels. These connections are differentiated from one another through the DLCI.
- Recognition of different types of frames.
- Transparent transport of layer-3 messages over the layer-2 connection between the layer-3 partner entities.
- Sequential control of the sequence of information (I)-frames.
- Error-handling in regard to format of frames and functional sequence.
- Notification of layer 3 when non-resolvable errors occur.
- Flow control.
- Resolution of competing situations on a dedicated channel after successful random access on the RACH (see Section 3.4.1.4).
- Change of a dedicated channel, without termination of a connection (see Section 3.4.1.4).

3.4.1.3 Service Primitives of LAPD$_m$

The service primitives are formed as follows:

DL_XX_req/ind/conf

Data from the higher layers is contained as parameters in the primitives DL_Data_req/ind, DL_Unit_Data_req/ind. GSM recommendation 04.06 includes an overview of all the data link layer service primitives.

3.4.1.4 Frame Types in LAPD$_m$

In addition to the frame types already presented in Section 2.7.4.1, there are other frame types that relate to the establishment and termination of a dedicated connection. These frame types are described to the extent that it is possible to understand the establishment and termination as well as the peculiarities of a dedicated connection. Details of the coding of *protocol data units* (PDU) and functional processes can be found in [10, 11]. Table 3.11 presents the different types of frames used in LAPD$_m$.

A frame can be either a command frame or a response frame. An I-frame is always a command frame. S-frames can occur as either command frames or response frames. SABM, DISC and UI are command frames, DM and UA response frames. If a layer-2 entity receives a command frame with a pollbit P = 1 then it must transmit a response frame with a finalbit F = 1. A command frame with P = 0 can be responded to by a command frame P = 0 or by a response frame with finalbit F = 0.

Table 3.11: Frame types in $LAPD_m$

Frame type	Function	Use
SABM	Set Asynchronous Balanced Mode	First frame necessary to enter confirmed mode
DISC	DISConnect	First frame necessary to exit confirmed mode
UA	Unnumbered Ack.	Confirmation of two preceding frames
DM	Disconnect Mode	Reply showing disconnected mode
UI	Unnumbered Information	Information frame in unconfirmed operation
I	Information	Information frame in confirmed operation
RR	Receive Ready	Proceed to transmit
RNR	Receive Not Ready	Stop transmission
REJ	REJect	Negative confirmation

Competition in setting up a dedicated connection As soon as an MS is allocated a dedicated channel, it attempts to use it to set up a layer-2 connection in acknowledged mode (see Section 3.4.1.5). To do so, the MS often sends an SABM to the BS for the `SAPI 0`.

In standard HDLC protocols (e.g., ISO 3309/4335/6159/6256) such as LAPD (see ITU-T Rec. I.441) an SABM frame is not allowed to transport information from a higher layer. In $LAPD_m$, however, it is essential that the SABM frame (and the UA frame) contains a signalling message for identification of the MS. This message is referred to as the *initial address message* (IAM) [13].

When an MS requests channel capacity for a dedicated connection using random access, an 8-bit-long Channel_req message is sent to the network. The content of this message is presented in Figure 3.32. It shows that 2 or 5 bits are available for the representation of a random number. At this point in time this random number is the only means by which the mobile station can be identified by the BSS. In response the BSS sends an Immediate_Assignment message [13] that contains the Channel_req information element along with the allocated dedicated channel. This message is transmitted on the point-to-multipoint channel AGCH. The mobile station for which the message element Channel_req from its random access burst and the message element from the response of the BSS are one and the same thing seizes the allocated channel.

With the length of the random number as it is indicated here, it can no longer be considered random. In highly loaded cells a situation will inevitably arise in which two mobile stations will send a Channel_req with the same content at the same time. The BSS's allocation of a dedicated control channel will be picked up by both mobile stations. This means that both will seize the same dedicated channel. Until one of the MSs clearly identifies itself (providing IMSI, TMSI, IMEI), it is not certain that only one of the two

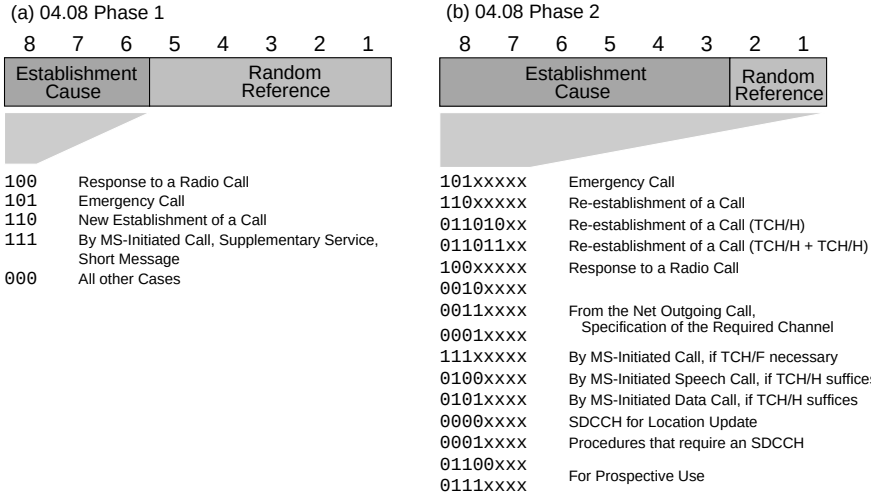


Figure 3.32: Content of Channel_req message

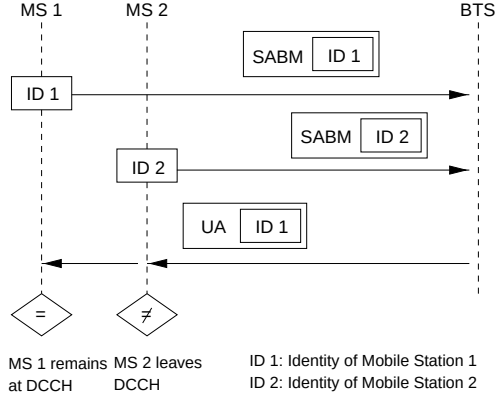


Figure 3.33: Contention resolution in call setup

mobile stations is seizing the channel. As a way of preventing these collisions LAPD_m offers a procedure called *contention resolution* (see Figure 3.33).

After the dedicated channel is activated, both mobile stations attempt to set up a bidirectional layer-2 connection in acknowledged mode. The identity (`IMSI`, `TMSI`, `IMEI`) of the mobile station is transmitted in the SABM frame. As a response frame to the SABM, the BSS must send back an UA response frame. The UA frame contains the identity of the first SABM frame received. Both mobile stations compare the identity contained in the UA frame with their own. If the identity matches, the respective mobile station changes to acknowledged mode (see Section 3.4.4). Messages from the higher layers

can then be transmitted bidirectionally and safely over the dedicated layer-2 connection. If the identities do not match the connection will be terminated.

Change of dedicated channel This procedure [11] only occurs on the signalling channels for `SAPI = 0`. A change of the dedicated channel is initiated by a BSS or an MSC. A dedicated channel is changed either at handover or during call setup if the first allocated dedicated channel was an SDCCH. In the first case the change is from one TCH to a new TCH in the same or in a different cell (intracell/intercell handover). In the second case the change is from an SDCCH to a TCH (nonOACSU/OACSU strategy); see Section 3.5.1.

If the request to change the channel is initiated by the BSS, the mobile station suspends the current dedicated channel (*suspension*). It stores the positions of the transmitting and receiving window of the I-frame that has not yet been acknowledged but should be transmitted next. The mobile station then sets up a connection on the new dedicated control channel and resumes with transmission (*resumption*).

3.4.1.5 Modes of Information Transmission

Basically two different types of LAPD_m operation are specified in GSM recommendation 04.05 [10]:

- *Unacknowledged operation*
- *Multiple frame operation*

The point-to-multipoint channels BCCH, PCH and AGCH only permit unacknowledged operation. The channels for a point-to-point connection, SDCCH, FACCH and SACCH, support both types of operation.

Unacknowledged operation The information of layer 3 is transported in UI frames. UI frames are not acknowledged. There are no control mechanisms such as timers or sequential numbering. Therefore no flow control or error correction takes place in this mode. This operating mode is available to all logical channels with the exception of the RACH.

Multiple frame operation The information of layer 3 is transmitted in I-frames. A send-and-wait logical link control protocol with window size 1 is used in this mode. The information frames are numbered sequentially modulo 8. Receipt of the I-frames must be acknowledged by the receiving side. This allows errors to be eliminated through the retransmission of the unacknowledged I-frames (error correction by repeated transmission). It also controls the flow of I-frames. If errors from layer 2 cannot be corrected, an error message is sent to layer 3.

3.4.1.6 Termination of a Layer-2 Connection

Multiple frame operation can be terminated in several different ways:

Normal release Layer 3 initiates a controlled termination of a layer-2 connection. The acknowledged connection between partner entities is terminated through an exchange of a DISC command frame and a UA response frame.

Local end release No exchange of command and response frames occurs. This termination is initiated and controlled by layer 3.

Abnormal local end release Here too there is no exchange of command and response frames. The command for this call termination comes from layer 3.

If layer 3 seizes the initiative for the termination of a connection, it communicates the type of termination using the appropriate primitives of layer 2 [11]. No termination mechanism exists in mode *unacknowledged operation*.

3.4.2 Services of the Physical Layer

The services required by layer 1 are specified in [9]:

- A layer-1 connection for the transparent transmission of frames is provided. In contrast to the LAPD protocol, which distinguishes individual layer-2 frames from one another through bit patterns (*flags*), in this case a fixed block format is provided.
- An indication is given whether a logical channel has been activated.
- Layer-2 frames are transmitted in the same sequence in which they arrived from layer 3.
- Synchronization to frame transmission.
- Use of channel coding procedures, as described in Section 3.3.7.1.
- Transmission (MS) and receipt (BTS) of random access bursts.

3.4.3 Influence of the Physical Layer on LAPD_m

The influence of the physical layer on LAPD_m is as follows:

- The error protection function in layer 2 is achieved through forward error correction in layer 1.
- The physical layer controls the transmission of layer-2 frames.
- The value of the control timer `T200` [11] is coordinated with the channel structure.

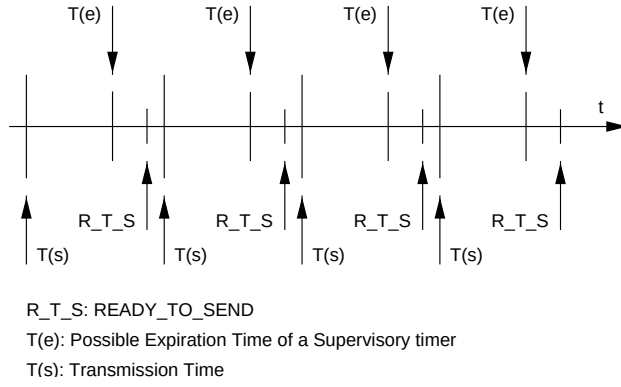


Figure 3.34: Time conditions with Ready_To_Send

3.4.3.1 Error Protection Functions of Layer 1

The channel coding measures covered in Section 3.3.7.1 protect data from destruction during transmission. LAPD_m does not add an additional CRC check sum to layer-3 messages. It depends on the Fire coder in layer 1. Consequently the error correction measures of LAPD_m are limited to block sequence control mechanisms and repeat requests of information frames.

3.4.3.2 Time Control of Layer 2

The physical layer does not provide queueing for the transmission of layer-2 messages waiting to be sent. Messages are never allowed to be buffered between an entity of layer 2 and layer 1, because the layer-2 entity must transmit a response or command frame appropriate to the momentary situation.

The physical layer in GSM uses the following primitive in order to inform the layer-2 entity above it when the next layer-2 block can be transmitted, adding a parameter indicating the logical channel [10]:

Ready_To_Send(`channel type`)

On one hand, this primitive must appear early enough to allow sufficient time for the formation of the next layer-2 frame. On the other hand, the time in which the control timer `T200` can run out must already have expired. Figure 3.34 shows the ideal timing.

3.4.3.3 Block Recurrence Time

The structure incorporating logical channels imposes transmitter bit timing on layer 2. This timing, called *block recurrence time*, depends on how frequently and at which intervals a logical channel will occur. The coder and interleaving techniques, as well as the size of the blocks arriving from layer 2, also have an effect. The block recurrence times

Table 3.12: Block recurrence times of the logical channels

Channel type	Net bit rate before layer 1 [kbit/s]	Block length [bit]	Block recurrence time [ms]
TCH/FS	13.0	$182 + 78$	20
TCH/HS			
TCH(9.6 kbit/s)	12.0	60	5
TCH(4.8 kbit/s)	6.0	60	10
TCH($\leq$2.4 kbit/s)	3.6	72	20
FACCH/F	9.2	184	20
FACCH/H	4.6	184	40
SDCCH	598/765 ($\simeq 0.782$)	184	3060/13 ($\simeq 235$)
SACCH/T	115/300 ($\simeq 0.383$)	$168 + 16$	480
SACCH/C	299/765 ($\simeq 0.391$)	$168 + 16$	6120/13 ($\simeq 471$)
BCCH	598/765 ($\simeq 0.782$)	184	3060/13 ($\simeq 235$)
AGCH	$n \cdot 598/765$ ($\simeq 0.782$)	184	3060/13 ($\simeq 235$)
PCH	$p \cdot 598/765$ ($\simeq 0.782$)	184	3060/13 ($\simeq 235$)
RACH	$r \cdot 26/765$ ($\simeq 0.034$)	8	3060/13 ($\simeq 235$)

- allow the calculation of the transmission rates of the logical channels of a physical channel, and
- allow the value of the control timer `T200` to be optimized in the LAPD_m protocol.

Table 3.12 presents an overview of the block recurrence times of the logical channels. The parameters n, p and r indicate the number of blocks in a block recurrence period and are specified by the network operator [13, 14].

3.4.3.4 Time-Critical Conditions

The LAPD_m data link protocol must be in exact agreement with the time conditions dictated by the channel structure. The following points must be taken into account:

1. Consideration of the block recurrence times of the logical channels.
2. The shortest anticipated response time for an I-frame, i.e., attention must be given to the time synchronization of the uplink and the downlink.
3. Timer values for frame control must be set on the FACCH in such a way that not too many speech data blocks are displaced.

Condition 2 reinforces the fact that blocks can be transmitted and received alternately on an SDCCH during a block recurrence period (see Figure 3.35). This explains why the control timer `T200` that monitors the blocks on the

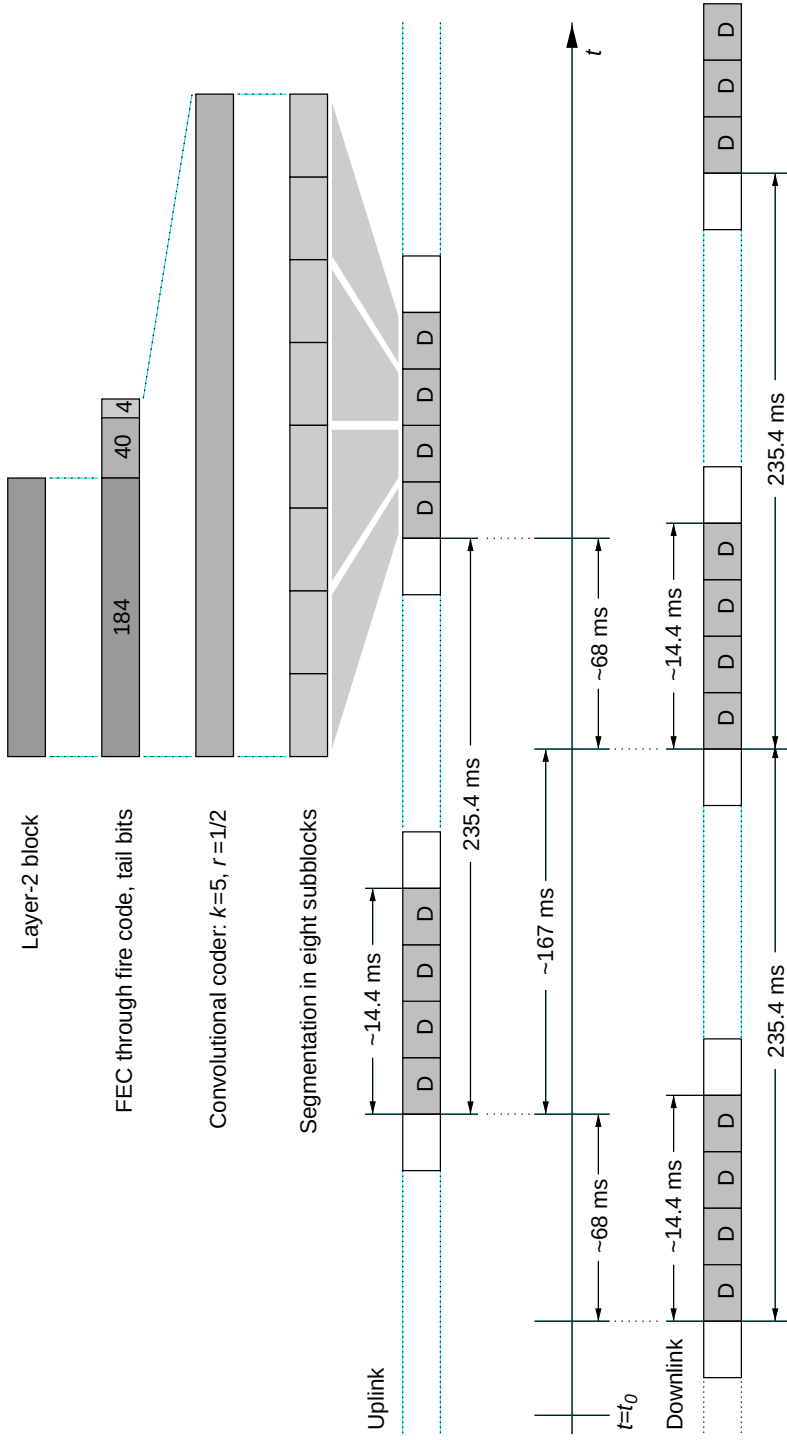


Figure 3.35: Time interleaving of uplink and downlink on an SDCCH

SDCCH is set at such short intervals that it runs out before the next transmission [11] and shortly after the expected arrival of a response frame. The value of a T200 on an SDCCH is 220 ms. Therefore it is definitely clear which frame type should be sent next before the next block recurrence time.

Condition 3 is important for the FACCH because the block recurrence time (see Table 3.12) is very short. An FACCH burst competes simultaneously with a burst carrying user data. In this case the value of the control timer T200 must be set large enough so that not too many user information data packets are displaced or delayed in succession. The value of a T200 on an FACCH is 150 ms.

3.4.4 $LAPD_m$ Services

The services of the physical layer are provided by the layer-1 services. Based on the two operating modes presented in Section 3.4.1.5, there are two different kinds of information services. Both are able to coexist in the same connection in layer 2.

Layer 2 does not provide any services that support random access. The parameters for the primitives DL_Random_Access_req/ind/conf and PH_Random_Access_req/ind/conf are transmitted in transparent mode between layers 1 and 3. A differentiation is made between four different services:

Priorities Two different layer-2 connections can be operated at the same time. But messages with SAPI = 0 are given higher priority than those with SAPI = 3.

Segmentation and reassembly In acknowledged operation layer-3 messages waiting to be sent and not fitting into a layer-2 information frame are segmented. When they arrive at the receiving layer-2 partner entity, the segments are reassembled.

Unacknowledged information transfer There is no acknowledgement of I-frames in this mode. However, it is possible for the higher layers to use similar acknowledgement procedures as $LAPD_m$ in acknowledged mode.

The characteristics of this service can be summarized as follows:

- A layer-2 connection is provided between two layer-3 entities for non-acknowledged mode.
- A unique identification of layer-2 connection endpoints is possible, thereby enabling a layer-3 entity to specify its partner entity.
- The priority of layer-3 messages is considered.
- There is no assurance that a message has arrived.

Acknowledged information transfer This service utilizes the acknowledged mode of operation. It has the following features:

- A layer-2 connection between two layer-3 entities is provided.
- A unique identification is provided for layer-2 connection endpoints, thereby enabling a layer-3 entity to specify its partner entity.
- The priority of layer-3 messages is considered.
- It provides an assurance that the correct sequence of successive I-frames will be maintained.
- Layer 3 is notified when sequencing errors or unresolved errors occur.
- Flow control.
- Segmentation and reassembly of layer-3 messages.
- Provision of procedures for *suspension/resumption* that ensure that no information is lost when there is a change of the dedicated control channel.

3.5 The Network Layer in GSM

The layer-3 signalling protocols provide the functions for establishing point-to-point connections between two mobile subscribers in a GSM-PLMN or between a mobile subscriber and a user of another network. If a connection has been established, it must then be maintained, even when there is radio channel interference. If one of the two mobile users wishes to terminate the connection, layer 3 ensures that the connection is terminated properly.

The network layer is divided into three separate sublayers, each offering their own clearly delineated functions. The specifications for the subprotocols of the network layer are found in [12, 13]. The lowest sublayer, RR, is based on the services of LAPD_m. The CC entity of the CM sublayer offers layer 4 the services of the network layer (see Figure 3.29). The individual sublayers carry out the following tasks:

Radio Resource Management (RR) Setup, maintenance and termination of a dedicated radio channel connection.

Mobility Management (MM) Registration, authentication and allocation of new `TMSI`.

Call Control (CC, in the CM sublayer) Setup, maintenance and termination of circuit-switched calls.

The CM sublayer contains three independent entities: CC (*Call Control*), SMS (*Short-Message Service*) and SS (*Supplementary Services*). The SS entity provides call-based and non-call-based services such as call diversion and billing. The SMS service permits short messages to be sent on the SDCCH and SACCH control channels. The call setup mechanisms will be discussed

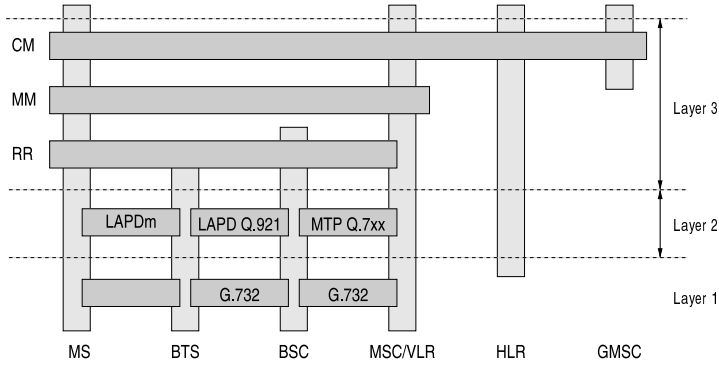


Figure 3.36: Network elements and related signalling protocols

later so that the emphasis here can be on the CC entity. The protocols for SMS and SS are specified in Recs. 04.10, 04.11 and 04.12.

A complete overview of the network elements involved in the various signalling protocols and on the range covered by the related layers and sublayers is given in Figure 3.36

3.5.1 Connection Establishment

A connection is established according to the message flow shown in the sequence chart of Figure 3.37. The following procedures have to be passed in chronological sequence:

1. Paging*.
2. Random access.
3. Assignment of SDCCH.
4. Definition of type of service.
5. Authentication.
6. Ciphering.
7. Assignment of a TCH.
8. Alerting at called terminal.
9. Connect after the called terminal is responding.

*With an outgoing call no paging by the BTS occurs, instead of the Paging_Response message a Service_Request message is sent containing the same information.

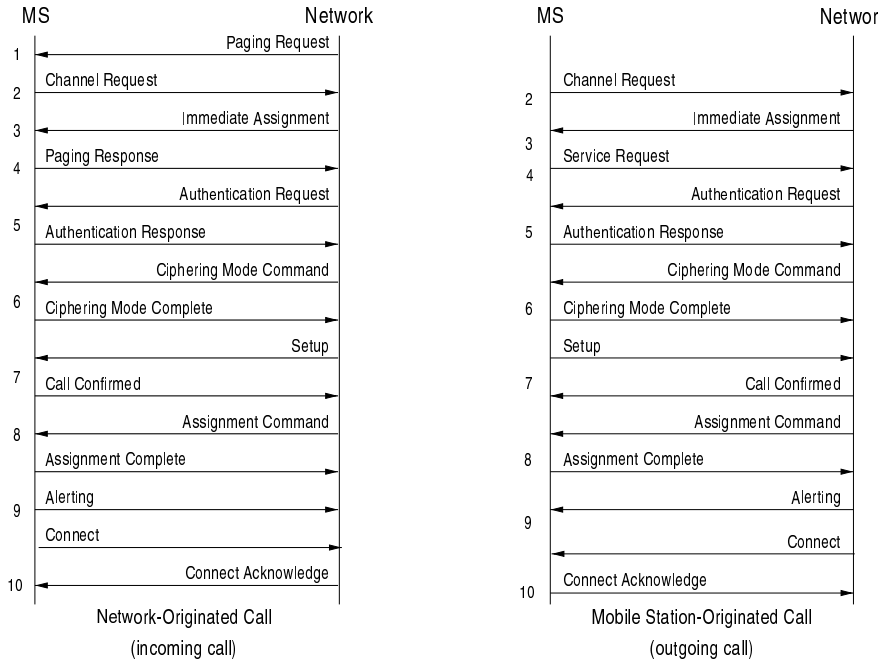


Figure 3.37: Connection establishment messages

To be able to make optimal use of the radio resources, three algorithms for connection establishment are contained in the GSM recommendations as alternatives:

- Off Air Call Setup (OACSU).
- Non-Off Air Call Setup (Non-OACSU).
- Very early assignment (VEA).

These alternatives differ in the time instant when a TCH is assigned to a call initiated by the mobile station (see Figure 3.38).

The VEA algorithm assigns the TCH for signalling directly after a random access message has been received at the BTS, and allows the fastest establishment of a connection, since the transmission rate of the TCH is much higher than that of the SDCCH. A disadvantage is the inefficient use of the TCH, since a percentage of outgoing calls is not answered by the called terminal and the TCH capacity is wasted then. Inefficient use of the TCH happens also when a location update is performed by using a TCH. According to the GSM recommendations, connection establishment in a cell must always apply the same algorithm. If the VEA algorithm is implemented, the resulting load to the TCHs from their use for signalling has to be taken into account when dimensioning the trunk of TCHs for a given traffic load of a cell.

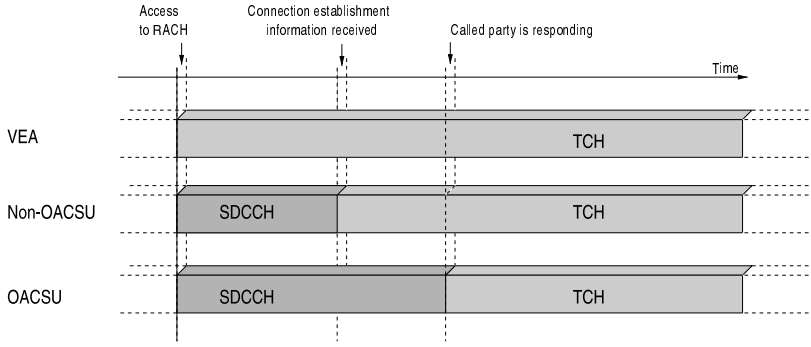


Figure 3.38: Alternate strategies for connection establishment

With the OACSU algorithm the TCH is assigned at the latest possible time instance, namely when the called terminal does respond. With the Non-OACSU algorithm the TCH is assigned as soon as the BSS has received the complete information to establish a call from the calling MS.

3.5.2 Services of the CC Sublayer

Simultaneous calls can be managed on different MM connections. The tasks of a CC entity is to provide the transport layer with a point-to-point connection between two physical subsystems. This connection is characterized by parameters relating to individual terminal equipment. For example, facsimile transmission in GSM requires coding and encryption measures different from those for the transmission of speech. These parameters must be set separately for each individual connection.

The term *call* describes a point-to-point connection that transmits data between two terminals via a channel. The point-to-point connection is set up and provided by the CC sublayer at the initiation of layer 4. The connection between terminals uses individual terminal-related data link control protocols (*bearer service*). The description that follows covers the CC entity within the CM sublayer that provides the parameters for setting up a connection with different types of terminals.

3.5.2.1 Call Establishment

Before a CC connection can be set up, an MM connection must be established. All parameters (Setup messages) that are available at the beginning of a call request are allocated to a circuit-switched connection. Calls initiated by a mobile station are differentiated as *basic calls* and *emergency calls*. With emergency calls the network may demand authentication and encryption; with basic calls these procedures are executed in a normal sequence.

A connection is initiated through the dialling of a number. The called terminal accepts the call with Connect. The calling terminal then acknowledges the Connect message with Connect_ack. This completes the call setup. The primitive MNCC_Setup_Compl_ind is sent from layer 3 to layer 4 in the subsystem network when the connection has been fully established.

3.5.2.2 Call Clearing

This is the termination of a point-to-point connection.

3.5.2.3 Change of Call-Related Parameters with an Existing Connection

These procedures are carried out during the active state of a call:

User notification This procedure allows the network to inform the mobile station of any occurrence affecting the call. Conversely, the mobile station is able to notify the terminal of the other side of the connection. An example would be the separation of the terminal from the network connection.

Change of parameter arrangements of a call (Call rearrangement) A rearrangement of parameters is necessary when a terminal attached to a mobile station is changed.

DTMF protocol control DTMF (*dual-tone multiple frequency*) is a digital set of tones produced by the terminal user pressing a key (0, ... 9, A, ... D, *, #). Sequences of DTMF tones can be used to operate voice mailboxes and answering machines. The control tones can only be used during the operation of the teleservices voice, voice/data transmission and voice/facsimile transmission.

In the network the DTMF tone is transmitted on the traffic channels (not over the signalling network). Between the mobile station and the network this tone is transmitted on the FACCH. The reason for this difference in transmission is the speech codec, which would distort the DTMF tones so severely that it would not be possible to maintain the quality requirements of other networks.

Change of channel parameters during a call (In-call modification) This service provides users with a point-to-point connection for the transmission of control data for different services in succession during the same call. Each change of service involves a change in the channel parameters, i.e., a change of channel or a modification of the parameters of the channel. The channel parameters of teleservices that are changed when they are being used are:

- Alternating between voice and transparent or non-transparent data transmission

- Change from voice to data transmission (transparent or non-transparent)
- Alternating between voice and facsimile transmission Group 3

The bearer services for this kind of teleservice must be switchable during a call. Therefore this service can only be accessed by special (*multiple capability*) mobile stations.

3.5.2.4 Other Procedures

The following procedures cannot be allocated to any of the above groups:

- Inband tones and announcements.
- Status enquiry. A CC entity has the possibility of querying the accuracy of the current state of a partner entity in the event of a fault. If it is established that the partner entity is in an incompatible state, the call is terminated.
- Call reestablishment.

Calls cannot collide. When the network and a mobile station initiate a call at the same time, the calls can be kept separate from one another through their different transaction identifiers `TI`; see Section 3.5.6.

3.5.3 Services of the MM Sublayer

The functions that support the mobility of a mobile terminal are in the MM sublayer. The network is alerted when a mobile station is switched on or off or when it leaves a location area. This requires a location update. The MM sublayer contains the functions for controlling the authorization of a mobile user (authentication procedure) and supplying a new code.

There are three groups of MM procedures.

3.5.3.1 General MM Procedures

These procedures can always be initiated if an RR connection exists. The procedures in this group include the following.

- Initiated by the network:

  Allocation of new TMSI (TMSI reallocation) The use of a `TMSI` instead of an `IMSI` ensures that the identity of the mobile user is kept confidential. A `TMSI` is only adequate for identification in an area that is allocated to a VLR. Outside this area it must be combined with a *location area identifier* (`LAI`). This procedure for the

allocation of a new TMSI must be carried out when a mobile user moves into a different VLR area (see Section 3.7).

The allocation of a new TMSI is usually carried out in conjunction with another procedure, for example during location updating or call setup.

Authentication Authentication serves two purposes. On the one hand, it provides a mechanism for checking whether the identity of the mobile station is valid. This can be implemented on the basis of the IMSI or, if it is available, a TMSI. On the other hand, it provides the mobile station with a new code.

Identification A mobile station receives a request from the network to send one of the identification parameters IMSI, TMSI or IMEI to the network.

- Initiated by the mobile station:

Removal of IMSI (IMSI Detach) This procedure is used when a SIM module is removed (detached) from a mobile station or a mobile station is switched off. The mobile station is then regarded by the network as unavailable.

3.5.3.2 Special MM Procedures

One of the following special procedures can only be initiated when no MM connection exists or when none of the other special procedures is active. Any general MM procedure except for *IMSI Detach* may be initiated during a special MM procedure.

Location updating This procedure updates the registration of the location of a mobile station. It can only be carried out if the mobile station contains a SIM. There is a reason for updating location when the LAI that the mobile station is receiving on the BCCH is different from the stored LAI. This occurs when a mobile station has moved into another location area or when it is switched on in an area whose LAI is not stored in the SIM. It therefore follows that only a mobile station can implement a location update.

Periodic updating Periodic updating only takes place when the appropriate parameter within system information is available on the BCCH.

IMSI Attach This is the opposite of the procedure *IMSI Detach*. After this procedure is carried out, a mobile station is identified as being available by the network.

All three procedures can be classed with the Location_Updating procedure in [13] (see Section 3.7). A special information element indicates which of the three procedures is being carried out.

3.5.3.3 Procedures for the Management of an MM Connection

These procedures are used for establishing, maintaining and terminating an MM connection between a mobile station and the network. The MM connection is for data exchange in the CC sublayer. More than one MM connection can be used at the same time. An MM connection can only be established if there is no MM procedure being run. There are three procedures:

- Establishing an MM connection, initiated by a mobile station. An MM connection may only be established if the location area has been successfully updated beforehand. Emergency calls are an exception.
- Establishing a connection, initiated by the network.
- Transmission of information about an MM connection. Each entity in the CM sublayer has its own MM connection. The messages of these entities are not given priority.

3.5.4 Services of the RR Sublayer

This sublayer contains procedures for the management of shared transmission resources. These include the physical channels and the layer-2 connections on the logical channels.

If an RR connection has been established, the RR sublayer provides the sublayers above it with a reliable RR connection. The duration of a point-to-point connection between two terminals normally exceeds the duration of a single RR connection. This means that there may be a change of dedicated channels during a connection, e.g., handover. In this context it is clearer to refer to an RR session during which an RR connection is suspended several times and re-established on another physical channel.

The dedicated channel is also changed during the setup of a connection if a non-OACSU strategy is pursued (see Section 3.4.1.4). In addition, the sublayer provides procedures for the transmission of messages on the BCCH and the CCCHs when no RR session exists. An example of this is automatic cell selection.

3.5.4.1 Idle Mode

When in idle mode, the mobile station is not allocated a dedicated channel but listens to the control channels BCCH and CCCH. In a subsystem a mobile station automatically changes cells when it moves beyond a cell boundary, and the higher sublayers are informed of this change. System information is passed on from the RR sublayer to the higher sublayers.

3.5.4.2 Setup and Termination of an RR Connection

When an RR connection is established, it consists of an active physical bidirectional point-to-point connection with the appropriate layer-2 connection in

acknowledged mode. This layer-2 connection is characterized as `SAPI = 0`. A mobile station is only able to maintain one RR connection. The MM sublayer gives the command for an RR connection to be established or terminated. A channel is activated when it is to be used for data.

3.5.4.3 RR Connection Established

In this operating state two dedicated channels are allocated to a mobile station. One of them is an SACCH, the other one is either an FACCH or a SDCCH. If an RR session has been set up, the RR sublayer provides the following services:

- Transmission of messages on all layer-2 connections. Layer-2 connections are characterized by their operating mode and the type of logical channel (see Figure 3.31).
- Establishment and termination of acknowledged mode in layer 2 connections with `SAPI = 3` (SMS).
- Establishment and termination of acknowledged mode on SDCCH, FACCH or SACCH.
- Suspension and resumption of data transmission during a change of dedicated channel or handover.
- Automatic cell selection and handover in order to maintain an RR session.
- A switch to transmission mode on a physical channel. During call setup this includes a change from *signalling only* to *user data* according to the VEA method; see Section 3.5.1. This can involve a change of channel type and coding and decoding mechanisms and an activation of the encryption process.
- Sequential control of messages in the CC and MM sublayers (see Section 3.5.5).

The RR sublayer uses the services of layer 2 and also has direct access to the services of layer 1. The latter includes automatic tracking of the BCCH and indicating whether continuous transmission is taking place on an SACCH. The tracking is for monitoring a connection.

3.5.5 Format and Coding of a Layer-3 Message

A layer-3 protocol data unit (PDU) consists of up to 249 octets, which are combined into one block. Each PDU consists of the following elements (see Figure 3.39):

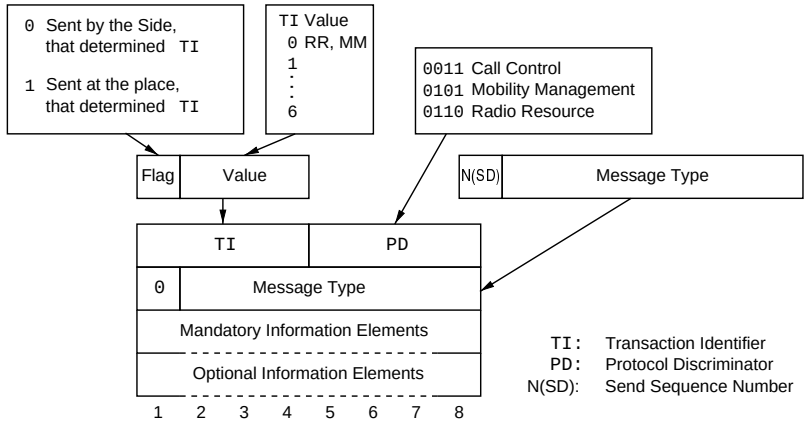


Figure 3.39: Establishment of a layer-3 message

Protocol discriminator (PD) The protocol discriminator indicates the subprotocol that has produced the message. It is the first part of a message and occupies the first four bits of the first octet. Figure 3.39 describes how it is coded. There is also a 4-bit code for SMS messages and SS messages that are not part of a call.

Transaction identifier (TI) The TI keeps transactions within a mobile station separate from one another. A transaction is a communication during which a particular service is used. Related to the CC protocol, this means that a TI is able to separate the signalling messages of different calls from one another. Current applications involving parallel transactions are *call waiting* and *call on hold*. Likewise a call on hold can be retrieved through *call retrieving*. In the future it will be possible to implement *multiple calls*. This will allow *conference calls* to be set up with participants being switched into a call or disconnected from it.

The TI corresponds to *call reference* of the ISDN layer-3 protocol. As shown in Figure 3.39, the TI forms the second part of every message. The TI is made up of the *TI-value* and the *TI-flag*. The *TI-value* is produced by the side that initiates the call. The side that is called takes over this value within the same call. MM and RR messages both contain the predefined value TI = 0.

The *TI-flag* prevents collisions between calls initiated at the same time by a mobile station and the network, and identifies which one has allocated the TI to the call. The side making the call always switches the *TI-Flag* to 1.

Message type The element message type consists of a coded name and a sequence number. The coded name identifies the function of a message.

A complete list of all message types that occur in a GSM network layer is presented in GSM Rec. 04.08. The sequence number N(SD) (here SD stands for Send Duplicated) is used to monitor MM and CM messages. It is incremented modulo 2.

MM and CM messages can be duplicated if there is a change of a dedicated control channel and the last layer-2 frame that contains an MM or CC message still has not been acknowledged by the layer-2 partner entity before the MS leaves the old channel.

Information elements: A message contains

- mandatory information
- optional information

Mandatory information elements are always transmitted before the optional elements. An optional information element must always be preceded by an *information element identifier* (`IEI`). This does not apply to mandatory information, because the existence and sequence of the information elements are clearly identified by the PD and the message type. In this case the `IEI` is removed from the information element.

If the length of an information element varies then its length will be marked by a *length indicator* (`LI`). In a mandatory information element the length is determined exactly by the `PD` and message type.

The information elements are divided into four types. The content of the information elements is abbreviated in Figure 3.40 with CIE (*Contents of Information Element*). An analysis of messages and error handling can be found in [13].

Type 1: Information element with half an octet of content

Type 2: Information element without content

Type 3: Fixed length information element containing at least 1 octet

Type 4: Information element of a variable length

3.5.6 Routing of Layer-3 messages

According to the directives of the OSI reference model, the header of an (N)-PDU is checked in an (N)-layer. Depending on the content of this header, the (N)-PDU is forwarded. If the content of the (N)-PDU is routed to the $(N+1)$-layer, the header is removed.

A method that does not comply with the ISO/OSI reference model is used for routing layer-3 messages to the correct sublayer. The advantage is that it reduces the number of octets per message.

This method is described based on the example of a mobile system subsystem. As is shown in Figure 3.41, the RR and the MM sublayers provide

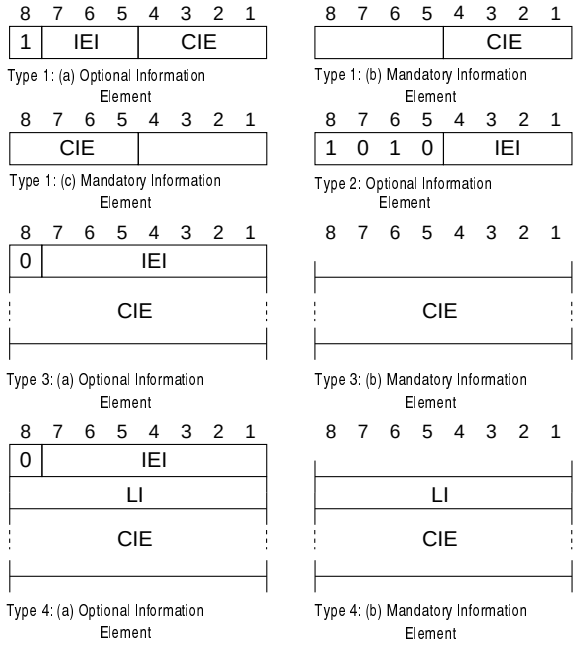


Figure 3.40: Optional and mandatory information element types 1 to 4

the routing and multiplexing functions. Messages arriving from an LAPD_m entity are routed according to their `PD`.

Messages with a `PD` value the same as RR remain within the RR sublayer, whereas all other messages are routed to the MM sublayer. Depending on its `PD` and the `TI`, a message is sent from the MM sublayer to the correct CM entity or it remains in the MM entity.

The MM sublayer forwards messages from the CM entities and its own messages to the RR sublayer in the uplink direction. In the process it multiplexes the parallel transactions of a call. The RR sublayer distributes the messages according to their `PD` and the updated channel configuration to the correct service access point (SAP) of layer 2.

3.5.7 Primitives of the Sublayers

The following primitives together with their parameters are described in GSM Rec. 04.07. The respective information elements are specified in GSM Rec. 04.08.

Primitives for the RR sublayer The primitives for the RR sublayer are formed according to the following rule:

RR_XX_req/ind/conf

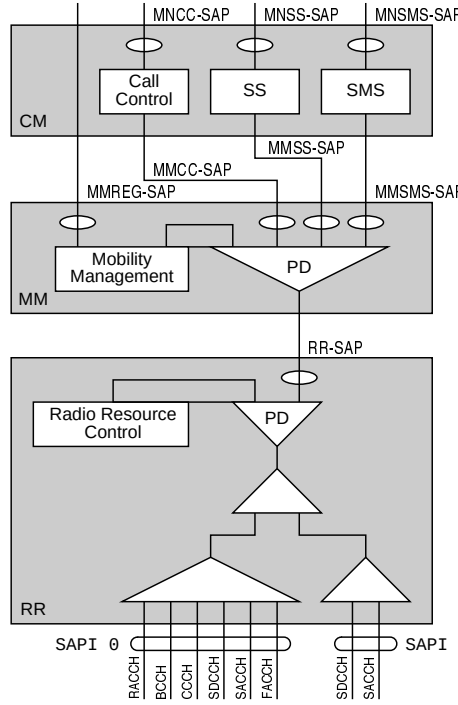


Figure 3.41: Routing of a layer-3 message

Primitives for the MM sublayer The primitives for the MM sublayer are formed according to the following rule:

MM_XX_req/ind/conf

Primitives for the CC sublayer The primitives for the CC sublayer are formed according to the following rule (MN = *mobile network*):

MN_XX_req/ind/conf

3.6 GSM Handover*

With new mobile radio networks an existing connection is automatically switched from one base station to another when the mobile subscriber moves to a different radio cell. GSM Rec. 05.08 even provides signalling functions for multiple *handover*. Handover always takes place when a mobile station leaves the area of a base station, and the MSC sometimes at the same time.

*With the collaboration of Martin Junius and Markus Scheibenbogen

3.6.1 Handover Causes

There can be many different reasons for a handover. Each mobile terminal attempts to use the radio channel that will provide the best connection quality, i.e., the best C/I (*carrier-to-interference ratio*). Co-channel interference is unavoidable because of multiple use of the same time and frequency channels due to existing cell layouts, and consequently quality can be poor (i.e., bit-error ratio high) despite a high signal level. The connection of a mobile terminal to the base stations can be the cause of interference to other mobile stations, even if it is a high-quality one. The interference can be minimized if the interfered station changes to a different radio channel. It is also possible for mobile users to have the same good receive quality from more than one cell. The service quality of the network can then be optimized if mobile users are equally distributed over the available cells.

It should be clarified that when co-channel interference occurs it is not the mobile stations that are causing direct interference to each other; instead the signals of the serving base station are superimposed on the mobile receiver by the signals of the co-channel base station and, in addition to the signals of the serviced MS, interference signals of one or more mobile stations from the cells of other systems are occurring at the receiver of the serving base station.

The possibility of being able to change cells during an active network connection without the connection being broken off is one of the most important functions of cellular networks in the support of mobile communications.

Section 3.6.3 deals with the principal handover variations and criteria. In addition, it provides a description of how, while actively connected to the GSM network, an MS produces a measurement protocol that is the basis for a handover decision by the network. Section 3.6.5 presents the handover criteria stipulated in GSM Rec. 05.08 [16] and an example of a suggested basic algorithm for a handover decision.

The GSM protocol for an intra-MSC handover is covered in Section 3.6.8. Section 3.6.9 explains the content of layer-3 handover messages.

3.6.2 GSM Recommendations

The recommendations relevant to the handover process and power control are:

03.03: *Numbering, Addressing and Identification* Reference [7] includes a description of the *Base Station Identity Code* (`BSIC`), which is contained in the measurement report.

03.09: *Handover Procedures* Reference [8] describes the signalling for the handover process in the network.

04.08: *Mobile Radio Interface Layer 3 Specification* Reference [13] describes the layer-3 functions *call control*, *mobility management* and *radio resource management*.

05.05: *Radio Transmission and Reception* Reference [15] includes a description of the power class of the MS and power levels.

05.08: *Radio Subsystem Link Control* Reference [16] describes handover, cell selection, power control, measurement reports and parameters, as well as a sample algorithm for handover and power control.

08.08: *Mobile Switching Centre to Base Station System Interface Layer-3 Specification* Reference [17] describes the *Base Station System Mobile Application Part* (BSSMAP) of signalling system CCS 7 and message formats, including signalling for handover.

GSM offers *mobile-assisted handover* (MAHO), which means that handover decisions are made on the basis of measurement reports received from an MS on its radio field situation.

3.6.3 Handover Preparation

A handover is based on a connection being measured and evaluated continuously by the respective base station and MS. The decision algorithm mainly contributes to the spectral efficiency of the radio network and the service quality as seen by the mobile subscriber. A mobile subscriber leaving the coverage area of a base station must receive coverage from a neighbouring base station in order to keep a connection intact. *Connection cut-off* or *call drop* are not acceptable to the mobile user during a conversation, and consequently it is very important to keep some quality of service level. Without automatic handover, the subscriber or the mobile station would be forced to establish a new connection. Handovers can also be initiated by the network when the traffic volume of a cell periodically reaches too high a level or when neighbouring cells are being underutilized.

3.6.3.1 Handover criteria

According to the different handover types described above, different criteria apply for the initiation of a handover. An effort is made to apply the measurement and evaluation of the receiving situation in the immediate past to the anticipated situation of the near future and on this basis to decide whether a handover seems appropriate. This extrapolation only offers a certain degree of probability of success, which depends on the criteria used and its evaluation.

Along with the signal level of the participating receivers, which can be used to estimate the path loss of a radio signal, the measured bit-error ratio is very important as an indication of the signal quality of the uplink and downlink. It is determined in the Viterbi decoder of the receiver through a comparison of certain received bit patterns with the known *training sequence* of each burst. In time-multiplexed systems time slots for transmitting bursts must be wider than the burst itself, so that it does not violate the time slot

boundaries because of the *round-trip signal propagation delay* of the signal from the base station to the MS and back. In GSM the round-trip delay is measured and corrected by the base station, so that the distance between the mobile and base station is known and a handover can be initiated in time in case the MS leaves the supply area (cell) and another suitable base station is available. Otherwise the connection must be cut off. As explained later, the base station control decides the time and type of handover. The MS transmits its measurement results to the BTS once or twice per second, depending on signalling volume.

Information on the signal quality of the uplink and the downlink of alternative neighbouring cells is not available during a connection. Therefore during the handover decision potential target cells are evaluated on the basis of signal level measurements undertaken by the mobile station of the broadcast control channels (BCCHs) of neighbouring cells, which work without power control but transmit continuously and consequently offer a constant signal level.

The GSM recommendation does not specify an algorithm for the handover decision or target cell selection. An example of a procedure that could be followed appears in the Appendix of GSM Rec. 05.08 (see Section 3.6.5). Network operators and manufacturers have the option of defining and using a handover algorithm based on the specified parameters.

The appropriate parameters are as follows:

- Permanent data, such as the transmitter power of:
  - the mobile station
  - the BTS of the supplying cell
  - the BTSs of neighbouring cells
- Results of real-time measurements carried out by the MS:
  - downlink signal quality (gross bit-error ratio)
  - downlink receive level of the current channel
  - downlink receive levels of neighbouring base stations (BCCHs)
- Results of real-time measurements carried out by the BTS:
  - uplink signal quality (gross bit-error ratio)
  - uplink receive level of the current channel
  - uplink receive level of neighbouring cells
- Traffic-oriented aspects (cell capacity, number of free channels, number of new connections waiting for a traffic channel in the BTS)

3.6.3.2 Measurement Protocol

Because of the complexity and importance of the parameters mentioned, a description of the *measurement report* is given below. The following explanations refer on the one hand to the information on measurement data sent by the MS to the network (*measurement reporting*), and on the other hand on the parameter measurements carried out in neighbouring cells by the MS (*neighbouring cells measurements*).

Measurement reporting So that the information required for a handover decision is available at all times, the MS measures system parameters on an almost continuous basis during a connection and sends this information to the network. This means that the MS is usually transmitting a measurement report to the network every 480 ms (block recurrence time of the SACCHs), but at a minimum once per second. It contains parameters that define not only the current network connection but also the radio conditions to neighbouring cells, which could possibly be used as target cells in a handover. The number of neighbouring cells monitored simultaneously by an MS depends on the network topology, and is stipulated by the network operator based on the cluster size. A maximum of six other cells are assumed in the measurement protocol.

An MS measures the receive level and quality of each downlink burst on the traffic channel allocated to its connection. Within a TCH multiframe (120 ms), measurement values for 24 TCH bursts and 1 SACCH burst are calculated; another time slot remains unused.

The interval between each individual measurement is 4.615 ms (duration of a TDMA frame), which corresponds to a route of approximately 4 cm at a speed of 50 km/h. Related to the duration of an SACCH block (480 ms $= 4 \cdot 120$ ms), this equates to 100 measurement values, the mean values of which are transmitted in the measurement report.

The measurement report is transmitted on the SACCH, which occurs every 120 ms and with an interleaving depth of 4 has a block recurrence time of 480 ms.

The SACCH only transmits control data, and is not dependent on the user data transmitted on a TCH. Discontinuous transmission on a TCH is only possible if control data is transmitted separately from user data so that the measurement report periodically expected by the network can be provided (see *Downlink DTX Flag* in Section 3.6.9.2). With a combined control and user data channel, discontinuous transmission would only be possible on a limited basis because of the measurement report, which is transmitted periodically.

Measurement of channels of neighbouring cells Measurement of the signal levels of neighbouring cells is more difficult because an MS must also establish which neighbouring cells it can even receive, and divide the possible measurement times among those cells capable of receiving.

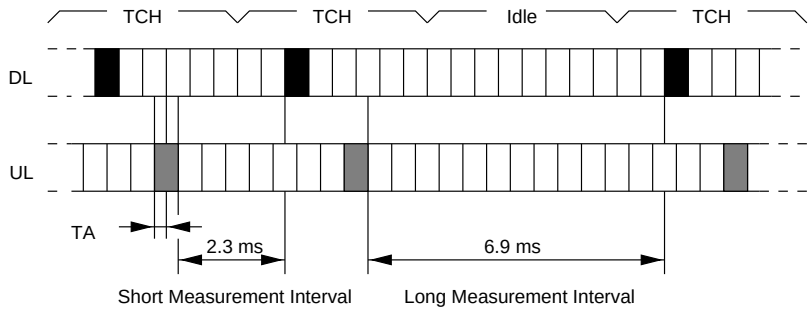


Figure 3.42: Measurement points for neighbouring cells

Thanks to the delay introduced between uplink and downlink slots of a traffic channel (see Figure 3.9), an MS can operate in half-duplex mode, and not only alternate the transmission of bursts in the appropriate time slots of the uplink and downlink frequency channel of its own active connection but also find out the channel characteristics of neighbouring cells in the unused time slots with the same receiver.

The period of time between the transmission of measurement reports varies, and depends on channel type. For example, the following applies to generating a measurement report in an MS with TACH/Fs (TACH = TCH + SACCH, F = full-rate channel).

The possible measurement times lie between transmission of a burst on the uplink (UL) and reception of a burst on the downlink (DL) of the traffic channel (see Figure 3.42). With a 26-frame multiframe (26 TDMA frames) of 120 ms duration, this equates to 24 time intervals of approximately 2.3 ms duration that are not suitable for the measurement of the channels of neighbouring cells. Furthermore, an interval of approximately 6.9 ms occurs because of an unused time slot in the 26-frame multiframe, and is used to seek out neighbouring cells capable of receiving.

The long interval is also used by the MS to find a suitable neighbouring cell. The criterion used is the BCCH bearer of the neighbouring cell, which is operated with no power control and continuous transmission (no *discontinuous transmission*) and therefore offers a constant signal level.

The length of these intervals determine what technical equipment is required by an MS so that it is able to carry out measurements not only in long intervals but also in short ones and in some cases on frequencies it is not using itself. The frequency synthesizer of the MS must be capable of changing sending and receiving frequency in less than 1 ms in order to have around 300 µs available per measurement.

An SACCH block has 100 measurement values divided among the suitable neighbouring cells. The measurement report transmits the signal level mean value of the six most suitable neighbouring cells.

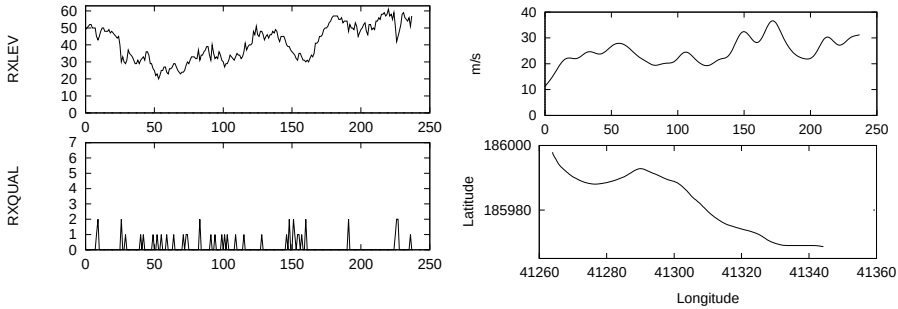


Figure 3.43: Measurement data: example 1

3.6.3.3 Real Measurement Data

Figures 3.43 and 3.44 show examples of mobile station measurement data recorded during two measurement trips in an existing network.

The downlink level (`RXLEV`) and the downlink quality (`RXQUAL`) can be seen in the two diagrams on the left of each figure; the speed of the MS (m/s) applied over the time (time intervals as multiples of the 480 ms SACCH frame duration) is shown in the diagram on the top right of each figure. The diagram on the bottom right of each figure shows the absolute position of the MS (longitude and latitude in arc seconds).

The two measurement data profiles clearly differ from one another. During the entire connection, measurement trip 1 was well supplied with a high signal level and in turn the quality of the connection was good. Measurement trip 2, on the other hand, shows a strong deterioration in signal level as the curve moves towards the end and this also reflects the very poor quality of the connection.

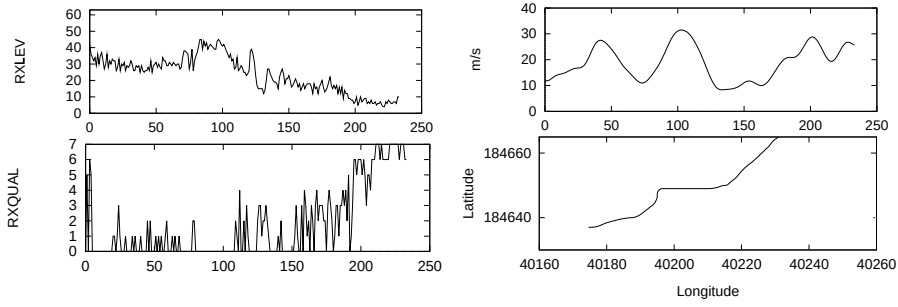


Figure 3.44: Measurement data: example 2

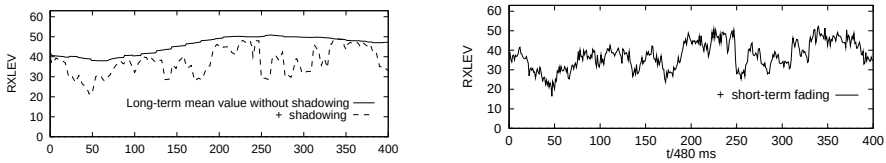


Figure 3.45: Long-term mean value, shadowing, short-term fading

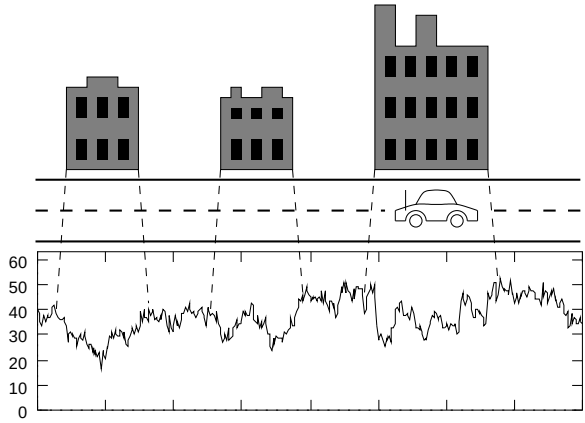


Figure 3.46: Level affected by topography

3.6.3.4 Analysis of Measurement Data

The measurement data clearly shows that the signal level characteristics can be divided into the following parts:

- Long-term mean value
- Short-term fading
- Shadowing

Figure 3.45 shows a superimposition of these components using simulation data. The short-term fading can no longer however be described here as Rayleigh or Rice fading but rather as an average calculated over 480 ms for a corresponding signal profile in GSM.

Figure 3.46 shows that the signal level profile is affected by shadowing and topographical circumstances.

3.6.4 Measurement Reports

In GSM *measurement reports* that are transmitted periodically from MS to BS on the SACCH assigned to each communication are available for each connection. The repetition duration of the SACCH produces a fixed time grid of 480 ms in which the measurement reports occur.

Table 3.13: Signal field strength RXLEV and signal quality RXQUAL

dBm			RXLEV
	...	−110	0
−110	...	−109	1
−109	...	−108	2
−108	...	−107	3
	⋮		⋮
−51	...	−50	60
−50	...	−49	61
−49	...	−48	62
−48	...		63

Bit error [%]			Average	RXQUAL
	...	0.2	0.14	0
0.2	...	0.4	0.28	1
0.4	...	0.8	0.57	2
0.8	...	1.6	1.13	3
1.6	...	3.2	2.26	4
3.2	...	6.4	4.53	5
6.4	...	12.8	9.05	6
12.8	...		18.10	7

3.6.4.1 Measurements

During a connection, the MS and the base station measure signal field strength and signal quality and the base station also measures the absolute distance to the MS.

Signal field strength From −110 dBm to −48 dBm the signal field strength is measured with relative accuracy within 1 dB. Absolute accuracy of 4 dB (up to −70 dBm) or 6 dB is required.

The signal field strength contained in a measurement report is an average value calculated over the duration of an SACCH multiframe. The following are measured:

- The field strength of the BCCH carrier of at least one of the known neighbouring stations in each TDMA frame. Four of the 104 TDMA frames can optionally be omitted.
- The field strength of all bursts in the allocated traffic channel.

This averaged data is transmitted in 64 steps as RXLEV parameters (see Table 3.13). The signal field strength coded as RXLEV is indicated below as the receive level or simply as level.

Signal quality Signal quality is measured as the bit-error ratio before channel decoding, and is divided into eight RXQUAL levels of quality (see Table 3.13). The required measurement accuracy and the allocation probability to the correct level is required to be 75 % for (RXQUAL 1) and 95 % to levels (RXQUAL 5–7).

The signal quality contained in the measurement report is an average value calculated over the duration of an SACCH multiframe. This value is indicated below as the receive quality or simply as quality.

Distance The absolute distance between mobile station and base station can be determined from the *timing advance* (TA) for the MS. This value is produced from the *round-trip propagation delay* of the burst sent by the MS relative to the time prescribed by the time slot and measured by the base station. It is calculated by the base station in multiples of the duration of a bit.

The distance of the MS is calculated from the timing advance as

$$d\mathtt{TA} = \frac{\mathtt{TA} \cdot ct_{\text{bit}}}{2} = \frac{\mathtt{TA} \cdot 3 \cdot 10^8 \text{ m/s} \cdot 3.69 \cdot 10^{-6}\text{s}}{2} = \mathtt{TA} \cdot 554\,\text{m}$$

c is the radio wave propagation speed, t_{bit} is the duration of a bit in GSM; the factor 1/2 is the result of the propagation time delay having to be calculated twice, namely as the round-trip propagation delay BTS–MS–BTS.

If a maximum error of ±0.5 bits is assumed for this measurement then the overall accuracy in establishing the distance is approximately 1 km. Since microcells can have a radius of ≤1 km, this value is too inaccurate, and in practical terms is only useful as a means to detect that the MS has definitively left the planned cell area.

Measurement values supplied While a connection is taking place with a traffic channel, the MS supplies the following values:

- Receive level RXLEV of the traffic channel.
- Receive quality RXQUAL of the traffic channel.
- Receive level RXLEV of the BCCH bearers of up to six neighbouring cells, specifically those with the highest receive level and a known and valid *base station identity code* (BSIC).
- Frequency of the BCCH bearer of these neighbouring cells.
- BSIC of these neighbouring cells. The BSIC is not a unique identification of the neighbouring base stations but a so-called colour code used to differentiate base stations transmitting at the same frequency in adjacent clusters.

3.6.4.2 SACCH Transmission

The measurement reports are transmitted over the SACCH as the Measurement Results information elements of a Measurement_Report message. An SACCH block has a length of 184 bits and the block recurrence time is 480 ms. The measurements carried out by an MS during a period of 480 ms are available to the base station for a further 480 ms. The time delay between an individual measurement and the receipt of the measurement data at the base station is also approximately 0.5–1 s.

Table 3.14: Measurement_Report message

Information element	Transmission direction	No. of octets
`Protocol Discriminator`	MS → network	2
`Transaction Identifier`	MS → network	2
`Message Type`	MS → network	2
`Measurement Results`	MS → network	16

A Measurement_Report message is structured as shown in Table 3.14 and Figure 3.47 (also see [13]).

The components of a `Measurement Results` information element are as follows:

- `DTX`: indication whether an MS is using discontinuous transmission.
- `RXLEV` Serving Cell: receive level of the supplying base station; the `RXLEV` levels from 0...64 described above are coded with 6 bits.
- `RXQUAL FULL` or `RXQUAL SUB`: receive quality of serving base station measured over all time slots or a subset. The eight `RXQUAL` levels are coded with 3 bits.
- # `NCELL`: number of measured neighbouring stations.
- `RXLEV NCELL` n: receive level of the neighbouring stations.
- `BCCH FREQ NCELL` n: BCCH frequency of the neighbouring stations, coded as an index in the BCCH channel list.
- `BSIC NCELL` n: `BSIC` of the neighbouring station as colour code. This always consists of 3 bits with the colour code of the PLMN and the base station [7].

3.6.5 Handover Decision

The handover process in GSM is implemented in the following steps:

1. *Measurement values*: Measurement and transmission of radio data as described.
2. *Handover request*: A handover request, if it is required, is generated on the basis of these measurement values.
3. *Handover decision*: A decision is made on this request.
4. *Handover*: The appropriate signalling and channel change on both the mobile station and fixed network sides finally take place.

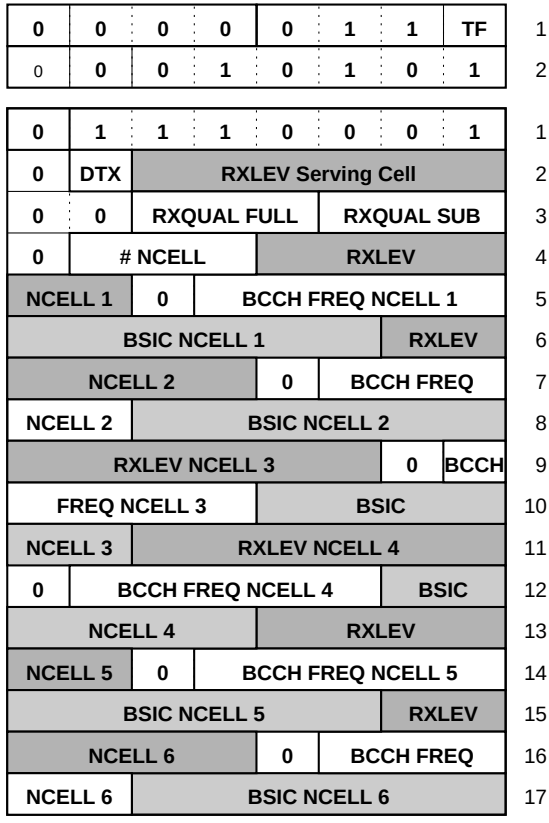


Figure 3.47: Components of `Measurement Results` information elements of a Measurement_Report message

In a *normal* handover process the handover request is generated by the BSC, and the handover decision and actual handover are the responsibility of the MSC. Depending on the type of handover carried out, functions 3 and 4 can be implemented in the BSC.

Criteria for a handover decision are given in Section 3.6.3.1. GSM Rec. 05.08 [16] only provides a selection of criteria, which are described in Section 3.6.5.1. There is no definitive handover algorithm, but the recommendation includes a basic algorithm, which is presented in Section 3.6.6.

3.6.5.1 Handover Request

As the solid basis for a handover decision by the controlling base station (BSS), each base transceiver station (BTS) and each mobile terminal generate measurement values on the receiving conditions of the current connection. These include the following:

- `RXLEV_UL`: Receive level (uplink), measured by BTS
- `RXQUAL_UL`:Signal quality (uplink), measured by BTS
- `RXLEV_DL`: Receive level (downlink), measured by MS
- `RXQUAL_DL`: Signal quality (downlink), measured by MS

The receive levels can be measured directly. The values for quality are calculated from the occuring bit error ratios and coded into eight levels of quality. The downlink measurement data is transmitted by the MS over the SACCH. The MS also measures the bearer signal `RXLEV_NCELL`(n) of the broadcast channel BCCH of the neighbouring cells, and likewise notifies the BTS over the SACCH. On the basis of this data, the BSC can decide whether a handover must be initiated and which of the adjacent cells are in a position to provide further radio coverage to the MS. The advantages of this method are as follows:

- Most of the measurement effort is undertaken by the MS.
- Abnormal radio propagation, radio shadowing, etc., are taken into account because the connection quality of both sides (uplink and downlink) is evaluated.
- The receive level of neighbouring cells is already measured by the MS before the handover takes place.

The following must be considered before a handover decision is made:

- The measurement results must be taken over an appropriate period of time in order to avoid a superfluous handover because of short-term signal level fluctuations.
- There must be a check to determine whether the connection quality can be affected by a change in transmitting power.
- The best radio cell is selected to ensure that the MS can also continue to be supplied with coverage if required.

The algorithm proposed in the GSM recommendations works on a relatively simple principle: several measurement reports are generated in a BSC and compared with threshold values. A handover is implemented if the quality values exceed the established threshold or the level values fall short of it.

The handover requests generated in the BSC are normally routed to the MSC. The signalling for a handover request from BSC to MSC is carried out through a Handover_Required message, the structure of which is shown in Table 3.15 [17]. The reason for the handover, a request for a response, a list of target cells sorted in order of best values and the current radio measurement values are included.

Table 3.15: Handover_Required message

Information element	Transmission direction	No. of octets
`Message Type`	BSS → MSC	1
`Cause`	BSS → MSC	3
`Response Request`	BSS → MSC	1
`Cell Identifier List pref.`	BSS → MSC	$2n+3 \ldots 7n+3$
`Current Radio Env.`	BSS → MSC	$15 \ldots n$
`Environment of BS` n	BSS → MSC	$7 \ldots n$

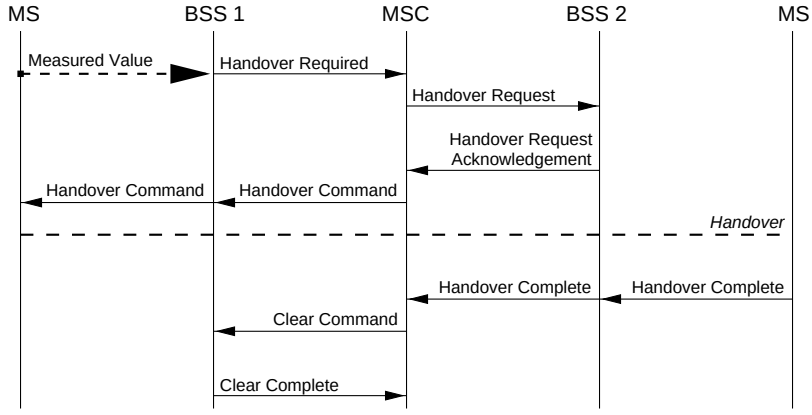


Figure 3.48: Handover (BSS 1 → BSS 2) executed with an MSC

Handover requests arriving from the BSS are processed in the MSC. The decision criteria include the availability of traffic channels in the target cell, the interference level in the target channel and other criteria relative to the network.

3.6.5.2 Handover Execution

A handover is initiated by an MSC, which first sends a Handover_Request message to request the necessary resources in the target cell (BSS 2) (see Figure 3.48). The target cell acknowledges with a Handover_Request_ack message. The MSC then sends a Handover_Command message to the previous cell (BSS 1), which in turn sends the same message to the MS. The MS then changes channel in the new target cell and sends a Handover_Complete message, which is also sent from the new cell (BSS 2) to the MSC. Finally the MSC sends a Clear_Command message to the old cell, which acknowledges the message with Clear_Complete.

This procedure applies to a normal handover signalling case. During actual operation, several different situations can occur:

Intracell handover The channel for the connection is changed within the cell, e.g., if the channel has a high level of interference. The change can apply to another frequency of the same cell or to another time slot of the same frequency.

Intercell/Intra-BSC handover In this case there is a change in radio channel between two cells that are served by the same BSC.

Inter-BSC/Intra-MSC handover A connection is changed between two cells that are served by different BSCs but operate in the area of the same MSC.

Inter-MSC handover A connection is changed between two cells that are in different MSC areas.

In the first two situations the handover can be executed by the BSC if it supports internal handovers. If this is not the case, intracell handover must also be carried out by the MSC.

3.6.5.3 Power Control

A reduction in transmitter power results in less co-channel and adjacent-channel interference, and reduces the power needed by mobile stations—which is especially important for handheld mobile equipment. Conversely, an increase in transmitter power makes it possible to compensate for short-term supply gaps, e.g., signal level fluctuations due to shadowing, thereby possibly preventing the need for handover.

Power control functions are analogous to the handover process in the BSC. Decisions on changing transmitter power are always made in the BSC. In turn, the measurement reports are the basis for the decision on whether transmitter power should be increased or decreased.

Like the handover algorithm, the algorithm for this process is not standardized. The basic algorithm in [16] also specifies power control.

3.6.6 Sample Algorithm GSM 05.08

Appendix A of GSM Rec. 05.08 [16] includes the specifications for a basic algorithm for handover and power control that is used to decide whether handover is necessary on the basis of measurement data provided by the MS and the fixed network. Network operators and manufacturers of GSM components use algorithms that range from a lightly modified or partially abbreviated version of 05.08 to heavily modified versions with new parameters, criteria, and so forth. The basic principles in terms of functions, however, correspond to the 05.08 algorithm.

3.6.6.1 Preliminary Steps and Threshold Comparison

The averaging of values and comparisons are generated every 480 ms; this time reference is dictated by the SACCH.

Averaging and power-budget value All measurement values (level, quality, distance MS–BS) are first averaged in order to reduce the effect of short-term signal fading. The averaging is influenced by the following parameters:

`HREQAVE`: Number of measurement reports averaged

`HREQT`: Number of averaged values in a Handover_Required message

Averaged values are provided for downlink and uplink level, downlink and uplink quality, and level of neighbouring cells, as well as distance MS–BS. The `HREQT` values that in turn represent an average value of `HREQAVE` measurement values are also indicated.

In addition, the *power budget* is calculated from the averaged values of the neighbouring stations:

$$\begin{aligned} \texttt{PBGT}(n) &= [\min(\texttt{MS_TXPWR_MAX}, P) - \texttt{RXLEV_DL} - \texttt{PWR_C_D}] \\ &\quad - [\min(\texttt{MS_TXPWR_MAX}(n), P) - \texttt{RXLEV_NCELL}(n)] \end{aligned} \tag{3.5}$$

This value represents the difference between the path loss on the downlink and the path loss anticipated in the respective neighbouring cell. Thus the value is greater than zero for neighbouring cells with better radio coverage and less than zero for those with worse coverage.

`MS_TXPWR_MAX` is the maximum allowable transmitter power for an MS in a current cell or in a neighbouring cell, `RXLEV_DL` is the averaged downlink level, `RXLEV_NCELL`(n) is the averaged level for a corresponding neighbouring cell, and `PWR_C_D` is the difference between the maximum allowable transmitter power and current transmitter power of a base station.

Threshold value comparison for power control The averaged values are always used in the threshold value comparison. The parameters are the threshold values and the `P` from `N` criteria. It is always necessary for the `P`n from `N`n values to exceed or fall short of the threshold if the transmitter power is to be increased or reduced, respectively. Table 3.16 summarizes the different posssibilities.

Threshold value comparison for handover The appropriate threshold values and `P` from `N` criteria are also used for initiating handover requests (see Table 3.17).

Figure 3.49 presents an overview of the threshold values. Figure 3.50 shows the behaviour of `PBGT` criteria during handover between two cells, the `HO_MARGIN` value represents an hysteresis value.

Table 3.16: GSM 05.08: threshold values for power control

Power control			
Transmitter power	Value	P/N	Threshold
+ DL	RXLEV_DL	P1/N1	< L_RXLEV_DL_P
+ UL	RXLEV_UL	P1/N1	< L_RXLEV_UL_P
− DL	RXLEV_DL	P2/N2	> U_RXLEV_DL_P
− UL	RXLEV_UL	P2/N2	> U_RXLEV_UL_P
+ DL	RXQUAL_DL	P3/N3	> L_RXQUAL_DL_P
+ UL	RXQUAL_UL	P3/N3	> L_RXQUAL_UL_P
− DL	RXQUAL_DL	P4/N4	< U_RXQUAL_DL_P
− UL	RXQUAL_UL	P4/N4	< U_RXQUAL_UL_P

+	Increase transmitter power	DL	Downlink
−	Reduce transmitter power	UL	Uplink

Table 3.17: GSM 05.08: threshold values for handover

Handover			
Cause	Value	P/N	Threshold
RXLEV DL	RXLEV_DL	P5/N5	< L_RXLEV_DL_H
RXLEV UL	RXLEV_UL	P5/N5	< L_RXLEV_UL_H
RXQUAL DL	RXQUAL_DL	P6/N6	> L_RXQUAL_DL_H
RXQUAL UL	RXQUAL_UL	P6/N6	> L_RXQUAL_UL_H
Intracell DL	RXLEV_DL	P7/N7	> RXLEV_DL_IH
	RXQUAL_DL		> L_RXQUAL_DL_H
Intracell UL	RXLEV_UL	P7/N7	> RXLEV_UL_IH
	RXQUAL_UL		> L_RXQUAL_UL_H
DISTANCE	Distance	P8/N8	> MS_RANGE_MAX
PBGT	PBGT(n)		> 0
			> HO_MARGIN(n)

3.6.6.2 BSS-Decision Algorithm

When a threshold value comparison establishes that handover is necessary, the BSC must send a Handover_Required message to the MSC. It contains a list of the neighbouring cells that have been evaluated, indicating the following conditions:

$$\begin{aligned} \texttt{RXLEV_NCELL}(n) &> \texttt{RXLEV_MIN}(n) \\ &\quad + \max(0, \texttt{MS_TXPWR_MAX}(n) - P) \end{aligned} \tag{3.6}$$

$$\texttt{PBGT}(n) > 0 \tag{3.7}$$

These conditions must be met by the respective neighbouring cell so that it can be considered as the target cell for the handover. The list is sorted on

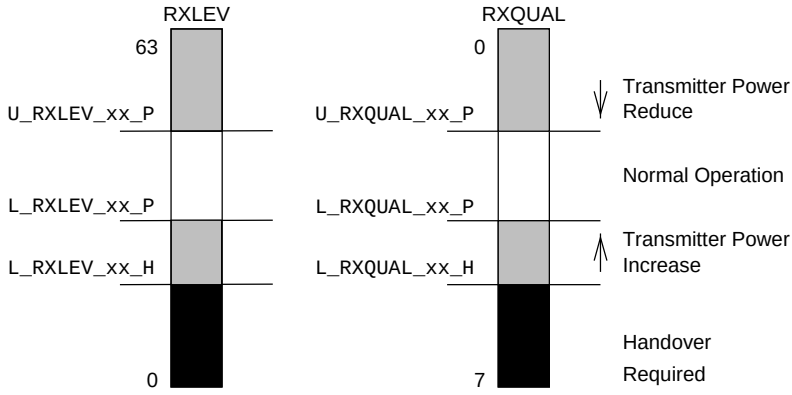


Figure 3.49: GSM 05.08 threshold values

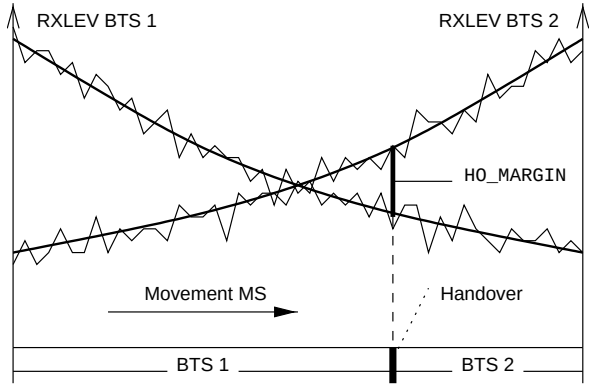


Figure 3.50: GSM 05.08 power budget handover

the basis of the PBGT value; the first cell is the one with the highest PBGT(n) value.

If a handover is considered to be *imperative* (the reason being level, quality or distance) then the condition (3.7) need not be met, and the list also contains neighbouring cells with PBGT(n) < 0.

An intracell handover is a special case, because two criteria are evaluated: if the signal level is good but the quality is bad, this usually means that co-channel interference exists. This can be avoided by a change in frequency or time channel.

When this is supported by a BSC, the intracell handover is carried out internally; otherwise a `Handover_Required` message must be sent to the MSC, with the serving base station at the top of the list.

3.6.6.3 MSC Decision Algorithm

The MSC evaluates arriving handover requests on the basis of the following priorities:

1. Quality
2. Signal level
3. Distance
4. Power budget

If a sufficient number of channels is not available in the respective target cell, handover requests made because of poor connection quality are given priority.

There is also a provision for giving priority to individual cells specifically to distribute traffic load. Therefore in a hierarchical cell layout the superimposed *macrocell* is only selected as a target for handover if a free channel is not available in the microcells.

3.6.6.4 Modifications

In practice the GSN 05.08 algorithm is used with modifications in GSM networks. Modifications are appropriate with averaging techniques for measurement values, in the administration of parameters by cell and with decision criteria.

Averaging The `P` from `N` decisions, based on averaged measurement values, in effect present another averaging technique specifically for threshold values. The dependence of the parameters makes optimization difficult. An option exists for simplifying this by introducing separate averaging parameters for each handover cause (quality, level, distance, power budget).

Otherwise modified averaging methods can be employed that, for example, do not consider outliers.

Cell administration The parameters for handover can be administered on a cell basis and thereby adapted optimally to local conditions. There is also an option of specifically switching on/off individual types of handover requests per cell.

Decision criteria The BSC decision criteria can also be changed, e.g., it can be practical to allow negative values to be used for `PBGT`(n) and `HO_MARGIN`.

3.6.7 Problems in the GSM Handover Process

As has already been mentioned, handover in GSM is based on radio measurement data. This is also what causes the main problem, because radio propagation in an actual environment is unpredictable and highly irregular.

In particular, shadowing caused by obstacles can produce some undesirable effects. This is mostly manifested by the fact that too many handovers are taking place.

3.6.7.1 Pingpong Handover

One very undesirable effect that occurs relatively frequently is so-called *pingpong handover*. This is a handover to a neighbouring cell that returns to the original cell after a short time. The cause of a pingpong handover is the power budget criterion. A handover is mainly executed on the basis of this criterion in cells with good radio coverage and only minimal disruption due to interference. The parameter `HO_MARGIN` determines which level of hysteresis must be exceeded so that a change to a neighbouring cell takes place (see Figure 3.50). In normal operation a 5–10 dB `HO_MARGIN` is selected to prevent minor variations in signal level of different base stations from causing a handover.

Strong shadowing created by large obstacles can cause fading up to 30 dB. If such an obstacle is found in the line-of-sight of the serving base station but not of the neighbouring station then it is possible that a handover may be triggered through power budget although the MS continues to find itself in a well-supplied cell. As soon as the MS moves out of the shadowing, the level again becomes normal and a handover takes place to the original cell.

With medium to higher mobility of an MS, this results in a handover to a neighbouring station and back to the original base station within a short period of time (<10 s) (*pingpong handover*).

This can be avoided if either the hysteresis value is increased so high that signal fading does not trigger a power budget handover or the averaging length is selected so that the time span in which an MS is in shadowing is bridged.

However, neither is appropriate in actual operation. A high hysteresis value would practically eliminate a power budget handover and delay a change in the boundary area of two cells far beyond the boundary. A high averaging length makes the handover process too sluggish and leads to a situation in which a really necessary handover sometimes is not carried out in time and the connection is then broken off.

This is why, with the 05.08 algorithm and the modified variants, pingpong handover cannot be avoided even by using appropriate parameter adjustments.

3.6.7.2 Number of Handovers

It was planned that the number of handovers per connection would be kept to a minimum in GSM because of the complexity of the signalling in the fixed network as well as from and to an MS. The traffic channel must be used for signalling in a handover between a mobile station and a base station, and this causes data loss and a corresponding forfeiture of voice quality.

It is therefore essential that unnecessary handovers be avoided. Control of transmitter power is optional for BTS and MS. Figure 3.51 shows a theoretical

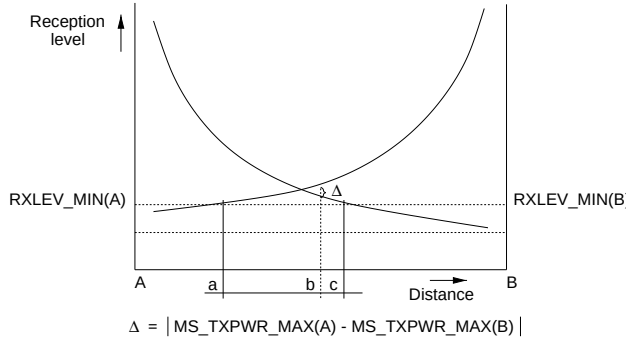


Figure 3.51: Recommended location for a handover during movement from base station A to base station B

profile of the average receive power during the movement of an MS from BTS A to B. A handover is theoretically only possible between locations a and c, but it is recommended at position b because that is where the level of BTS B has first fallen below the allowable hysteresis value Δ = `HO_MARGIN`, in other words is sufficiently above that of BTS A.

Owing to propagation-related signal level fluctuations at the receiver, the curve characteristics shown only apply to the statistical average (see the actual curve characteristics in Figure 3.50). Therefore the location of a handover is randomly distributed in the area around b. When an MS has poor reception and a poor parameter selection exists for the handover algorithm, several handovers instead of only one may be necessary before the MS leaves the area where reception is critical and a stable allocation to a BTS has taken place.

3.6.8 Intra-MSC Handover

Figure 3.52 shows the message sequence chart of the protocol for a handover between two different BSCs within the same MSC area (intra-MSC or inter-BSC handover) based on GSM Rec. 04.08 [13], Rec. 08.08 [17] and Rec. 08.58 [18].

The handover protocol can be divided into three separate parts [17]:

- Indication of a necessary handover: Handover_Required_ind
- Allocation of radio resources: Handover_Resource_Allocation
- Handover execution: Handover_Execution

3.6.8.1 Handover Required Ind

With Handover_Required_ind a BSS requests a handover for one of its active (*dedicated mode*) MSs.

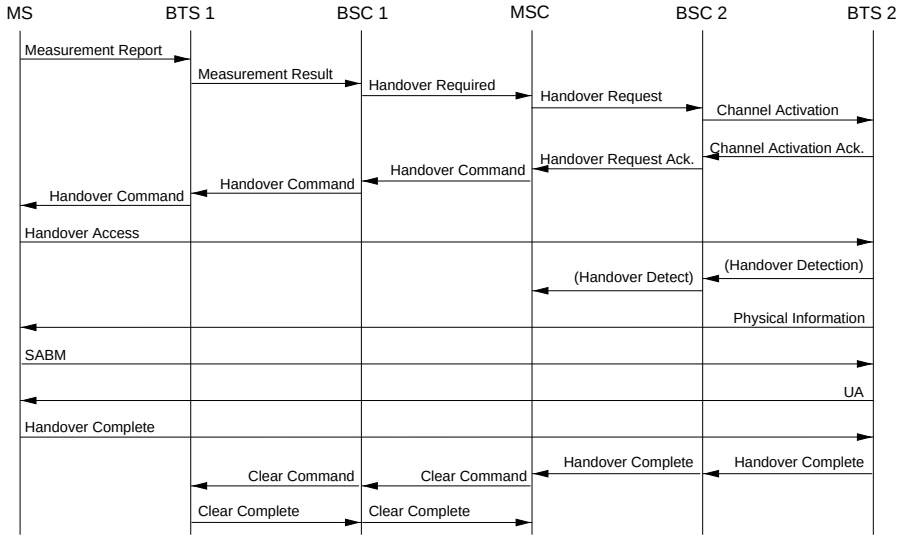


Figure 3.52: Intra-MSC handover in a GSM system

Based on the measurement protocol described in Sections 3.6.3.1 and 3.6.3.2 and a supplementary measurement of the corresponding MS uplink signal parameters by the BTS, the network decides on the necessity of a handover.

A handover decision is made by the BSC according to the criteria given in Section 3.6.5. If a handover is required for a certain MS, the BSC (BSC 1 in Figure 3.52) informs its higher-ranking MSC through a Handover_Required, at the same time providing the information needed regarding possible target cells and the identity of the original cell (*cell identifier list*) (see Section 3.6.9.1). The MSC then selects a suitable target cell and, if the cell happens to be in its area of responsibilty (*intra-MSC handover*), makes contact with the responsible BSC (BSC 2 in Figure 3.52); otherwise it forwards the handover request to the appropriate MSC (*inter-MSC handover*).

3.6.8.2 Handover Ressource Allocation

The process Handover_Ressource_Allocation allows an MSC to reserve suitable radio resources in a cell through the BSC.

If an MSC has been notified of a requirement for a handover through a Handover_Required, and if the target cell proposed in the message is located in its area of responsibility, then it sends a Handover_Request message to the BSC responsible for this cell. This message informs the BSC of the situation of the target cell and the characteristics required of the channel that is to be allocated to the MS. If a pure data or speech channel is involved, it provides the channel type used between the MSC and the BSC.

The channel type required for the radio interface can be different from the current channel type [17], e.g., immediately after a connection has been established in which the MS was only able to obtain a half-rate channel instead of the full-rate channel required. On this occasion a direct change of channel (*directed retry*) takes place.

The network operator can assign a priority to a Handover_Request depending on the urgency of the handover execution. Details of the content of a Handover_Request message can be found in Section 3.6.9.2.

A BSC selects Channel_Activation to request that the BTS responsible for the target cell check its radio resources on a specific available channel and provide it with a handover reference number (`Handover Reference`) that identifies the MS when it uses Handover_Access to access the channel. This is the only way in which a channel can be allocated to an MS without the identity of the user being divulged.

In addition to the channel characteristics required and the handover reference number, the BTS is given the technical parameters relating to transmission, e.g., for the power control of the new channel. The content of the Channel_Activation message is described in Section 3.6.9.3, whereas *power control* is explained in more detail in Section 3.6.9.5.

If the BTS is able to reserve a suitable channel for the MS, it confirms this to the BSC through a Channel_Activation_ack. The content of this message (see Section 3.6.9.4) also includes the current `Frame Number` of the BTS, which the BSC uses to determine the time displacement factor with which the MS begins its normal transmission activity after it has successfully accessed the channel. The MSC is then informed through Handover_Request_ack that a channel has been reserved. This completes the process Handover_Ressource_Allocation.

3.6.8.3 Handover Execution

For the MS the handover decision and initiation take place unnoticed in the background. As shown in Figure 3.52, it is first made aware of a handover when it receives the command Handover_Command.

The execution of a handover begins with a Handover_Request_ack message from the new BSC to the MSC that a requested channel has been reserved. It receives a data field `Layer 3 Information` with all the required information for the MS in order to register for the reserved channel with the new BTS. The MSC therefore forwards this data field unchanged through a Handover_Command to the original BTS, which sends it to the respective MS. The MS conveys the situation of the new cell only on the basis of the elements beacon frequency, contained in the data field, and `BSIC` (*Base Station Identity Code*, which provides a unique identification of the correct cell when several cells are using the same frequency for a BCCH or when there is a cell cluster with respect to the BCCH frequency; see [33]). The MS is not made aware of the total identity of the cell until the completion of handover over the SACCH of the new BTS.

Another important element in the data field is the handover reference number used by the new BTS to check whether the MS is authorized to receive the reserved channel. The MS notifies the BTS of the number using a Handover_Access message (see Section 3.6.9.7).

The Handover_Command indicates whether the MS should be executing an asynchronous or a synchronous handover. In both cases it is able to switch quickly to the frame and time slot clock of the new channel. An important difference in the procedure is the way in which the time advance parameter (TA) is determined.

With synchronous handover the MS itself calculates the parameter TA with which it begins radio transmission on the uplink. So that the BTS can specify the parameter for synchronization, the MS first sends some *access bursts* without time advance before it continues with normal transmission. This cancels out the need for a Physical_Information message, so that the MS goes straight to establishing a layer-2 connection to the network using an SABM frame.

With asynchronous handover the MS is dependent on the support of the network. In this case it sends *access bursts* so long until it receives a response from the network in the form of a Physical_Information message, the content of which allows it to calculate the time advance parameter (see Section 3.6.9.8). As with a synchronous handover, this is followed by the establishment of a layer-2 connection.

The timer T3124 monitors and controls the process of an asynchronous handover in an MS between the first Handover_Access and the arrival of the message Physical_Information. If the timer has expired, the handover process is broken off and the MS returns to its original channel. Since the FACCH is used to transmit the signalling data for a handover, the timer value according to GSM recommendation 04.08 [13] is 320 ms.

Immediately after receipt of the expected Handover_Access, the BTS informs its higher-ranking BSC through a Handover_Detection message that the reserved channel has been seized. With Handover_Detection the BSC thereupon informs the MSC, which then changes the connecting path to the MS through the network, sometimes not waiting for a Handover_Complete in order to reduce the switching times on the network side. After the MS has switched to normal transmitting (with time displacement), it establishes a layer-2 connection to the BTS using an SABM frame with an LAPD_m protocol. The BTS acknowledges the frame with a UA frame, etc.

As soon as the layer-2 connection has been acknowledged by the network, the MS informs the BTS through Handover_Complete that the handover has been successfully executed. This message is forwarded by the BSC to the MSC, which consequently reroutes the network connecting path to the new BSC. It uses a Clear_Command to release the reserved network resources at the original BSC and corresponding BTS. Consequently the MS can now only use the new radio channel. The BSC sends an acknowledgement Clear_Complete to the MSC after the resources have been released.

Table 3.18: Information elements of a Handover_Required message [17]

Information element	Transmission direction	No. of octets
`Message Type`	BSC → MSC	1
`Cause`	BSC → MSC	3–4
`Response Request`	BSC → MSC	1
`Cell Identifier List` (preferred)	BSC → MSC	$2n+3 \ldots 7n+3$

3.6.9 Intra-MSC Handover Protocol

The content of layer-3 messages from the intra-MSC handover protocol based on GSM Rec. 04.08 [13], 08.08 [17] and 08.58 [18] is introduced below. This section does not provide a complete description, but focusses only on the important elements of layer-3 messages. LAPD_m switching messages have already been discussed in Section 3.5.

3.6.9.1 Handover Required

If a BSC makes a positive handover decision for an MS that already has a dedicated channel, it indicates to the MSC with Handover_Required (see Table 3.18) that a handover is required for the MS concerned because of a particular reason (*cause element*).

`Message Type` This 8-bit (1-octet) long information element clearly identifies the type of message and is contained in all layer-3 messages. Section 3 of GSM Rec. 08.08 [17] particularly deals with message types used in the management of radio resources (*radio resource management*, RR).

`Cause` The `cause` element describes the reason for the occurrence of a particular message. It is based on a 3- to 4-octet long data field with the actual cause coded in the last or in the last two octets (see [17]).

The following are typical of the 40 possible causes of a handover:

- *Uplink quality*
- *Uplink strength*
- *Downlink quality*
- *Downlink strength*
- *Distance*
- *Better cell*
- *Response to MSC invocation*
- *O&M intervention*
- *Directed retry*

Table 3.19: Information elements of a Handover_Request message [17]

Information element	Transmission direction	No. of octets
`Message Type`	MSC → BSC	1
`Channel Type`	MSC → BSC	5
`Encryption Information`	MSC → BSC	3–20
`Classmark Information 1`[a]	MSC → BSC	2
`Classmark Information 2`	MSC → BSC	5
`Cell Identifier`	MSC → BSC	5–10
`Priority`	MSC → BSC	3
`Circuit Identity Code`[b]	MSC → BSC	3
`Downlink DTX Flag`[c]	MSC → BSC	2
`Cell Identifier` (target cell)	MSC → BSC	3–10
`Interference Band to be Used`	MSC → BSC	2
`Cause`	MSC → BSC	3–4
`Classmark Information 3`[d]	MSC → BSC	3–14

[a]Only one of the two elements is required.
[b]Only incorporated if the `Channel Type` element indicates the transmission of either speech or data.
[c]Only added if there are speech TCHs. If the indicator is omitted then it has no affect on the current DTX mode of the BSC.
[d]The MSC forwards this element only if it receives it itself.

`Response Request` This 1-octet long information element indicates that a BSC is waiting for a Handover_Required_Reject in case the Handover_Required message has not resulted in a handover.

`Cell Identifier List` This list contains all the target cells being considered for a handover. It is produced on the basis of the detailed measurement protocol in the BSC when a positive handover decision has been made according to a specific network-oriented algorithm. Depending on the number of cells (indicated as n in Table 3.18) and the coding used for cell identity, it can have different data field sizes. The number of target cells possible is generally determined by the operation subsystem (OSS) or by the responsible network operator, but, according to the GSM recommendations, should be in the area between 1 and 16. The `LAI` or the `CI` code (5 or 2 octets/cell) is normally used for cell identification in a GSM network. However, according to [17], other codes can also be selected.

3.6.9.2 Handover Request

An MSC sends this message to inform the respective BSC that the relevant target cell for a handover is located in its area of responsibility (see Table 3.19). The MSC provides the identification of the target cell in which the BSC is to reserve appropriate radio resources.

`Channel Type` This information element contains all the information on channel characteristics that the BSC requires in order to select the correct channel. The information includes application (voice, data or signalling), channel type and speech codec or data rate.

The following different channel types are appropriate [17]:

- Full-rate TCH
- Half-rate TCH for speech or data
- SDCCH

There is also some flexibility in the selection of channel type. For instance, if a half-rate or a full-rate channel is not available, the other channel type is allowed as an option. Furthermore, the BSC can be instructed to change the channel that has initially been allocated to the desired channel type as soon as it becomes available again. An exact breakdown of the different possibilities and the associated coding can be found in [17].

Similarly to ISDN, full-rate channels are referred to as B_m-channels and half-rate channels as L_m-channels. The control channels are called D_m-channels [13].

The following rates are available for the transmission of data in a tranparent bearer service: 9.6, 4.8, 2.4, 1.2, 0.6 kbit/s and 1200/75 bit/s (network → MS/MS → network).

The non-transparent services offer the following data rates [17]: 12 (B_m-channel) or 6 kbit/s (L_m- channel), which vary depending on available resources.

`Encryption Information` This information element includes the necessary ciphering information for control data transmission in order to set the target BTS to ciphering mode on the mobile side.

The basis for encryption is the ciphering key K_c mentioned in Section 3.13. `Encryption Information` notifies the BSC and the BTS of the ciphering key and the index of the coding algorithm used. In total there are up to seven different algorithms, however; it is also possible to select to transmit without ciphering. The information element consists of three fixed ciphering keys and one ciphering key up to 17 octets long.

`Classmark Information` Mobile stations differ from one another in a variety of different ways, for instance in their maximum transmitter power and in the bearer services they support. A network must be familiar with the characteristics of each active MS in order to avoid causing errors. Therefore the participating BSS network elements are informed of the characteristic features of the respective MS before each new connection is established.

The element `Classmark Information Type 2` is one of three types of data field, which (in contrast to the other field types) contains a complete list of

characteristics relating to the mobile terminal and is made up of the following elements [13]:

`Revision Level` This is the revision number for the mobile terminal, which characterizes the current stage of development through 3 bits. Terminal equipment that technologically is classified as belonging to the first phase of GSM standardization is identified by the code `000` [33].

`RF Power Capability` This indicates the maximum transmitter power of a terminal. As shown in Table 3.4, the transmitter power of equipment is specified according to its classification.

The appropriate information is required for the dynamic control of the transmitter power (*power control*) and in preparation for handover.

`Encryption Algorithm` This element, a 2-bit long code word with the value `00` [13] for the identification of terminal equipment with A5 encryption, is used to indicate to the network whether a terminal is using an encryption algorithm and, if so, which one. Only one alternative, algorithm A5, is planned for GSM phase 1. According to the currently valid MoU, terminals with the A5 algorithm have been introduced in all the participating countries. As indicated in `Encryption Element`, A5 can be produced as one of seven variants by the equipment manufacturer.

`Frequency Capability` This element enables the network to support terminal equipment in different frequency bands. A GSM-900 terminal currently must have the capability of being able to receive at least all the carrier frequencies in the 2×25 MHz band. It will not be possible for all terminal equipment to support the expansion to a 35 MHz band (EGSM band). Consequently a BSC must be able to distinguish between different types of equipment and allocate an appropriate channel based on the frequency area supported.

This element is a 3-bit long code, which has the setting `000` to characterize terminal equipment that is able to operate with 2×25 MHz wide bands.

`Short-Message Capability` This `Classmark Type 2` element is coded by one bit and relates to *short messages*. It indicates whether mobile side support exists for this bearer service (`1`) or not (`0`) [13]. `Classmark Type 1` represents an abbreviated version of the data field in order to map certain layer-3 messages, such as Location_Updating_req, to a single layer-2 frame, and only contains the most important elements [13]:

- `Revision level`
- `RF Power capability`
- `Encryption algorithm`

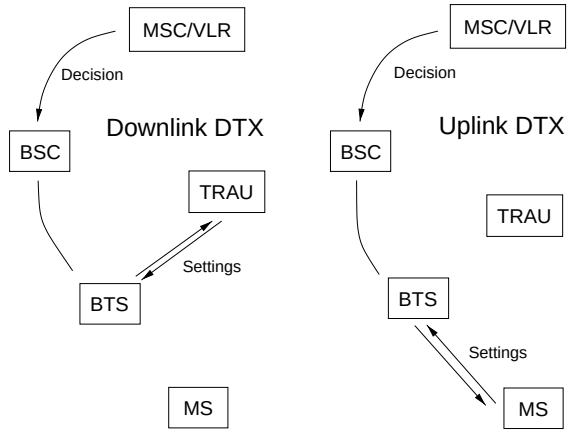


Figure 3.53: Process of discontinuous transmission

`Cell Identifier` From the *cell identifier list* presented in Section 3.6.9.1, the MSC selects a suitable target cell for a handover and forwards the corresponding identity code for the cell `(Cell Identifier, Target)` to the respective BSC. The MSC also provides the identity of the original cell `(Cell Identifier, Serving)` in which a handover decision has been made for a particular MS. The coding of the cell identity corresponds to the form given in the `Cell Identifier List`.

`Priority` This information element indicates the importance of message types specified for processing. In addition to 14 different levels of priority, existing connections or impending calls are classified in order of priority levels for the allocation of radio channels. What is also allowable is that if a channel group is momentarily occupied, a Handover_Request will not immediately be rejected but placed in a queue to be processed as quickly as possible.

`Circuit Identity Code` This information element defines PCM multiplex mode and the corresponding time slot of the fixed network channel provided by the MSC.

`Downlink DTX Flag` DTX indicates an optional transmission mode for speech and non-transparent data services in which background noise (*comfort noise*) is inserted during periods when minimal information is being transmitted. Since DTX can be carried out independently of the transmitting direction, consideration must be given to two DTX modes: uplink and downlink DTX.

DTX affects the operating behaviour of the MS, the TRAU (see Section 3.2.1.2) and the BTS, which adapts its behaviour dynamically to the

data transmission modes of the MS (uplink) and of the TRAU or MSC/IWF (downlink).

The MSC decides whether DTX will be applied, whereas the BSC controls the service and informs the BTS on a particular connection path whether DTX mode will be supported and, if so, which one. As shown in Figure 3.53, the BTS configures the participating radio resources TRAU (downlink-DTX) and MS (uplink-DTX).

Unlike downlink-DTX, which is associated with a particular connection, uplink-DTX is a cell parameter that is conveyed to an MS in *standby* over the BCCH and to an active MS over the SACCH, and therefore applies to all uplink connections. A downlink-DTX can only be changed in combination with a change in transmission mode (half-rate ↔ full-rate channel, data rate adaptation) [33]. This kind of situation can arise with handovers, and is indicated to the target BSC by a `Downlink DTX Flag` during the allocation of a channel.

`Interference Band to be Used` With this information element a BSC considers the maximum allowable interference band and the lowest allowable interference level during channel allocation.

3.6.9.3 Channel Activation

A BTS receives this message from its higher-ranking BSC when it is reserving a channel. It is provided with the necessary channel information (see Table 3.20).

`Channel Number` A BSC uses a Channel_Activation message to notify the BTS of the channel or subchannel available for transmission. If a channel is free, the BTS acknowledges the BSC's message with a `Channel Number` element as part of the message Channel_Activation_ack (see Section 3.6.9.4). The following channel types and combinations are possible with the channel number [18]:

- B_m + SACCH (+ FACCH)
- L_m + SACCH (+ FACCH)
- SDCCH/4 + SACCH
- SDCCH/8 + SACCH
- Uplink CCCH (RACH)
- Downlink CCCH (PCH and AGCH)

Table 3.20: Information elements of a Channel_Activation message [18]

Information element	Transmission direction	No. of octets
`Message Type`	BSC → BTS	1
`Channel Number`	BSC → BTS	2
`Activation Type`	BSC → BTS	2
`Channel Mode`	BSC → BTS	6
`Channel Identification`	BSC → BTS	4–10
`Encryption Information`[a]	BSC → BTS	3–20
`Handover Reference`[b]	BSC → BTS	2
`BS Power`[c]	BSC → BTS	2
`MS Power`[c]	BSC → BTS	2
`Timing Advance`[cd]	BSC → BTS	2
`BS Power Parameters`[e]	BSC → BTS	2
`MS Power Parameters`[e]	BSC → BTS	2
`Physical Context`[f]	BSC → BTS	2

[a]Only available when user data is to be ciphered.
[b]Only part of a Channel Activation message when a handover is taking place.
[c]Only required when the elements `BS Power`, `MS Power` and/or `Timing Advance` occur in order to establish the initial transmitting power and the content of the first layer-1 message header.
[d]Required during a channel change within a cell.
[e]Only required with *Power Control*. The maximum allowable transmitting power is indicated in the `BS` or `MS Power` element.
[f]This is an optional element that contains additional channel information.

`Activation Type` The information element `Activation Type` is used to clarify the purpose of a channel activation and reservation, namely that a BSC has one of the following intentions [18]:

- *Intracell handover* (channel change within the same cell)
- *Intercell handover* (channel change between different cells), asynchronous or synchronous
- *Call setup* in conjunction with an *immediate assignment procedure* or a *normal assignment procedure*

`Channel Mode` The information element `Channel Mode` allocates the transmission modes for both interfaces (U_m- and A_{bis}-interfaces) to a BTS. For the functionality of the TRAU this means that an uplink or downlink DTX may have to be established (see Section 3.6.9.2). Both transmitter directions are identified by a flag (1 bit), with the value `1` standing for DTX and the value `0` indicating that DTX is not allowed.

The `Channel Mode` element also indicates the purpose of use and the type of radio channel requested. Since this parameter area corresponds totally

to the Channel Type element of a Handover_Request message, the reader is referred to Section 3.6.9.2 for an exact description.

Channel Identification The Channel Identification element provides a BTS with a complete description of the channel to be reserved by the Channel Description element. With frequency hopping, a list of the channels available for alternating use, called Mobile Allocation element, is added. Both elements are part of the Layer 3 Information data field, which is forwarded to the MSC via a Handover_Request_ack message. An explanation of this parameter therefore appears in Section 3.6.9.4.

Handover Reference This information element allows an MS to identify itself as the authorized candidate by means of Handover_Access when a channel is being allocated. It contains an access number in the area between 0 and 255 (1 octet) that is allocated to the reserved channel either until the authorized MS has requested it or the handover process has been broken off.

So that the MS as well as the BTS are notified separately of the reference number, the BSC passes this information on to the BTS and, once a channel reservation has been made, to the MSC.

BS Power The parameter BS Power is used to adapt the BTS transmitter power. It reduces the transmitter power based on the existing nominal value (20–30 dB, depending on the manufacturer) in increments of 2 dB. This covers a range between 0 and 30 dB [18]. With this parameter, the network operator can have a direct influence on the particular BTS coverage area. Additional information on the mechanism of *power control* is provided under Power Command in Section 3.6.9.5.

MS Power The parameter MS Power gives the transmitter power for the mobile side.

Timing Advance The information element Timing Advance (TA) contains a time displacement factor used by the MS to compensate for signal propagation delays that originate because of the distance between MS and BTS on the uplink and downlink channel. Because the BTS requires fixed synchronization between both channels (distance of three burst periods = 1.73 ms), the MS, by timely transmission, ensures that the signal propagation delays are offset.

The coding of the time displacement factor is carried out in an octet, with the value being the dimension of the bit duration (48/13 µs ≈ 3.69 µs) [13].

BS Power Parameters The data field BS Power Parameters contains information for the BTS transmit/receive facility (TRX) for regulating its transmitter power.

Table 3.21: Information element of a Channel_Activation_ack message [18]

Information element	Transmission direction	No. of octets
`Message Type`	BTS → BSC	1
`Channel Number`	BTS → BSC	2
`Frame Number`	BTS → BSC	3

`MS Power Parameters` The `MS Power Parameters` establishes data field recommended values and limits for *Power Control* of a mobile station controlled by the BTS, analogous to `BS Power Parameters`.

`Physical Context` The data field `Physical Context` contains information about the current transmit/receive process related to the current TRX. It should be passed from one TRX channel to the next, for example in a handover. The BSC has no influence on this data. The extent and content of this data field are left to the manufacturer.

3.6.9.4 Channel Activation Ack

Through Channel_Activation_ack, a BTS announces to a BSC that, as expected, it was able to reserve the desired channel (see Table 3.21).

`Frame Number` This information element informs the BSC of the current TDMA frame number and consequently the current time grid of the BTS. In an asynchronous handover situation the BSC requires this information in order to obtain the value of the `Starting Time` element in the `Layer 3 Information` data field (see Section 3.6.9.5), which the MS uses for channel access. Four numbers that, according to a formula given in [13], can constitute an area between 1 and 2 715 648 represent the frame number in the `Frame Number` as well as in the `Starting Time` elements. The resulting frame is produced from a modulo operation between the generated number and 42 432.

3.6.9.5 Handover Request Ack

If a suitable channel can be found in the desired target cell, the responsible BSC informs the higher-ranking MSC that it and the target BTS are able to support the handover by issuing Handover_Request_ack (see Table 3.22). It also notifies the MSC of the identity of the reserved channel (`Channel Description`) to which the MS is to be rerouted. A complete description of the channels and channel allocation is given in the data field `Layer 3 Information`.

Table 3.22: Information elements of a Handover_Request_ack message [17]

Information element	Transmission direction	No. of octets
`Message Type`	BSC → MSC	1
`Layer 3 Information`	BSC → MSC	9–56
`Chosen Channel`[a]	BSC → MSC	2
`Chosen Encryption Algorithm`[b]	BSC → MSC	2

[a]Is at least included when the BSC has made a channel selection.
[b]Is at least included when the BSC has selected a coding algorithm.

`Layer 3 Information` This data field contains all the information about the new channel, including the parameters required for a channel allocation (see Table 3.23).

`Cell Description` This information element describes the target cell. Since the MS has been presynchronized with the target cell through the measurement protocol, the details on BCCH frequency and the `BSIC` are sufficient for establishing the cell identity. Both elements have a code length of one octet.

`Channel Description` This information element provides a general description of a channel [17]:

- Channel type (B_m-TCH, L_m-TCH, SDCCH)
- Type of allocated control channel (SACCH)
- Time slot number
- Indicator for support of dynamic frequency change (*frequency hopping*)
- Channel number (1–124)

`Power Command` Similarly to discontinuous transmission (DTX) described in Section 3.6.9.2, dynamic power control increases spectral efficiency and battery operating time. The transmitter power is reduced to the extent that the signal quality is still acceptable. Because co-channel interference depends on the transmitter power of the channels concerned, it can sometimes be reduced through power control.

In the GSM system the transmitter power in both directions of the radio route between BTS and MS (uplink and downlink) is controlled separately by the base station subsystem (BSS). Depending on the type of terminal equipment, the control range for the uplink channel is between 20 and 30 dB, and, depending on the manufacturer, for the downlink channel it is up to 30 dB. The transmitter power is controlled bidirectionally every 60 ms in 2 dB increments.

Table 3.23: Information elements of a `Layer 3 Information` data field [13]

Information element	Transmission direction	No. of octets
`Cell Description`	BSC → MSC	2
`Channel Description`	BSC → MSC	3
`Handover Reference`	BSC → MSC	1
`Power Command`	BSC → MSC	1
`Synchronization Indication`[a]	BSC → MSC	1
`Cell Channel Description`[b]	BSC → MSC	17
`Channel Mode`[c] `Channel Description`[d]	BSC → MSC	4
`Channel Mode 2`[e]	BSC → MSC	2
`Frequency Channel Sequence`[f]	BSC → MSC	10
`Mobile Allocation`[g]	BSC → MSC	2–10
`Starting Time`[h]	BSC → MSC	3

[a] If omitted means there is an asynchronous handover.
[b] Only occurs when the target cell supports *frequency hopping*.
[c] Included if there is a change in the transmission mode of the allocated channel.
[d] Only appears if the mobile side is utilizing two dedicated channels in which case it allocates the second channel.
[e] Included if there is a change in transmission mode in the optional `Channel Description` element.
[f] Is a combination of the allocation and channel descriptions intended for systems using *frequency hopping*. The data quantity is adapted so that a Handover_Command can be transmitted in a single signalling block. `Cell Channel Description` and `Mobile Allocation` are then omitted.
[g] Only necessary with *frequency hopping* and generally can be used with all channel types, but appears in conjunction with a `Cell Channel Description` element.
[h] Available when a change in carrier frequency occurs with a channel change.

The BSS calculates the transmitter power required by the mobile side from the receive level measured by the BTS and the maximum possible transmitter power for a terminal, at the same time taking into account the signal quality measurement, for which the BTS is also responsible. For the downlink the BSS determines the required transmitter power for the BTS through an evaluation of the periodic detailed measurement report of the MS.

The use of power control in the BSC or in the BTS is up to the manufacturer. The associated BTS-BSC protocol is only specified through dummies.

For handover the BSC establishes the transmitter power for both the mobile station and the fixed network. But the terminal equipment power is usually predetermined by the cell, and is passed on to the target BTS in the `MS Power` element of the Channel_Activation message as well as to the MS through a `Power Command` in the `Layer 3 Information` data field. In the `Power Command` 5 bits are reserved for the coding of the signal power level, thereby enabling it to assume 32 different values [13].

`Synchronization Indication` This element indicates at handover whether or not the target cell is synchronized with the original cell. It is coded with one bit, and with unsynchronized cells it has the value `0` and otherwise the value `1` [13].

`Cell Channel Description` The element `Cell Channel Description` is a data field that provides a complete overview of all available and non-available channels in the new cell. Channels 1–124 are numbered in sequence and indicated with `1` if they belong to the cell and with `0` if they do not [13].

`Channel Mode` This element indicates the coding and decoding modes, and differentiates between the following transmission modes [13]:

- signalling only
- speech full-rate channel (B_m-channel)
- data channel
- speech half-rate channel (L_m-channel)

`Frequency Channel Sequence` This data element contains a channel sequence that should be used in frequency hopping. In total, 16 different channels are entered, and the one with the lowest channel number (1–124) is explicitly listed again. Two channel numbers always form an octet [13].

`Mobile Allocation` The data field `Mobile Allocation` contains all the channels listed in the `Cell Channel Description` that belong to the new cell (channel number is indicated as `1`), identifying all the channels that can be used in the frequency hopping of the MS with `1` and all others with `0`. It contains a standard form of eight channels per octet. If the number of overall channels available to a cell is less than a multiple of eight then the remaining positions in the third octet are stuffed with `0` [13].

`Starting Time` This informs an MS with which TDMA frame it can make a connection. The calculation of the frame number is implemented in the same way as the `Frame Number` element described in Section 3.6.9.4. The MS can then be instructed to wait to transmit for a maximum of up to 42 431 frame durations (around 195.8 s) [13].

`Chosen Channel` This informs an MSC which of the following channel types has been allocated to the mobile side [17]:

- SDCCH
- L_m-TCH
- B_m-TCH

`Chosen Encryption Algorithm` This information element informs an MSC whether the BSC supports ciphering and which of the seven different A5 types is being used [17].

Table 3.24: Information elements of a Handover_Command message [17]

Information element	Transmission direction	No. of octets
`Message Type`	MSC → BSC BSC → BTS BTS → MS	1
`Layer 3 Information`	MSC → BSC BSC → BTS BTS → MS	9–56

Table 3.25: Information elements of a Handover_Access message [13]

Information element	Transmission direction	No. of octets
`Handover Reference`	MS → BTS	1

Table 3.26: Information elements of a Physical_Information message [13]

Information element	Transmission direction	No. of octets
`Message Type`	BTS → MS	1
`Timing Advance`	BTS → MS	1

Table 3.27: Information elements of a Handover_Complete message [13]

Information element	Transmission direction	No. of octets
`Message Type`	MS → BTS	1
`RR Cause`	MS → BTS	1

3.6.9.6 Handover Command

After a layer-3 Handover_Request_ack message has been received by an MSC to acknowledge a channel query, the MSC forms a Handover_Command from it that essentially only includes the BSC data field `Layer 3 Information` (see Table 3.24). Via BSC and BTS, this message reaches the relevant MS, which receives all the information it requires about the channel characteristics and access parameters through the data field mentioned.

3.6.9.7 Handover Access

With Handover_Access an MS sends a request to the target BTS to allocate the reserved channel to it. This layer-3 message only contains the `Handover Reference` element (1 octet) known from the `Layer 3 Information` data field, which is initially sufficient for the identity control of the MS in order to allocate a channel (see Table 3.25). A more comprehensive authenticity test of the MS and the user takes place after handover.

Table 3.28: Information elements of a Clear_Command message [17, 13]

Information element	Transmission direction	No. of octets
`Message Type`	MSC → BSC BSC → BTS	1
`Layer 3 Header Information`[a]	MSC → BSC BSC → BTS	4
`Cause`	MSC → BSC BSC → BTS	3–4

[a]Serves no particular purpose and is an optional element. However, it should be sent all the same in order to facilitate the functional process between network elements unless the MSC is certain that the BSC can manage without it. This will no longer constitute part of this message in any future GSM recommendation.

3.6.9.8 Physical Information

A Physical_Information message is directed by the network to an MS to instruct the mobile side to stop sending a Handover_Access message and to activate the reserved channel for the MS (see Table 3.26).

3.6.9.9 Handover Complete

Once a connection has been established between MS and BTS layer 2 through the exchange of an SABM and a UA frame, the MS notifies the network with Handover_Complete that it has completed the handover successfully (see Table 3.27). This message is sent back over the network entities to the MSC that originated the Handover_Command [17].

`RR Cause` An MS uses this information element to notify the network of the reason why it has left the original channel. A number of different reasons and the related coding are mentioned in [13, Section 10 and Appendix F].

3.6.9.10 Clear Command

If a handover has been terminated by the mobile side with Handover_Complete then the MSC has the task of also completing the handover process on the network side. To do so, it sends the BSC responsible for the original cell a Clear_Command in order to initiate the release of the allocated radio resources there (see Table 3.28). The BSC then forwards this message to the relevant BTS.

`Layer 3 Header Information` This element informs a BSS of the content of a layer-3 message header that is to be used for sending user data over the radio interface. At this point in time, however, it is of no significance in the handover process.

Table 3.29: Information elements of a Clear_Complete message [17]

Information element	Transmission direction	No. of octets
`Message Type`	BTS → BSC BSC → MSC	1

`Cause` The `Cause` element has already been briefly described in Section 3.6.9.1. The following are some typical reasons for releasing a channel [17]:

- *Call Control*
- *O & M Intervention*
- *Equipment Failure*
- *Handover Successful*
- *Protocol Error between BSC and MSC*

3.6.9.11 Clear Complete

As soon as the BTS has released the channel allocated to the MS that is being supplied by another cell after handover is completed on the mobile side, it immediately reports the process to its higher-ranking BSC with a Clear_Complete (see Table 3.29).

The BSC then forwards this message to the MSC, thereby also concluding the network part of the handover protocol.

3.6.10 Inter-MSC Handover

In an inter-MSC handover the channel is switched to a radio cell managed by another MSC. There are two different scenarios:

- In a *basic handover*, handover first takes place from one MSC service area to another (see Figure 3.54a). In the process the connection from MSC A is routed to MSC B over the wireline ISDN network.
- A *subsequent handover* occurs when at least one *basic handover* has taken place (see Figure 3.54b). Also in this case MSC A continues to be responsible for controlling the connection. Here too the connection is routed by MSC A over the fixed ISDN network to MSC C.

3.7 Location Update

When a mobile user leaves a location area, the HLR and the VLR must be updated immediately to ensure that the user can be reached no matter which location he is currently moving in. The mobile station continuously evaluates the receive quality of all base stations capable of receiving, and

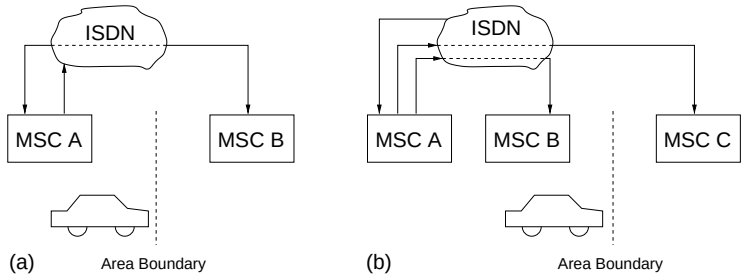


Figure 3.54: (a) First area change (*basic handover*). (b) subsequent area change (*subsequent handover*)

logically attaches itself to the one with the strongest signal. If in the process it has allocated itself to a base station that is located outside the previously valid location area, it initiates a *location update*, thereby informing the network of its new location. Two different situations can arise:

- Change within the same VLR area
- Change of VLR area

If there is a change in VLR area at the same time the location area changes, and the sequence of the process is as is shown in Figure 3.55. Because no entry for the mobile station exists in the new VLR, its data must be transmitted from the HLR and deleted in the old VLR. If a mobile station requests a location update, a four-character long temporary mobile subscriber identity (`TMSI`) as well as the old location area identification (`LAI`) usually also exists. This makes it possible to determine which was the former VLR and consequently also the international mobile subscriber identity (`IMSI`) of the MS. Alternatively the mobile station can also immediately be requested to transmit its IMSI. After the user has been authenticated, the new VLR informs the HLR about the location change by sending an Update_Location message. The HLR subsequently sends the necessary user data to the new VLR using Insert_Subscriber_Data.

Cancel_Location then deletes the subscriber's entry in the former VLR. Through Location_Updating_Accepted the new VLR confirms to the mobile station that the transfer has taken place.

3.7.1 Roaming Support

Roaming is a service provided by a mobile radio network allowing a mobile user freedom of movement while still being within reach of coverage. It allows subscribers to make calls and to receive calls, and independently to make use of several services at the same time. There is a difference between roaming within the same network (across location areas), national roaming (between different networks—this is not yet supported, however) and international roaming.

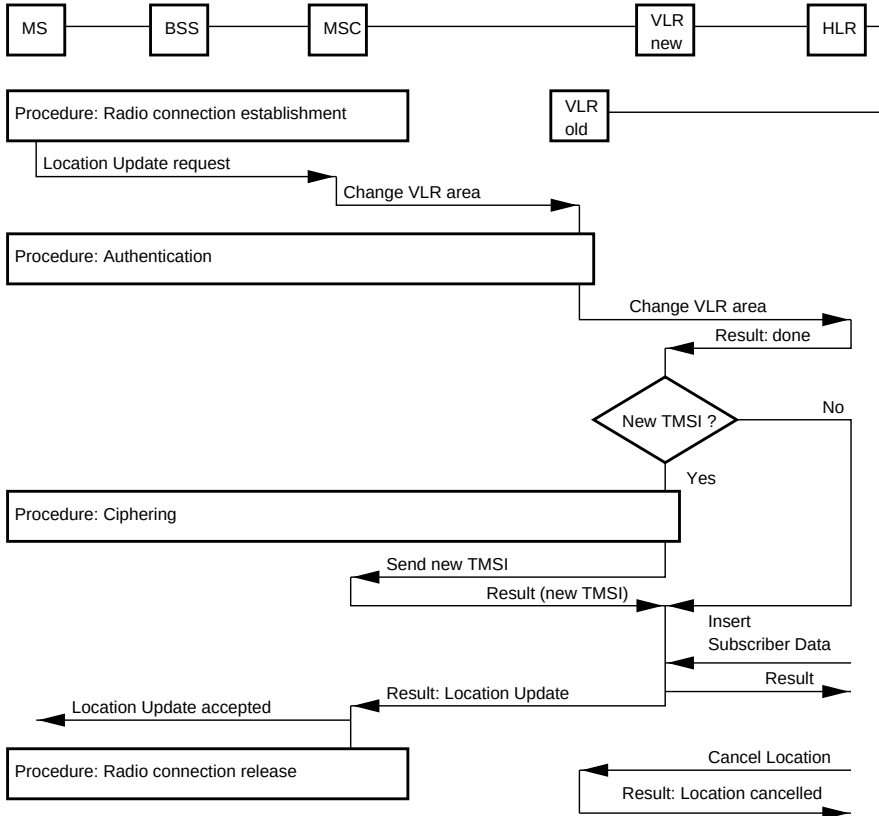


Figure 3.55: Sequence of location update procedure

3.7.2 Numbering Plan for Roaming

The numbering in GSM follows the rules of ITU-T Rec. E.164 for ISDN (see Figure 3.56). In addition to an `MSISDN` telephone number, a mobile subscriber has another number—an `IMSI` with a maximum length of 15 digits that provides him with unique identification. The numbers are structured as follows (ITU-T Rec. E.212); see Figure 3.57:

- A *Mobile Subscriber ISDN Number* (MSISDN) consists of the home country code (`CC`), the national destination code (`NDC` identifies the home PLMN and is therefore the HLR address) and the subscriber number (`SN`) in the HLR; thus `MSISDN=CC+NDC+SN`.
- The *International Mobile Subscriber Identity* (IMSI) consists of the home mobile country code (`MCC`), the code for the mobile network, i.e., the HLR address `MNC`, as well as the subscriber identity (`MSIN`) in the HLR.

Table 3.30: Country and network codes for European GSM networks

Country	Network operator	MCC	MNC	Message displayed or short message		Dialling code xx is internat. prefix
Belgium	RTT Belgacom	206	01	BEL MOB-3	MOB-3	00 xx
Denmark	Tele Danmark Mobil	238	01	DK TDK-Mobil	TD MOB	009 xx
Denmark	Dansk Mobil Telefon	238	02	DK SONOFON	SONO	009 xx
Germany	Deutsche Telekom AG	262	01	D1-Telekom	D1	00 xx
Germany	Mannesmann Mobilfunk	262	02	D2 PRIVAT	D2	00 xx
Finland	Telecom Finland	244	91	FI TELE FIN	TELE	990 xx
Finland	Oy Radiolinja AB	244	05	FI RADIOLIJNA	RL	990 xx
France	France Telecom	208	01	F FRANCE TELECOM	FT	19 xx
France	SFR	208	10	F SFR	SFR	19 xx
Great Britain	Cellnet	234	10	UK CELNET	CLNET	010 xx
Great Britain	Vodafone	234	15	UK VODAFONE	VODA	010 xx
Ireland	Eircell	272	01	IRL EIR-GSM	E-GSM	00 xx
Italy	SIP	222	01	I SIP	I SIP	00 xx
Luxemburg	P&T Luxembourg	270	01	L LUXGSM	P&T L	00 xx
Netherlands	PTT Telecom	204	08	NL PTT TELECOM	NL PTT	09 xx
Norway	Tele Mobil Norway (Nor. Telecom)	242	01	N TELE-MOBIL	TELE	095 xx
Norway	NetCom GSM A/S	242	02	N NETCOM GSM	N COM	095 xx
Austria	PTV Austria	232	01	A E-Netz	MN-E	0 xx
Portugal	Telecom. Moveis Nacionals	268	06	P TELEMOVEL	TMN	00 xx
Portugal	Telecel	268	01	P TELECEL	TLCL	00 xx
Spain	Telefonica	214	07	E TELEFONICA	TLFCA	07 xx
Sweden	Televerket Radio (Sw. Telecom)	240	01	S TELIA MOBITEL	TELIA	009 xx
Sweden	Comvik GSM AB	240	07	S COMVIQ	IQ	009 xx
Sweden	AB Nordic Tel	240	08	S EURIPOLITAN	EURO	009 xx
Switzerland	Swiss PTT Telecom	228	01	CH NATEL D GSM	NAT D	00 xx
Turkey	PTT Turkey	286	01	R TURKIYE GSM	TR GSM	00 xx

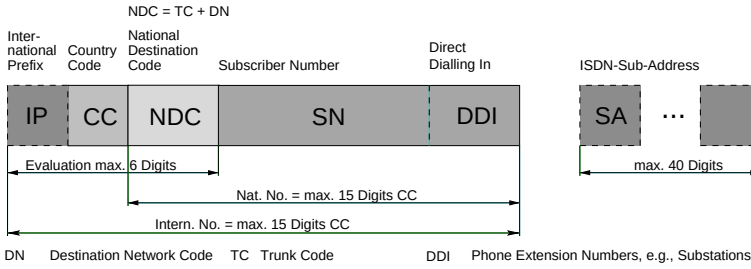


Figure 3.56: Numbering structure according to ITU-T Rec. E.164

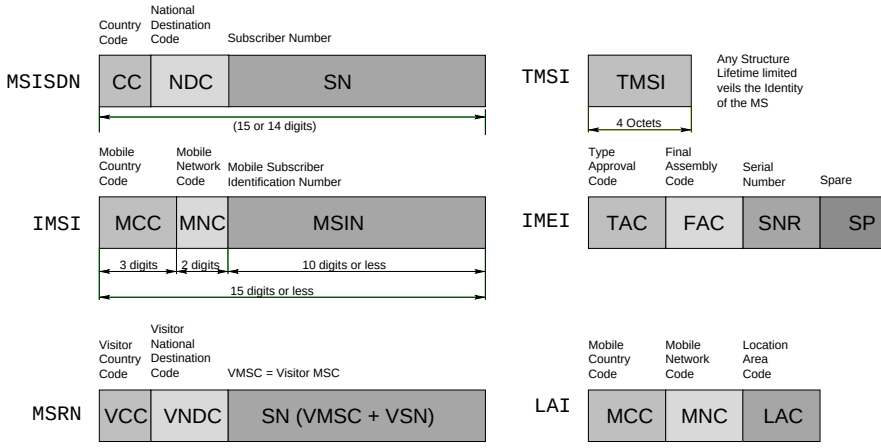


Figure 3.57: Structure of GSM numbers

Table 3.30 lists the MCC and MNC codes for the current European GSM networks.

In addition, a temporary address, the *Mobile Station Roaming Number* (MSRN), provides the link to the mobile subscriber's current location. An MSRN is linked to location, is temporarily allocated by a VLR (and stored in the HLR) at the request of the MSC, and is required to make a connection from the fixed network to a mobile station over the GMSC. It consists of the country code of the visited mobile network (VCC), the local code (area that the user is currently visiting (VNDC)), identification of the visited MSC (VMSC) and the subscriber number (VSN) allocated by the VLR. The MSRN protects the identity and the location of the mobile user from being discovered through eavesdropping on the signalling traffic.

If a mobile station wishes to send a signal in the network, it uses a local temporary number for identification, a so-called *Temporary Mobile Subscriber Identity* (TMSI), to conceal the IMSI. The TMSI is allocated and coded and transmitted together with an LAI from the current VLR to the mobile sta-

tion. Therefore a TMSI only has local significance and is only valid within the area administered by a visitor location register (VLR). A TMSI is periodically changed to ensure the confidentiality of the mobile subscriber's transmitted information. The *location area identity* (LAI) in the first two elements is constructed in the same way as the IMSI; the third element contains the *location Area Code* (LAC).

Roaming has a constant high requirement for signalling resources in a mobile radio network, even if a user is not communicating. Therefore the current location and access rights of a mobile user must constantly be updated and, if required, be specified. Because roaming not only necessitates a change in location area in the home PLMN but also between international networks, several networks are sometimes affected.

3.8 Connection Setup

Similarly to ISDN, a service-integrated GSM mobile radio network allows different telecommunications services to be provided on the same subscriber line. Therefore, in addition to a destination address, the terminal placing a call must also specify details on the desired transmission characteristics of a connection and the conditions of compatibility for the terminal called. The called unit can then select the terminal equipment suitable for the desired service. In contrast to ISDN, there are additional requirements concerning the specification of the mobile subscriber's location address with a mobile-terminated call and the selection of special interworking functions between the protocols used at the radio interface of the mobile network and the protocols of the fixed network, e.g., ISDN.

3.8.1 Mobile-Terminated Call

If a caller from the fixed network wants to reach a mobile subscriber, he must dial his ISDN number (MSISDN). If the switching centre of the fixed network recognizes that the number is one within a PLMN, the call with an *initial address message* (IAM) is routed from the public network to the next gateway-MSC (GMSC) (1) (see Figure 3.58).

From the code NDC contained in the IAM message, the GMSC establishes the mobile subscriber's home location register (2). The HLR checks that the number exists and, if the services desired were also included in the IAM message, that the mobile subscriber is authorized to use the services. The HLR also asks the VLR to provide a mobile station roaming number MSRN for the pending connection request (3).

After receipt of the MSRN (4), which the HLR uses to determine which MSC is responsible for the current location area, it communicates the number to the GMSC (5), which then establishes a connection to the responsible MSC (6). The MSC instructs the VLR to forward the availability status of the mobile station (7). If the MS is available, the VLR notifies the MSC (8),

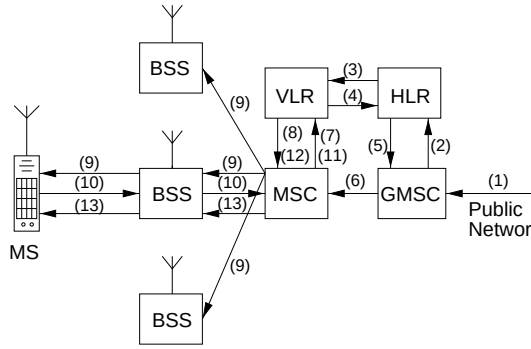


Figure 3.58: Mobile-terminated call

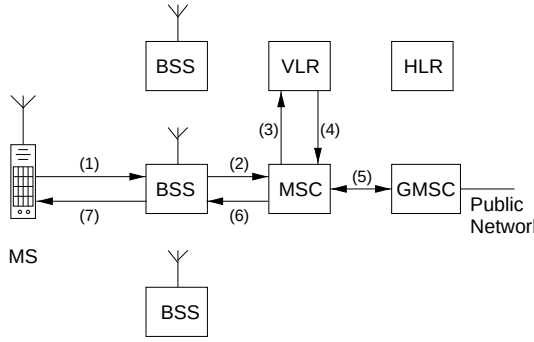


Figure 3.59: Mobile-originated call

which then initiates a paging call to the subscriber in all the radio cells of the MS's location area (9). If the mobile subscriber answers (10) then, after all security procedures have been completed (11), the MSC is instructed by the VLR (12) to establish the connection in the radio network (13).

3.8.2 Mobile-Originated Call

A mobile subscriber in a GSM-PLMN can request a connection at any time. A mobile-originated call begins with an operation that initiates access to the network and informs the VLR of the type of request and the transmission ciphering key currently stored in the mobile station along with the subscriber identification. To avoid an unnecessary occupancy of the radio channel, the desired call number is indicated in a display on the terminal equipment so that in case of input error the user may correct the number before pressing the calling key.

The steps that take place during a connection request are illustrated in Figure 3.59. After completion of the required security procedures and the

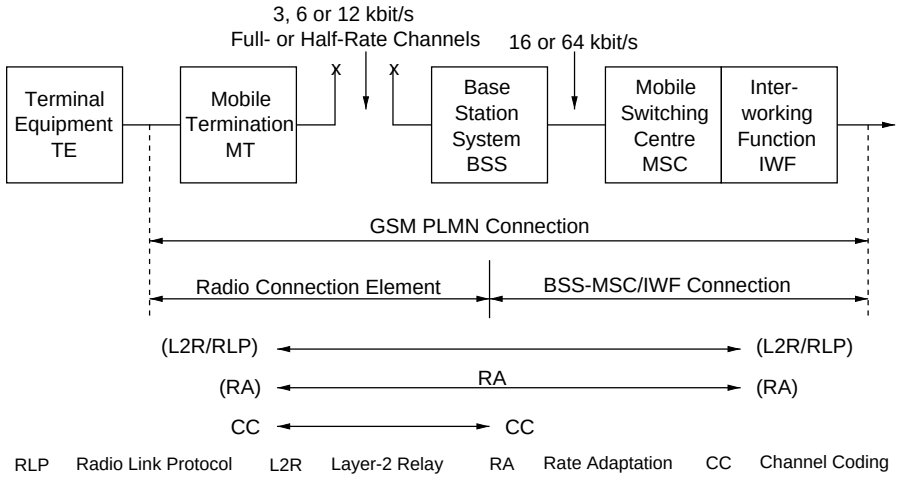


Figure 3.60: Connection for data transmission in PLMN

resulting call request (1), the mobile station making the call forwards a RIL3_CC_Setup message to the MSC containing the telephone number of the called subscriber as well as requirements in respect of quality of service of the network path and compatibility of the subscriber unit being called (2).

To meet the desired requirements, the MSC must check the mobile subscriber's authorization by sending a query to the VLR (3 + 4) as well as reviewing the resources available to it (IWF, free line to fixed network, etc.) (5). If the MSC has been successful, it allocates the resources to the connection, selects the necessary transcoding function (*interworking function*, IWF), and, if a connection has been established successfully, sends the mobile station a RIL3_CC_Alerting message (6 + 7), which is converted into an alert signal in the mobile station.

3.9 Data Transmission and Rate-Adaptation Functions

Figure 3.60 illustrates the principles of a GSM-PLMN connection. Different adaptation functions must be carried out to ensure that a connection has been established correctly. In respect of the physical interface, the so-called S-interface (ITU-T: I.430-series; see Figure 3.3) is to be supported. Mobile stations are connected via two 64 kbit/s B-channels as traffic channels and a D-channel as the control channel. However, the whole range of ISDN services cannot be offered because of the restricted bandwidth of the radio channels. Conventional R-interfaces are also supported. The data interfaces are supported according to ITU-T X.21, X.21bis, X.25 and V.24.

3.9.1 Rate Adaptation to Traffic Channel Performance

Rate adaptation is defined according to ITU-T X.30/V.110. Modifications were required because the radio interface only supports data rates lower than 64 kbit/s. ITU-T describes a three-level process with the functions:

RA 0 This converts an asynchronous input stream into a synchronous output stream of $2^n \cdot 300$ bit/s.

RA 1 This function is dependent upon synchronous input data as well as status information on the subscriber interface, control information and synchronization patterns. Rate adaptation is then carried out through multiplexing on an 8 or 16 kbit/s output stream in the form of 80 bit-blocks. In ISDN the rates 8/16 kbit/s are called intermediate rates.

RA 1' This modification of RA 1 produces the GSM intermediate rates of 12, 6 and 3.6 kbit/s. Intermediate rates are produced through the elimination of the synchronization patterns and control information. The latter is explicitly conveyed over the L_m-control channel (also called the D_m-channel). With the 3.6 kbit/s output rate even some redundant data bits are removed.

RA 2 Through so-called bit stuffing, this function forms a datastream of 64 kbit/s from the blocks received. This third level is only used in the base station.

Figure 3.61 shows the application of rate adaptation with transparent data transmission and the R-interface; Figure 3.62 shows a case of non-transparent data transmission with the R-interface. The S-interface to which a mobile station with an ISDN interface can be directly connected also appears in both illustrations.

3.9.2 Rate Adaptation in the Connection BTS/Transcoder to an MSC or MSC/IWF

Because an MSC can only switch 64 kbit/s channels, the GSM intermediate rates must be increased to 64 kbit/s through function RA 2, which is usually allocated to the speech-transcoder function (TRAU). The functions for (non-)transparent services are almost identical. Bits that in transparent transmission indicate the transmission rate are used in transparent transmission to indicate the alignment of the RLP blocks and to control discontinuous speech transmission (DTX), if applicable.

3.9.3 Layer-2 Relay Function and Radio Link Protocol

A traffic channel transmits with error correction (FEC) (see Section 3.3.7.1). The RLP function (see Section 3.9.4) offers an *automatic repeat request (ARQ)*

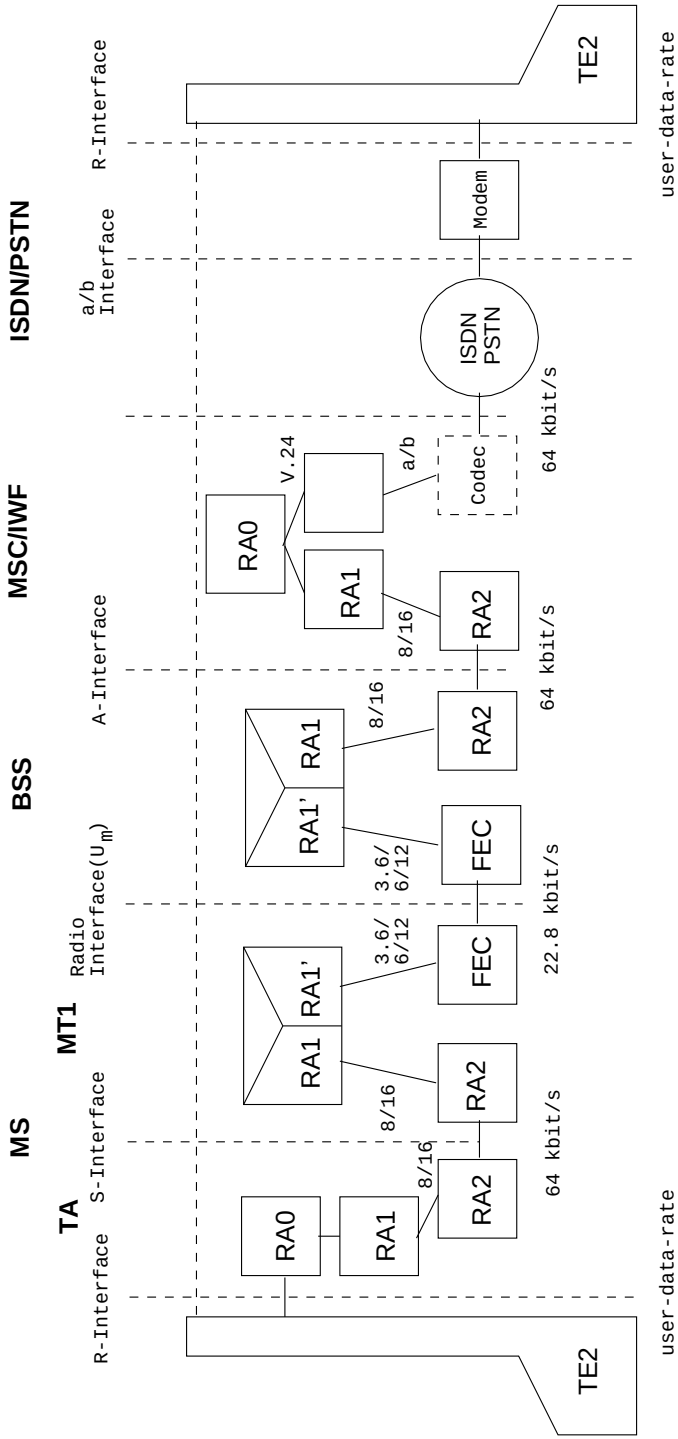


Figure 3.61: Example of asynchronous transparent data transmission

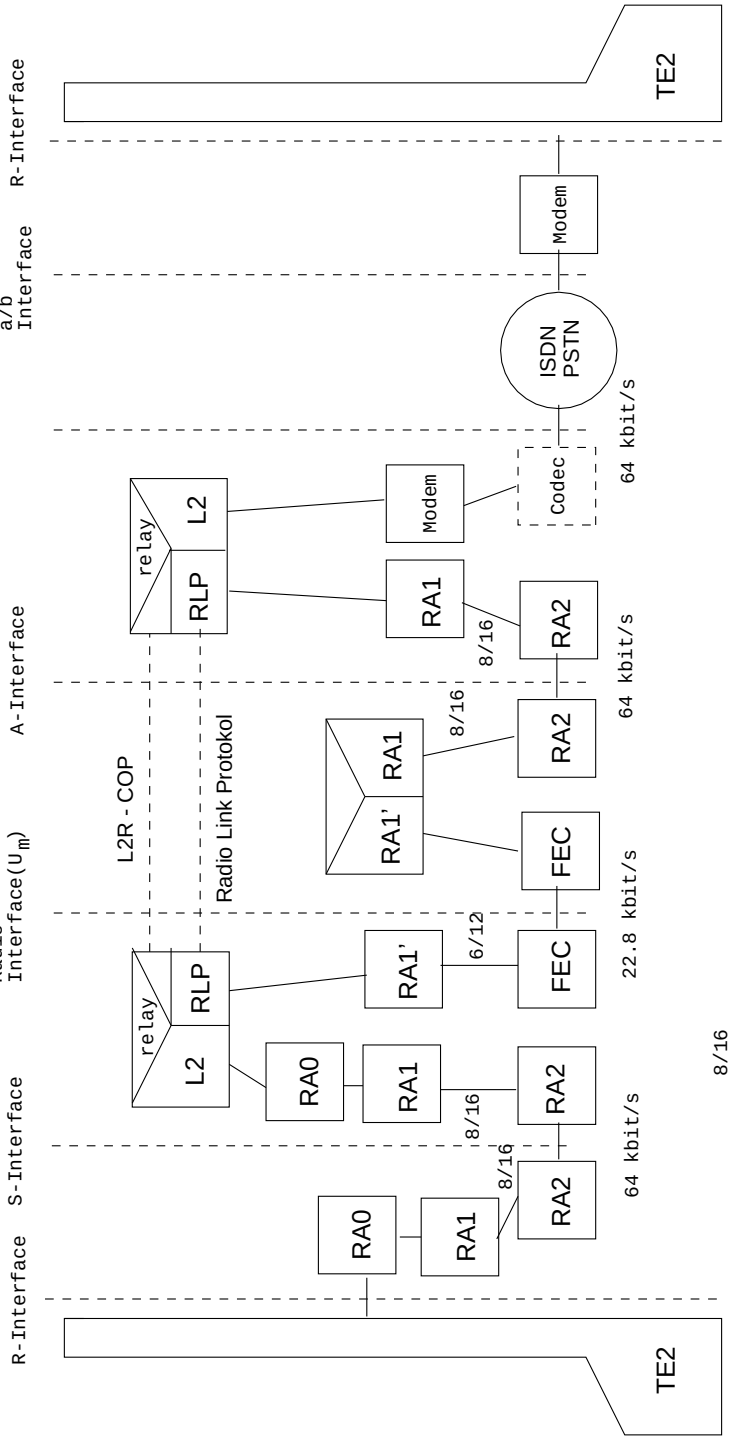


Figure 3.62: Example of asynchronous non-transparent data transmission (see Section 3.9.4)

protocol that extends from the mobile station to the *network interworking functions* (IWF) in the MSC in order to detect errors that were corrected/not corrected by the FEC procedure and to eliminate them by repeating the transmission (see Figure 3.62). The L2R function converts the layer-2 protocol of the terminal equipment into a *connection-oriented* L2R protocol (COP) that uses transmission protected by an RLP. RLP and L2R functions are only available in the mobile radio network termination and in the network gateways, because only MSCs with IWFs are the main interworking points (GMSC) so that problems of protocol synchronization can be kept to a minimum.

3.9.4 Radio-Link Protocol (RLP)

A bearer service with error correction (FEC) is sufficient for transparent telecommunications services in a GSM network. Some applications, however, require a higher level of transmission quality that needs an ARQ protocol, which means that a bearer service with error detection is being used. For the non-transparent service a layered full-duplex protocol, called the radio link protocol (RLP), is used, which corresponds to layer 2 of the ISO/OSI reference model.

3.9.4.1 Characteristics of the Radio Link Protocol

The radio link protocol supports the transmission of user information up to 9.6 kbit/s without any noticeable delay. According to GSM Rec. 04.22, throughput can typically be maintained in 90 % of a cell area for 90 % of the time, with a maximum delay of 120 ms per direction. The FEC method of the bearer service used offers a bit-error ratio (BER) of approximately 10^{-3} at 9.6 kbit/s and approximately 10^{-7} at 4.8 kbit/s with a full-rate traffic channel. With the ARQ mechanism a BER of approximately 10^{-7} at 9.6 kbit/s is possible. The RLP was further developed from the HDLC protocol (ISO IS 4335) because radio network characteristics differ from those of wireline connections for which HDLC was intended. Figure 3.63 shows the three formats of the RLP protocol data units and the control commands used. U- and S-blocks do not carry information. The main differences with HDLC are:

- No synchronization patterns (flags) are used.
- Blocks are numbered modulo 64 instead of modulo 8 or 128.
- The check sum (FCS) has been extended from 2 to 3 bytes.
- RLP permits simultaneous transmission of user and status information (see S+I-block).
- There are no addresses, because of exclusive use of only one connection at a time.

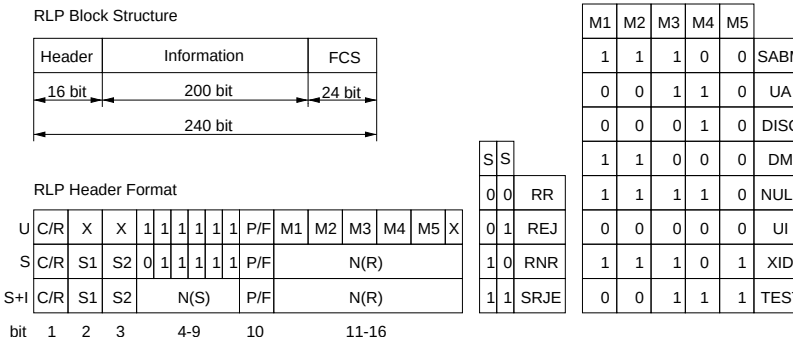


Figure 3.63: Protocol data units of RLP

- Commands/responses are differentiated by a C/R-bit, the P/F-bit (`poll/final`) is set accordingly.
- An X-bit (E, *evolutionary*) is reserved (e.g., for point-to-multipoint connections).

3.9.4.2 Functioning of the Radio Link Protocol

Stations can establish, cancel or terminate a connection at any time, which is called *Asynchronous Balanced Mode Procedure* (ABM). RLP supports point-to-point duplex connections. The RLP entity can be found either in *asynchronous disconnected mode* (ADM) or in ABM in which data transmission is taking place. U-blocks are unnumbered control blocks for establishing or terminating a communication link. Their function is coded in the "MMMMM" field. S-blocks (*supervisory*) are control blocks with sequence numbers; they acknowledge data blocks that have been received correctly and initiate the retransmission of data blocks detected to have errors or temporarily suspend the transmission of data. Their function is coded in the "SS" field. The definitions of U- and S-blocks with RLP are closely similar to those of HDLC. In RLP information blocks, called I-blocks in HDLC, also carry control information. They are therefore called S+I-blocks. The N(R)-fields (*Receive Sequence Number*) in S- and S+I-blocks report on receiving status. S+I-blocks also have an N(S)-field (*Send Sequence Number*) for numbering transmitted blocks modulo 64.

3.10 Services in the GSM Mobile Radio Network

An important element in the conception of the Pan-European radio network was the definition of the services standardized in the network (see Figure 3.64).

The GSM network enables the integration of different voice and data services and also offers gateway functions for interconnection to other telecom-

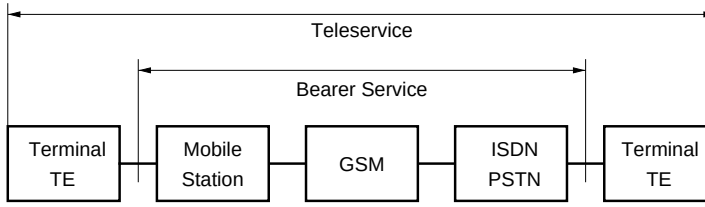


Figure 3.64: Relationship between bearer services and teleservices in a point-to-point connection in GSM

munications networks for voice and data transmission. In addition to the voice service, the GSM recommendations are planning a phased introduction of data and teleservices.

The telecommunications services offered by GSM are divided into three main categories:

Bearer Services Bearer services are all telecommunications services that allow mobile subscribers to transmit signals in bit-transparent mode between the mobile subscriber and a user of any partner network. They only offer pure transport services as defined for the lower three layers of the ISO/OSI reference model, namely connection-oriented channel and packet-switched data transmission.

Teleservice A teleservice is a telecommunications service that allows applications-oriented communication between a mobile user and a second user in accordance with standardized protocols. This requires protocols from all seven layers of the ISO/OSI model. The GSM recommendations intend that the teleservice will make use of only one or in any case a small number of different bearer services.

Supplementary services The supplementary services are further facilities that are offered to subscribers in addition to the services mentioned above. These are not independent services, and are always offered in conjunction with the teleservices or the bearer services.

Support for Value-Added Services Value-added services like *Tariff Area Indication*, *E-mail Access*, *Electronic Banking* or *Travel Information* are not standardized in the GSM specifications. However, in the GSM specifications two mechanisms are foreseen to facilitate the provision of operator-specific value-added services.

Figure 3.64 shows a point-to-point connection in GSM and the relationship between bearer services and teleservices.

Table 3.31: The introduction phases for services in the GSM network

Phase	Introduction in Europe	Services
E1	1991	Telephone services as well as some supplementary services
E2	1994	Telephone services and a limited number of data and supplementary services
E3	1996	Telephone services, advanced data and supplementary services

3.10.1 Service Introduction Phases

Not all the services specified in the standard can and will be available to subscribers at the same time the network is operational. Therefore GSM Rec. 01.06 presents a concept for a phased introduction of the different services according to their importance and the anticipated market situation. The services are classified accordingly as *essential* (E) or *additional* (A):

E-service must be provided by all mobile communications networks (PLMN).

A-service can be offered by the network operator.

E-services are relegated to one of three introduction phases (see Table 3.31), with the index providing the values (1,2,3) to indicate which services must be offered first. Thus E1-services were already available at the same time when most GSM networks were implemented, whereas the subsequently ranked services, with consideration of upward compatibility, were introduced at a later time. On the one hand, this approach takes into account the requirements of network operators who want to provide their services in line with prevailing market conditions, and, on the other hand, it addresses the necessity for harmonization in the telecommunications area.

3.10.2 Bearer Services

GSM supports different variants of data transmission, thus functioning as a *bearer* for data transmission. The bearer services allow GSM to be used for non-voice services up to 9600 bit/s. Because voice transmission is expected to represent the main traffic component and it is projected that data services will equate to less than 10 % of the overall transmission volume, only some of the bearer services were classified as *essential*.

In GSM networks a radio path might be interrupted briefly owing to radio shadowing. Because of the redundancy available in natural speech, this is less serious for voice transmission than it is for data transmission. Appropriate measures such as error-recovery protocols are necessary for the data services to eliminate the effects of interference.

The bearer services support transparent and non-transparent, synchronous or asynchronous data services. In a non-transparent service the time transparency is violated, i.e., data experiences variable delays and throughput is dependent on the current signal quality in the radio network.

The transparent bearer service This is one in which only functions of ISO/OSI layer 1 are involved and none of the higher layers.

A user is allocated a traffic channel without the support of ARQ protocols but with a defined transmission rate. Because a transparent bearer service uses a channel coding method with forward error correction (FEC), data is transmitted with constant throughput and constant delay. The use of error-correction methods with different code ratios produces the following possible data rates:

- With a full-rate channel: 9.6 / 4.8 / 2.4 kbit/s
- With a half-rate channel: 4.8 / 2.4 kbit/s

The quality of a connection (measured on the basis of its bit-error ratio) depends on the quality of the radio channel.

The non-transparent bearer service This uses protocols whose functions correspond to layer 2 of the ISO/OSI reference model in order to protect communication through error detection and repeated transmission and to optimize throughput by data flow control.

A non-transparent service is based on a transparent bearer service and also uses the *radio link protocol* (RLP). In addition to the *reject* command, where transmission is repeated once errors have occurred of all the received frames starting with the first frame number indicated in the reject command, it also allows the *selective-reject* mechanism in which only the disrupted frame needs to be repeated (see Section 3.9.4 and Figure 3.63).

Unlike HDLC, control commands can also be transmitted in information frames. The maximum window size of 61 is adapted to the delay caused by bit interleaving.

This hybrid ARQ protocol results in a residual bit-error probability of $< 10^{-7}$, with throughput and transmission delay times subject to strong fluctuations due to radio field conditions. With a non-transparent data service the error-correction method FEC corresponds to that with a data rate of 9.6 kbit/s, although equipment can also transmit at a lower bit rate.

The following are some of the services available as bearer services:

- Data circuit-switched, duplex-asynchronous (E2/A)
- Data circuit-switched, duplex-synchronous (A)
- Circuit-switched access to PAD of a packet-switched data network (Datex-P) (E2/A) (see ITU-T Recs. X.25, X.28, X.29)

- Synchronous duplex access to a packet-switched data network (E3/A)
- Alternate speech/unrestricted digital
- Speech followed by data
- 12 kbit/s unrestricted digital

3.10.2.1 Data Circuit-switched, Duplex-Asynchronous

In cooperation with ISDN or the telephone network, which throughout Europe offers a full-duplex data transmission rate of 1200 bit/s, GSM is planning to provide bearer services with a transmission rate of between 1200 bit/s and 9600 bit/s. In the beginning two services at 300 bit/s and 1200 bit/s were supported by GSM through an appropriate gateway into the telephone network (PSTN). Because of the anticipated high demand and the relatively simple implementation, these bearer services were classified as E2.

3.10.2.2 Data Circuit-switched, Duplex-Synchronous

In the analogue telephone network (PSTN) and in the circuit-switched data network four synchronous data transmission rates of 1200, 2400, 4800 and 9600 bit/s are used. Correspondingly, four PLMN bearer services have been standardized for GSM.

3.10.2.3 Circuit-switched Access to PAD

Access from data terminal equipment over the public telephone network to the data packet-switched service (*packet assembler/disassembler*, PAD) has been regarded as an attractive and easily implementable option and given the classification E2. Two bearer services with transmission rates incoming/outgoing of 300/300 bit/s and 1200/1200 bit/s have been standardized for this purpose. A service with 1200/75 bit/s (videotex) has also been specified, and although it is important in the fixed and mobile networks of some countries, it has been classified as A.

3.10.2.4 Data Packet-Switched, Duplex-Synchronous

Packet-switched data network services with transmission rates of 2400, 4800 and 9600 bit/s provide the only option for transmitting data with high rates and quality of service to a CEPT-wide area. Consequently PLMN bearer services that use ARQ protocols with flow control were defined. Although important, these bearer services have been classified as E3 because of the high implementation cost. At the request of individual operators, a 1200 bit/s service (classified as A) has also been specified.

Table 3.32 presents a summary of the bearer services described, with T standing for transparent and NT for non-transparent transmission mode.

Table 3.32: Bearer services in the GSM network

Bearer service	Transmission rate	Mode	Partner network
Data cda	300–9600 bit/s	T/NT	PSTN/ISDN
Data cds	1200–9600 bit/s	T	PSTN/ISDN/CSPDN
PAD cda	300–9600 bit/s	T/NT	PSPDN
Data cda	2400–9600 bit/s	T/NT	PSPDN

cda	circuit data asynchronous	CSPDN	Circuit-Switched Data Network
cds	circuit data synchronous	PSPDN	Packet-Switched Data Network

3.10.3 Teleservices

Two important characteristics of the teleservices are system-optimized coded voice transmission and data transmission in which the use of special adapters ensures compatibility with terminal equipment in ISDN or over a modem to the PSTN (see Figure 3.3).

The following teleservices are standardized in the GSM Rec. 02.03 [5]. The parantheses indicate whether a service is classified as *essential* (E1, E2, E3) or *additional* (A) or is marked for *Further Study* (FS).

- Telephone service (E1)
- Emergency call service (E1)
- Three videotex access profiles (A)
- Telefax Group 3 (E2)
- Access to electronic mail (A)
- Three short-message services:
  - *Short-Message Mobile-Terminated Point-to-Point* (E3)
  - *Short-Message Mobile-Originated Point-to-Point* (A)
  - *Short-Message Mobile Cell Broadcast* (FS)

3.10.3.1 Telephone Services

The telephone service is the most important service offered, and therefore its introduction is classified as E1. Telephone services are provided with a number of additional features such as call diversion, call barring and closed user groups. The transmission of voice is exclusively carried out digitally and has been optimized for telephony, which as anticipated is the prevailing form of communication. The most important aspect of this optimization is the economic use of the frequency spectrum. Because the speech codecs transmit the analogue data signals processed by a data modem in a distorted form, they are replaced by special modems during data transmission over the communications channel of the air interface.

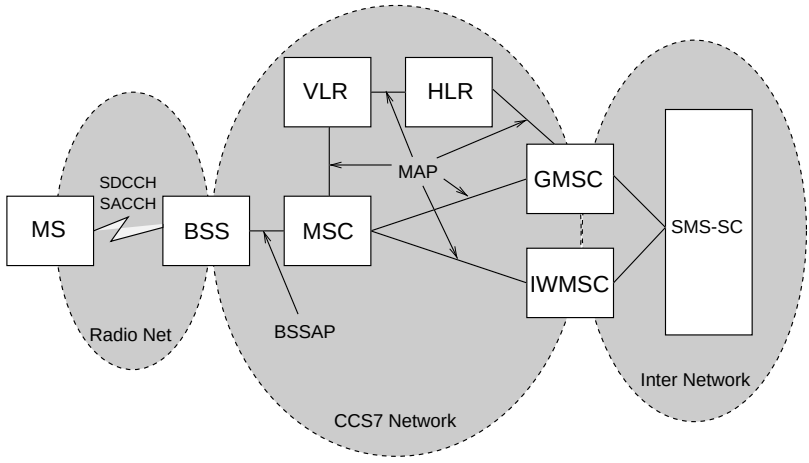


Figure 3.65: Implementation of the short-message services in GSM

3.10.3.2 Emergency Call Services

The emergency call service is available with the introduction of GSM (E1) to enable voice communication to be established with the regional emergency centre in the area responsible for the mobile station on the basis of a standard access procedure or by dialling a national emergency number.

3.10.3.3 Short-Message Services

A short-message service (SMS) for point-to-point connections allows messages up to 160 bytes in length to be sent between a GSM mobile station and an *SMS centre* (see Figure 3.65) [4]. Longer messages are broken down into parts 160 bytes in length, albeit without the assurance that the sequence will be maintained.

The following features typify the *short-message service*:

- The transmission of messages between the relevant GSM network entitites is carried out over signalling channels (SACCH and SDCCH).
- The SMS is a store-and-forward service that is implemented through the *SMS centre*.

Standardized interfaces, such as for the packet-switched public data network or common channel signalling system No. 7, are usually required for access to different service providers. The transmission of short messages requires the same functions as normal data and voice communication (e.g., setting up a signalling channel, authentication).

Because short delays (a few seconds) can be tolerated with SMS, and short messages can also be transmitted in parallel to an existing connection, there is

a better utilization of the signalling channels. Messages arriving at a mobile station are indicated on the display field or, if the receiver is temporarily not available, stored and then resent within a predetermined period of time. Outgoing messages can either be stored ones or messages that have been entered through the keyboard of a mobile station or an external terminal.

The SMS point-to-multipoint service is also called *cell broadcast*. With this service messages are transmitted to all mobile stations that are located in a particular region. This allows messages to be sent to a narrow local area of reference (so-called location aware services).

3.10.3.4 Videotex Access Services

The access of mobile users to the databases of interactive videotex services was considered an appealing service and was therefore standardized. Since all three communications protocols for videotex available in the CEPT countries are not compatible, three access services were specified. These allow an appropriately equipped mobile station to use the videotex service in its own country or one it is visiting.

3.10.3.5 Facsimile Service

This data service provides for the use of standard Group 3 facsimile machines, which transmit digitally in the analogue telephone network and are available today as compatible equipment in all CEPT countries [3]. The GSM specifications relate essentially to ITU-T Recs. T.30 and T.4, with facsimile signalling described in T.30 and image transmission between facsimile units described in T.4. The adaptation of facsimile equipment to the public mobile communications network is executed through a fax adapter both on the mobile side and on the side of the internetworking function (see Figure 3.66).

The signalling and message data received by the modem in the fax adapter of the originating GSM terminal is transmitted to the adapter of the partner facsimile unit (in the MSC/interworking unit) over the GSM traffic channel and sent from there by modem over the fixed network to the corresponding facsimile unit. Different modem procedures are required for the signalling and transmission of messages.

The main tasks of a fax adapter involve monitoring and processing protocol T.30 and, in non-transparent mode, the T.4 protocol as well to ensure that the activities of the facsimile units are reacted to appropriately. Communication between fax adapters is under the control of the respective protocols, which are service-dependent.

The transparent facsimile service The transparent facsimile service was the first one introduced by all network operators. The coded data and associated signalling are transmitted via a transparent bearer service.

A full-duplex traffic channel was selected in order to ensure that synchronization would be fast and secure, although the T.30 protocol provides for

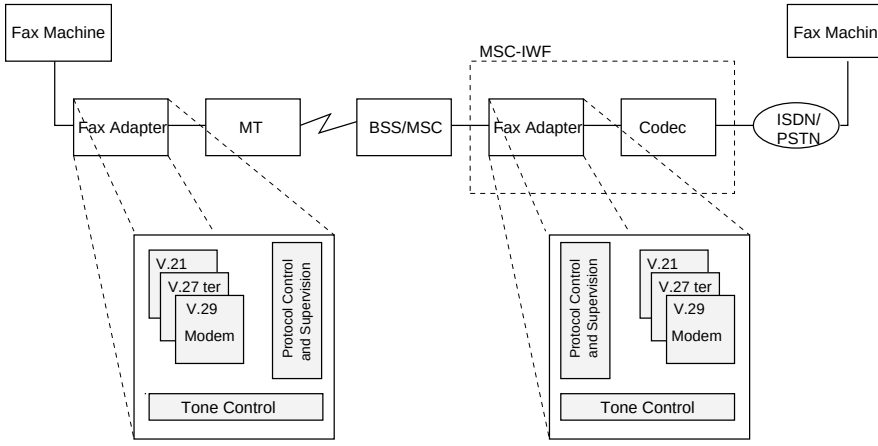


Figure 3.66: Fax adapter function and interworking in GSM

half-duplex transmission. During the signalling phase, in which the data rate is 300 bit/s, additional redundancy can be achieved through retransmission of the bits. During data transmission, document lines can be more affected by strong interference caused by conditions in the radio field than is normally the case in a fixed network.

Aside from the signalling phases, the duration of a fax transmission is the same as in a fixed network. When transmission quality in a transparent service is poor, a facsimile unit reduces the transmission rate. This reduction is registered by the T.30 control and leads to a change in channel operating mode on the radio interface (*Channel Mode Modify Procedure*). The procedure involves selecting a bearer service with a lower transmission rate whereby a change of channel coder is requested in the mobile station as well as in the base station system. In some cases the bit-rate adaptation must also be changed in the IWF.

The non-transparent facsimile service The non-transparent facsimile service requires an adapter to adapt a terminal to the bearer service provided by GSM. The adapter has modem functionality for the attached facsimile unit, and controls and manipulates the facsimile protocol by storing data, transcoding and adapting the facsimile rate to the radio channel quality.

In contrast to the transparent facsimile service, transmission between fax adapters is in half-duplex mode corresponding to the T.30 protocol. For improved transmission quality on the traffic channel, the RLP, which also protects data transfer over the radio interface and between fax adapters, is used in non-transparent mode. Consequently, with a non-transparent facsimile service it is possible to send at a transmission rate of up to 9.6 kbit/s. Use of the RLP also means, however, that the retransmission of transmitted frames

that have been received with errors takes extra time. This results in the need for intermediate storage measures in the transmission chain, which must be compatible with T.30 or T.4.

Fluctuation in radio channel quality, which causes lower transmission rates and requires documents to be stored in the sending facsimile adapter, was considered a problem in the standardization. Therefore in 1994 the service was specified as optional (A). As soon as a timer in the receiving fax unit expires, a transmission is broken off. Simulation tests carried out by the author have shown that these terminations do not occur in normal illuminated cells, because a handover is usually initiated if the receiving quality is poor (7 dB $> C/I >$ 5 dB) (see Section 3.6). Therefore the non-transparent facsimile service represents an interesting alternative to the transparent service, since it provides a higher quality of service with a much lower bit-error ratio.

3.10.3.6 Access to Electronic Mail (E-Mail)

A PLMN does not have *electronic mail* (e-mail) but offers an access service to systems in fixed networks that exist in many CEPT countries based on ITU-T series X.400.

3.10.4 Supplementary Services

Supplementary services are offered to subscribers who use the teleservices and bearer services described above, expanding or supporting these services as follows:

Subscriber identification This function group specifies services that allow for or restrict the identification of the other mobile user. It also provides a possibility for registering unwanted calls, even if the caller does not want to reveal his identity.

Call rerouting There is a difference between unconditional and conditional rerouting of call requests. With unconditional call rerouting, a connection is automatically switched to the line of another user, whereas conditional call rerouting is only possible if, for example, the line is occupied, the subscriber does not respond or the radio network is overloaded.

Call forwarding In contrast to call rerouting, this function supports the forwarding of a connection that already exists.

Call holding: This feature allows a subscriber to maintain an existing connection and at the same time temporarily establish another connection.

Conference call This extends the number of users participating in a connection from two to more. The basis for a successive structuring of the conference call is the supplementary service *call holding*.

Closed user group This function supports the forming of logical subnetworks within the entire GSM. Communication is only possible between the registered subscribers of a subnetwork or of a subscriber group. This service can be used, for example, for setting up a company-specific GSM subnetwork to which only the employees of the company have access.

Call restriction This function group makes it possible to impose either a total restriction or a partial restriction on calls. The restriction can apply to incoming and/or outgoing calls. Examples include:

- Barring outgoing calls (possibly international ones)
- Barring incoming calls, for instance if the user is outside his own home network.

3.10.5 Support for Value-added Services*

Owing to the restrictive standardization of the GSM network, operators have only little possibility to differentiate their service portfolio from their competitor's offering. In many cases the tariffs are the only means to distinguish one GSM operator from another. Value-added services are, however, beyond the boundaries of the GSM standard and offer a way to increase the attractiveness of an operator for a specific group of users. Such services are not treated in the GSM specifications, but GSM provides dedicated mechanisms to facilitate the provision of value-added services. As it is up to the respective network operator to provide that type of a service, the term "operator-specific" services is used as well.

3.10.5.1 Unstructured Supplementary Service Data (USSD)

The MAP-process USSD allows for the communication between a GSM user and a network-based application. In the current phase 2 of USSD both network- and user-initiated USSD dialogues are specified. This communication is transparent to both the mobile station and the intermediate network nodes. Transparency to the mobile station means that the messages sent from the network application are straightforwardly displayed on the handset, and similarly the messages typed in by the user are sent to the network without further processing in the mobile station.

USSD Application The value-added services making use of the USSD mechanism can be referred to as the *USSD Application*. It is a sequence of input/output strings exchanged between the network and the user, possibly triggered by certain events. The USSD application is fully stored in the network, since the handset functions only as a simple man–machine interface.

*With the collaboration of Eckhardt Geulen

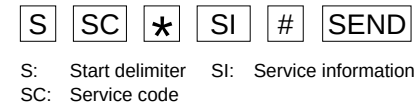


Figure 3.67: Structure of a USSD sequence

This explains why control over USSD dialogues is always with the network-based applications, including in the case of user-initiated USSD dialogues.

USSD applications can either be implemented in the MSC/VLR or in the HLR. Thus it is possible that in the roaming case USSD sequences from the user are either interpreted locally in the visited network or forwarded to the HLR in the Home PLMN. Which of the two methods applies is specified by the string entered by the user and the capabilities of the visited network.

Structure of a USSD Sequence Dedicated key stroke sequences (DTMF signals) are defined for all standard operations like call setup and supplementary service control. Any of these sequences are translated by the mobile station into functional signalling, which means that predefined messages are sent over the air interface in order to invoke the desired funtion. In principle, all key sequences that cannot be interpreted in the mobile station are packed into an USSD message and sent to the NSS. However, to have a certain amount of additional functional signalling available for future use, a format for USSD messages has been defined (see Figure 3.67).

The *start delimiter* consists of one, two or three star or hash symbols. The *service code* consists of two symbols, and first of all identifies the location of the USSD application (visited or home PLMN) and secondly specifies which application has to be started upon reception of the USSD string. The field *service information* can carry service-specific information. Its length can vary between zero and a number of symbols limited by the maximum USSD string length.

An example of a value-added service offered by means of the USSD mechanism is the e-mail service allowing the user to read electronic mails on the mobile station. The USSD sequence entered by the user to activate the service could be, for example, `**13*2#`, where `13` would specify the e-mail service (located in the home PLMN) and `2` could mean to only transmit sender and subject field of the messages received in the mail server. Upon reception of this USSD sequence, the network would send e-mail messages directly to the display of the user's mobile station.

USSD Dialogues The USSD process is dialogue-oriented. Instead of simply sending unrelated messages back and forth, a context is established to monitor proper exchange of messages in an USSD dialogue. The dialogue is in any case controlled from the network side, including in the case of a user-initiated dialogue. In that case the control is taken over by the network as soon as the first

message is received in the node hosting the USSD application. This implies that the mobile station cannot send subsequent requests to one application without being explicitly requested to do so.

The network can send two different types of messages: the reception of a notification is not acknowledged by the mobile station, whereas a request must be acknowledged.

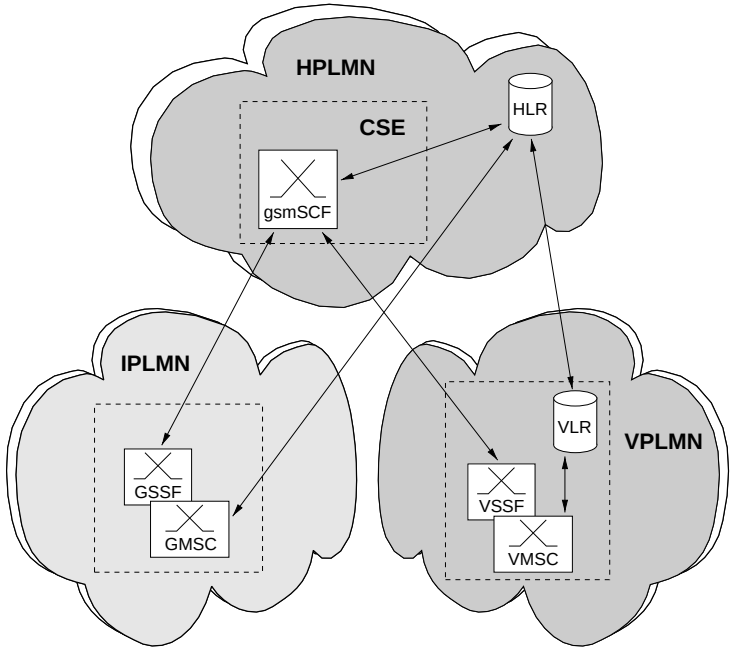
Performance of USSD For a USSD dialogue it does not matter whether or not a traffic connection is established. In both cases USSD messages can be send over the air interface—only the way in which this is done differs. In the case of an established connection the fast associated control channel (FACCH) is used, whereas in the other case USSD messages are sent over the stand-alone dedicated control channel (SDCCH). A fact to be considered for massive use of the USSD mechanism in the case of established connections is that the FACCH is realized by so-called pre-emptive dynamic multiplexing. This means that part of the traffic channel (TCH) is cut out to form the FACCH. Thus the quality of the traffic is lowered.

The performance of USSD is different for the two cases described above. The theoretical throughput for USSD traffic over the FACCH is no more than 1200 bit/s. For the SDCCH the value is 670 bit/s. Owing to interleaving and other mechanisms, there are strong dependences on the amount of data to be transmitted as well as on other parameters. But in any case the figures show that the use of USSD is limited to notification-type services and that USSD cannot be used as a serious bearer for data transport.

3.10.5.2 Customized Applications for Mobile Network Enhanced Logic

The concept of intelligent networks (IN) (see Chapter 15) allows easy deployment of new services in telecommunication networks. In an intelligent network service control logic is stored in a central place (the *Service Control Point*, SCP) and can be used to control many instances of a service in different *Service Switching Functions* (SSP). The standard IN functions are today mostly intended for fixed networks. As more and more operators run both fixed and mobile networks, a requirement has arisen to re-use parts of fixed network IN installations in mobile networks. At the same time, the arguments for the introduction of fixed network IN are valid for mobile networks as well. Finally the operator's wish to provide operator-specific value-added services as described above drove the standardization for a feature called *customized applications for mobile network enhanced logic* (CAMEL).

Introducing IN into GSM The GSM system has been designed without consideration of the IN standard. CAMEL can be seen as a way to introduce IN into GSM. The network model for the introduction of CAMEL is depicted in Figure 3.68.



HPLMN:	Home PLMN	GSSF:	Gateway Service Switching Function
IPLMN:	Interrogating PLMN	VSSF:	Visited Service Switching Function
VPLMN:	Visited PLMN	GMSC:	Gateway Mobile Services Switching Centre
CSE:	CAMEL Service Environment	VMSC:	Visited Mobile Services Switching Centre
gsmSCF:	GSM Service Control Function		

Figure 3.68: The principle of introducing CAMEL into GSM

The CAMEL feature allows the separation of service logic from the mobile switching centre (MSC). Therefore a *Service Switching Function* (SSF) is added to the MSC, and a so-called *CAMEL Service Environment* (CSE) will be introduced within the GSM network. This CSE contains a *Service Control Function* (SCF) similar to a traditional IN solution. With the help of CSEs, *Operator-Specific Services* (OS-Ss) can be offered to customers.

Mobile Originating Calls Subscribers to the CAMEL feature will be marked with a *CAMEL Subscription Information* (CSI). If an active originating CSI is found in the VLR during the call setup of a mobile station, the *Visited Service Switching Function* (VSSF) sends an `InitialDetectionPoint` message to the *GSM Service Control Function* (gsmSCF) and the VMSC suspends the call processing. According to GSM Specification 03.78 [28], the `InitialDetectionPoint` must always contain

- the service key

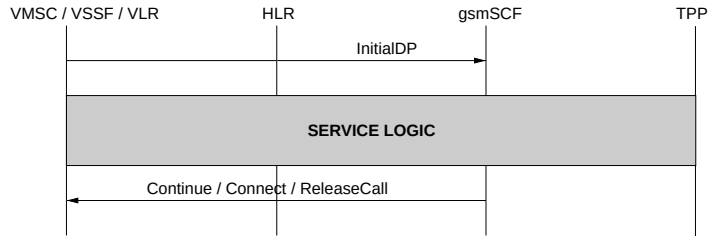


Figure 3.69: Signalling procedure for mobile-originating calls

- called and calling party numbers
- calling party's category
- location number
- bearer capability
- event type *Basic Call State Model* (BCSM)
- location information
- the international mobile subscriber identity (`IMSI`)

After the service logic has been processed, CAMEL-specific handling is initiated from the gsmSCF.

The signalling procedure in the case of mobile-originating calls is illustrated in Figure 3.69. A Third Party Provider identified by the number n (TPP n) might then be involved.

Mobile Terminating Calls In the case of a mobile-terminating call, the gateway MSC (GMSC) in the interrogating PLMN identifies the HLR of the called party with the help of the mobile station international ISDN number `MSISDN`. Then the GMSC sends a `RoutingInformationRequest` to the HLR. The HLR checks the CSI of the called party and sends the information stored in the subscriber record back to the GMSC. Now, the GMSC acts according to CSI. If the terminating CSI is active, the call processing is suspended as soon as the trigger criteria of a *Detection Point* (DP) of the CAMEL service logic is fulfilled. An `InitialDP` message is sent to the CSE and the service logic execution is started. Thereafter, CAMEL specific handling is initiated.

Figure 3.70 shows the signalling procedure for mobile-terminating calls.

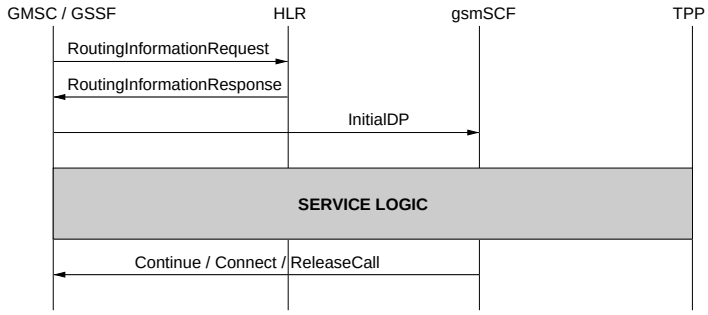


Figure 3.70: Signalling procedure for mobile-terminating calls

3.11 Future Voice and Data Services in GSM*

The data services introduced by GSM in mid-1994 are based on circuit-switched transmission. As with voice transmission, each user is provided with an exclusive connection over a TCH. Since data sources often have fluctuating traffic volumes, circuit switching results in an inefficient utilization of the radio channels.

Owing to the continuing strong growth projected in the number of mobile subscribers, frequency economy as well as flexibility in the use of radio channels is becoming more and more important. Internet services in particular are to be offered at a favourable cost to the mobile terminal.

Current data services offer a maximum data transfer rate of 9.6 kbit/s and are not able to meet the needs of many applications. From the standpoint of the mobile subscriber, there is also the matter that charging is based on the duration of a transmission and not on the amount of data transmitted.

In principle there are three options for implementing new data services in GSM that would offer a higher data transfer rate than 9.6 kbit/s:

- High-bit-rate circuit-switched data services
- Packet-oriented data services with variable bit rates
- Multimedia data services

High-bit-rate circuit-switched data services are based on the parallel use of several traffic channels. A maximum data rate of 57.6 kbit/s can be achieved by using four time slots (TCHs) of a carrier frequency. This approach is being standardized by ETSI under the heading of *High-Speed Circuit Switched Data* (HSCSD) [19]. The implementation of this service requires that changes be

*With the collaboration of Götz Brasche

made in channel allocation, connection setup, handover procedures and access to the fixed network (*interworking*). This function became commercially available in the end of 1998.

In contrast to the HSCSD service, a *packet-oriented data service* concept not only provides a high data rate but also offers flexibility in the use of channel capacity for applications with variable bit rates because of the possibility to multiplex several connections at the same time on the same traffic channel or on several traffic channels used in parallel. A maximum data rate of 115 kbit/s is aimed at. Because GSM transmits using circuit-switching, significant modifications would be required in order to integrate a packet-switched service. This technique will be commercially available in 1999 and will be carried over to the UMTS system later; see Section 5.1.

The *Multimedia data service* is a circuit-switched service, and will extend the GSM services to that expected from 3rd-generation systems. Under the acronym *Enhanced Data rates for GSM Evolution* (EDGE), a technique totally compatible with the GSM channel spacing of 200 kHz is being studied at ETSI/SMG. A modified air interface with especially a new modulation method applied to the 200 kHz channel as a whole is expected to reach 384 kbit/s per carrier. EDGE was initiated by Ericsson, and will be applicable to all the existent GSM frequencies such as 900, 1800 and 1900 MHz. The related products and services are projected to be available by 2000.

In addition to new data services, future applications will also require new voice services (e.g., group communication), which until now have only been offered in trunked radio systems.

Since 1994, at the instigation of the International Railway Federation (*Union des Chemins de Fer*, UIC) under the heading *Advanced Speech Call Items* (ASCI), group and broadcast services with fast call setup and priority control in the list of GSM Phase 2+ working points are being run and processed.

3.11.1 ASCI—Advanced GSM Speech Call Items

Currently in Europe the national railways are using many different non-compatible radio systems. National railcars need separate communication systems for national and for international rail services. Therefore the development of a standard European Train Control System (ETCS) is underway under the overall control of the UIC. Although the GSM radio channel is only specified for a maximum terminal speed of 250 km/h, field studies have shown that no noticable restrictions occur even at speeds of 300 km/h.

The new communications platform will be called GSM-R(ailway) and will be operated directly below the GSM expansion band.

Group and broadcast radio calling with fast connection setup must be integrated into GSM in order to meet the requirements of UIC train radio control. Both services are being enhanced with a priority control mechanism adopted

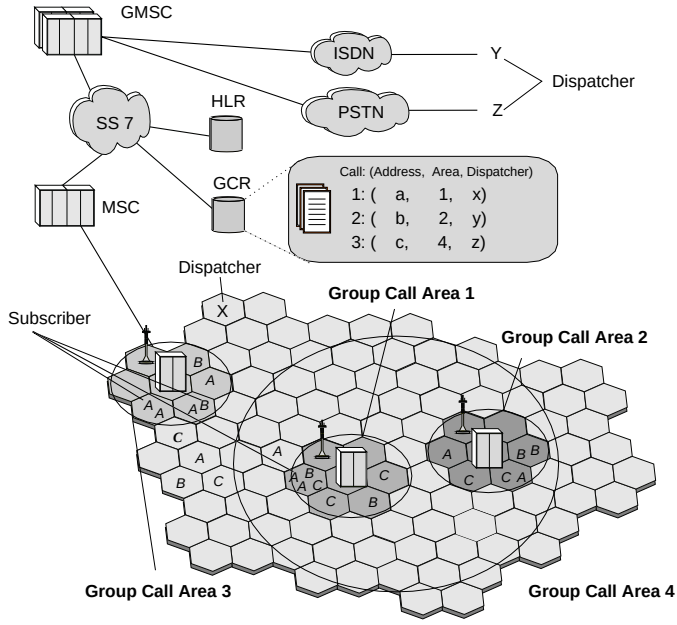


Figure 3.71: Logical concept for expanded voice services

from ISDN. The ETSI standard designs consequently comprise specifications for:

- *Enhanced Multi-Level Precedence and Pre-Emption* (eMLPP) [21, 25]
- *Voice Group Call Service* (VGCS) [23, 27]
- *Voice Broadcast Service* (VBS) [22, 26]

3.11.1.1 Voice Broadcast Service

The *Voice Broadcast Service* (VBS) allows users of the mobile radio and fixed networks to send messages to several so-called *listeners*. Figure 3.71 provides a clarification of the logical service concept.

If a broadcast call is initiated by a mobile station (see Figure 3.72), the identity of the appropriate cell with the requested group identity is routed to the *Group Call Register* (GCR) of the applicable MSC.

With a call initiated from the fixed network the appropriate user and requested group identity is transmitted to the GCR. The GCR then sends back a list of cells in which the call is to be broadcast according to the make-up of the group and the location. The responsible MSC routes the list to the appropriate MSCs. The MSCs instruct the appropriate BSCs to set up a broadcast channel in each of the relevant cells and to send out a call *notification* on a

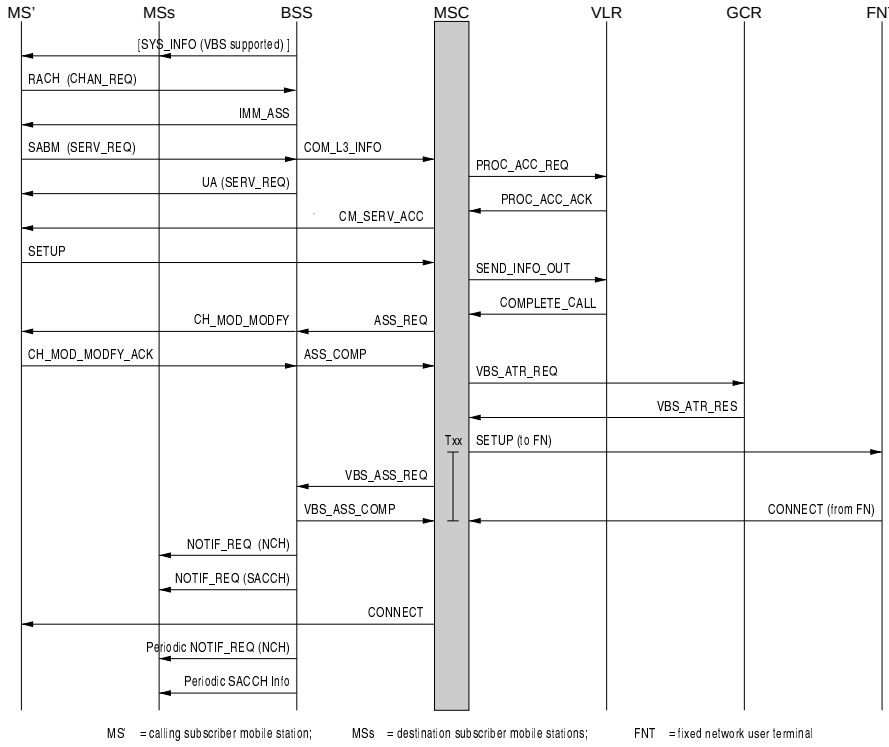


Figure 3.72: Call setup in the voice broadcast service

newly specified signalling channel. In contrast to the conventional GSM use of voice, this process is not described as *paging* because the mobile stations are not explicitly being addressed and are not responding to the call notification. In the corresponding GSM Rec. 08.58 [24] these GSM control channels have been expanded to include the *notification common control channel* (NCCH).

A notification is sent out at periodic intervals until the call has been completed. Mobile stations that receive a call notification switch over to the broadcast channel indicated and listen in on the appropriate downlink. The call initiator remains on his dedicated channel during the course of the call, and completes the call after having transmitted his message.

Accordingly, a broadcast call can be established and managed in the same way as a GSM point-to-point connection, except for the additional signalling required for the rerouting. This means that no additional handover procedures are required during an impending cell change.

This does not apply to the mobile stations participating in the broadcast call, although a so-called *idle mode cell reselection* algorithm can be used for them.

The way to prevent a mobile station from moving into a cell in which the broadcast call is not being transmitted is by not including the cell in the list. Signalling requirements are minimized through the fact that the mobile stations are only notified of the frequencies of the signalling channels of the surrounding cells, i.e., the frequencies on which the call notification is being sent. The mobile stations must then listen in on the corresponding control channels in order to determine the actual broadcast channel.

3.11.1.2 Voice Group Call Service

The Voice Group Call Service (VGCS) supported by ASCI provides a service that allows the fixed network or the mobile stations to set up a group call channel on which the group members can listen in or also transmit. After the call indicator has sent his message, he releases the channel and transfers over to *listener mode*.

As with VBS, the addressee group of VGCS is divided into the following:

- Mobile stations that are members of the group and are located in a predefined geographical area
- A fixed group of fixed network stations

As soon as all the group call participants finish talking, any one of the users can request to be allocated the channel. If the channel is allocated, the user is allowed to transmit until he releases the channel and in turn changes to listening mode.

A broadcast call is normally explicitly ended by the initiator. A connection between initiator and network that is disrupted owing to interference cannot be detected immediately, because the initiator could be lingering in listening mode during a call. This only becomes clear if there is a pause in conversation of a defined length after which the group call is automatically terminated by the network. Therefore call termination with a group call requires the *Voice Activity Detection* (VAD) function.

3.11.1.3 eMLPP—A Priority Control Service

The ASCI priority strategy is derived from the *Multi-Level Precedence and Pre-Emption* (MLPP) scheme [31] used in SS 7, and correspondingly will be introduced into GSM under the name of *enhanced Multi-Level Precedence and Pre-Emption* (eMLPP). Whereas MLPP defines a five-level priority, GSM is planning seven priority classes (see Table 3.33). The MLPP priorities 0–4 will correspond to the eMLPP priorities 0–4.

In addition to these five classes, two further classes, A and B, have been specified for the GSM system (see Table 3.33). They are exclusively being reserved for processes within the network, e.g., for the configuration of group and broadcast calls VGCS and VBS described above.

Table 3.33: Priority classes in eMLPP

Class	Use	Call interruption	Comments
A	Operator	Yes	Highest priority
B	Operator	Yes	
0	Subscriber	Yes	
1	Subscriber	Yes	
2	Subscriber	No	
3	Subscriber	No	Standard priority
4	Subscriber	No	Lowest priority

Calls with priority A or B can only be made locally—in other words only within the coverage area of an MSC. If this kind of priority is used globally, in other words a GSM call is routed over an ISDN network, the priority classes A and B are changed to priority 0.

The maximum priority allocated to a mobile user, and reflected in the monthly basic tariff, is negotiated with the service provider when the contract is signed and is stored in the SIM card.

3.11.2 HSCSD—The High-Speed Circuit-Switched Data Service

The High-Speed Circuit-Switched Data Service (HSCSD) simultaneously allocates several full-rate traffic channels (TCH/F9.6) to a mobile station within a 200 kHz frequency channel for the duration of a transmission.

With the parallel use of all eight time slots, and depending on the bearer service used, data rates of up to 76.8 kbit/s are achievable on the basis of TCH/F9.6 coding in conformance with GSM (see Table 3.8). (Costly transmitting and receiving facilities are required in mobile stations if more than four channels are used at the same time. The number in the standard is currently restricted to four channels reaching 57.6 kbit/s with different coding.)

3.11.2.1 The Logical Architecture

In the current GSM the functions needed for data transmission are mainly embedded in the *terminal adaptation function* (TAF) of the mobile station and in the *interworking function* (IWF) of the MSC. In principle this functional division has been retained in the HSCSD service (see Figure 3.73). Essentially only a *splitting/recombining* function needs to be added to the above components of the MS or MSC when several traffic channels are used simultaneously.

Logically only *one* connection exists between MS and MSC. The segmentation and reassembly is therefore based on the consecutive numbering of the individual data frames.

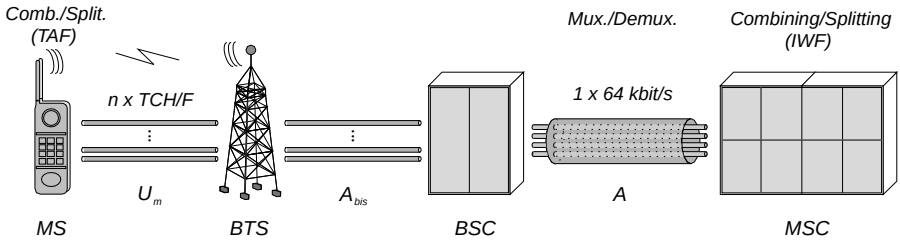


Figure 3.73: Architecture of an HSCSD

The time slots used on the radio interface at the A_{bis} reference point between BTS and BSC are mapped transparently. The HSCSD channels are then multiplexed onto a 64 kbit/s connection at the A reference point (or E reference point between two MSCs).

3.11.2.2 The Radio Interface

At the radio interface an HSCSD connection can consist of up to eight traffic channels (TCH) (*multi-slot assignment*, MSA). All the channels in an HSCSD connection use the same frequency hopping procedures and the same training sequence. For security reasons, however, separate enciphering is being planned for each channel. The channel coding, interleaving and rate adaptation of the current traffic channels will be retained in order to keep the implementation costs to a minimum. Each subchannel is allocated an SACCH. This provides individual transmitter power control and improves the interference level.

Time-oriented synchronization begins after time slot 0. The *idle* TDMA frames are therefore not used in a 26-frame multiframe, so that synchronization can be carried out with the neighbouring cells. Each HSCSD connection only has one *fast associated control channel* (FACCH). This channel is designated as the *main HSCSD subchannel* (MHCH).

3.11.2.3 Bearer Services

In accordance with the current GSM bearer services, transparent and non-transparent bearer services are supported in the HSCSD service (see Figure 3.64). The transparent service guarantees a constant data rate, even with fluctuating quality of service. The number of allocated channels can be increased if the quality of service rate falls below a threshold value. The non-transparent bearer service, in contrast, guarantees that quality of service will remain constant with fluctuating throughput.

Transparent bearer service The transparent bearer service uses the ITU-T X.30/V.110 protocol [30], which provides a three-level rate adaptation to the user interface (R- and S-interface) (see Figure 3.61).

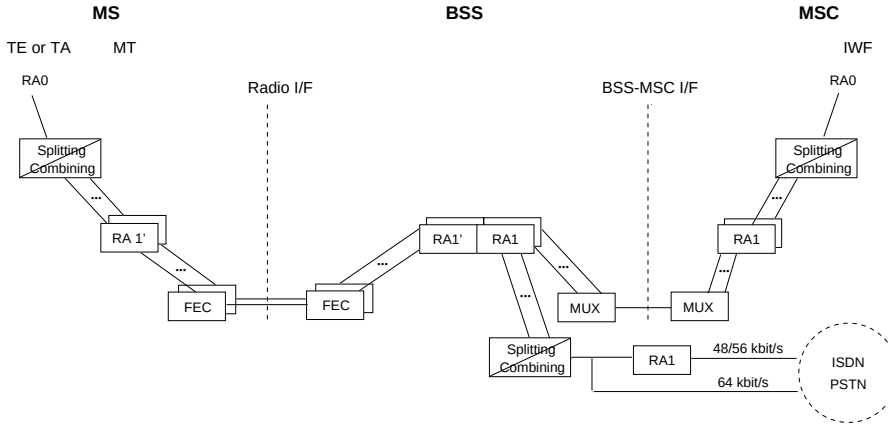


Figure 3.74: The non-transparent HSCSD bearer service

The time displacement of the individual HSCSD channels between TAF and IWF does not have a considerable effect on frame length. In practice, a data frame that is being sent in one of the channels cannot overtake a frame being sent in another channel. Nevertheless there must be a safeguard to recognize that the frames are being transmitted in the correct sequence.

In GSM the status of the V.24 interface used for data transmission at the network gateway is transmitted in the status bits `SA`, `SB` and `X`. Because more than one subchannel can be used for a logical connection in the HSCSD service, the redundant status bits can be used for the numbering of the channels without causing a reduction in the repetition rate of the status bit per connection. An extra bit can be used for a modulo 2 numbering within the subchannels in order to prevent problems caused by the time displacement of the subchannels. Rate adaptation for data rates that are not multiples of 9.6 kbit/s can be effected through a corresponding number of filler bits in the last four V.110 frames.

Non-transparent bearer service Like the corresponding bearer service in GSM, the non-transparent HSCSD bearer service is based on the RLP protocol (see Section 3.9.3).

According to the service concept, an RLP entity will manage all the subchannels (see Figure 3.74), because, with the numbering only based on one transmitting and receiving window, any number of RLP frames can be sent on the subchannels.

A maximum window size of 61 is allowed for a conventional RLP. A total of 6 bits is sufficient for numbering each transmitting or receiving window. In a non-transparent HSCSD service a maximum of 8 traffic channels can be allocated to an RLP entity. The addressing space must be increased correspondingly for the management of the transmitting or receiving window. An

additional 6 bits are required in the RLP header for the management of the transmitting/receiving window; this reduces the data rate by 3 % (Since it is not necessary for each RLP frame to use the status byte of the *Layer-2 Relay* function, L2R, the maximum data rate of a subchannel of a 9.6 kbit/s subchannel can be maintained).

Pseudo-asymmetrical transmission A pseudo-asymmetrical transmission mode is planned for the non-transparent bearer service. This allows mobile stations with half-duplex transmitting and receiving facilities to receive in multi-slot mode but to transmit in single-slot mode. This will result in a higher data rate—at least on the downlink. A mobile station will decide which of the allocated lower channels is to be used in the uplink direction and which are to remain free. As in discontinuous transmission mode (DTX), the frames are regarded by the network as not being used and are rejected by the IWF. Although not specified in the standard, with additional signalling the time slots that are not being used can be made available to other services.

3.11.2.4 Signalling

There is a greater chance of blocking with the HSCSD service because parallel traffic channels are being occupied at the same time.

In the current GSM a bearer service exists for each data rate. Depending on the configuration, different variable data rates will be possible once the HSCSD service is introduced. It is not useful to define another bearer service for each possible data rate. Therefore the data rate for the HSCSD service is merely considered a quality of service parameter. This *flexible bearer services* (FBS) concept provides for a desirable and required data rate (Desired number of channels (DNC) or Required number of channels (RNC)). The desired data rate represents the maximum and the required data rate the minimum data rate necessary to maintain the quality of service required by an application.

Depending on radio channel availability, a mobile station can be allocated any data rate between a desired and a required data rate. This concept promises a low blocking rate, particularly with handovers.

If a target cell does not have a sufficient number of free channels available during a handover, the FBS concept can be applied to maintain the connection as long as the target cell has enough channels for the required data rate. The number of blockings can be reduced when there is an overload, since channels can be released up to the minimum data rate. The two parameters are transmitted in the Setup message when a connection is established. The so-called `Bearer Capability` element (BCE) is expanded accordingly. When transmission is carried out with a fixed data rate, the parameters for desired and maximum data rate have the same value.

In addition to these two new parameters, the characteristics of the mobile station in terms of its multi-slot assignment (MSA) capability and the channel coding supported (*class mark*) must be communicated in the BCE.

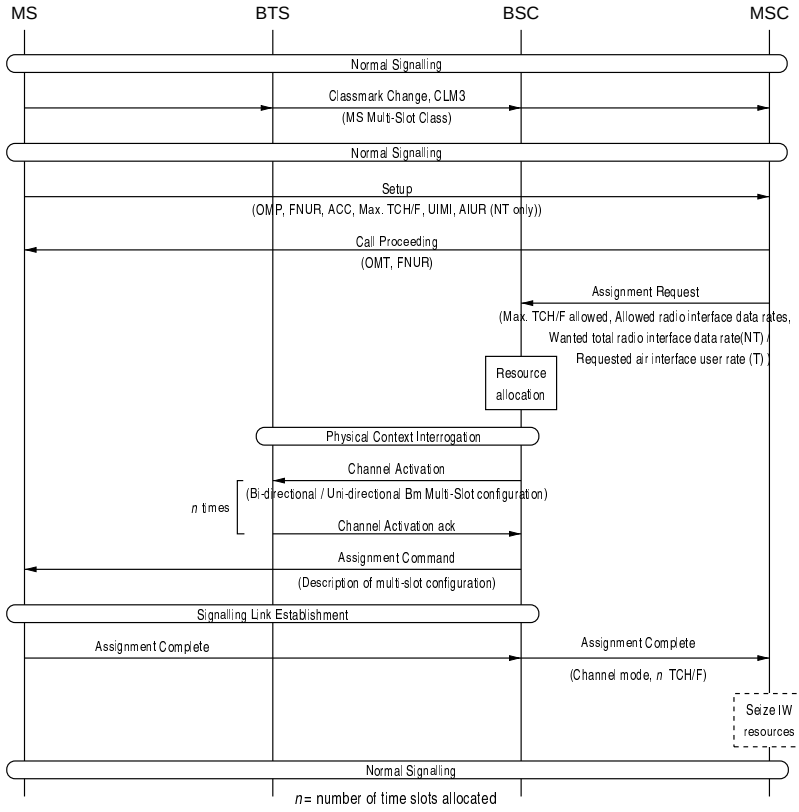


Figure 3.75: Call setup procedure in HSCSD

Figure 3.75 illustrates the call setup procedure. After receipt of a Setup message, the MSC sends a modified Assignment_req message to the BSC. This message contains a list of channels at the A-interface in the `Message Content` element. The parameters `DNC` and `RNC` are added to the `Channel Type` element. Before this message can be transmitted, the MSC reserves the required capacity. If the requested number of radio channels cannot be allocated, a Channel_Update procedure is used between MSC and BSC in order to adapt the resources to the A-interface.

A separate Channel_Activation message is required for all lower channels before the selected channel configuration as well as information on channel coding can be forwarded in an Assignment_Command message to the MS. In this message the time slot numbering parameter indicates the first of the consecutively allocated time slots. The `Channel Description` element accepts the time slots.

3.11.3 GPRS—The General Packet Radio Service

Within the framework of the continuing development of GSM Phase 2+, ETSI has recently been working on a packet-oriented service concept for the transfer of data. Standardization of the new service *General Packet Radio Service* (GPRS) was scheduled to be completed in 1997 (see Appendix E, Table E.2 for an overview of the GPRS standards).

3.11.3.1 General Characteristics of GPRS

A major consideration in the introduction of the new service is the anticipated acceptance by mobile radio subscribers, who will be particularly affected by the cost of the service in addition to the quality of service. Therefore one of the development premises was to modify existing GSM components as little as possible and to develop the new service based on current teleservices and bearer services. First proposals on this type of service have been published in [2, 34, 37, 36].

GPRS offers a packet-switched service alongside existing circuit-switched data services, eliminating the need to replace current GSM services (although the possibility of transmitting packetized voice using GPRS is not precluded in the standard, it has not been addressed in the current specifications).

The area of application can be divided into a horizontal and a vertical market segment:

- Horizontal markets:
  - wireless personal computers
  - mobile office
  - electronic money transfer at the time of transaction
- Vertical markets:
  - traffic reports
  - fleet management
  - goods/supply logistics

The GPRS specifications do not set an upper limit to the data set that can be transmitted per access. However, the service is primarily adapted to:

- frequent regular transmission (several times per minute) of small quantities of data up to 500 bytes
- irregular transmission of small to medium-sized quantities of data up to several kbytes

Three different load models for identifying the typical applications have been defined by ETSI.

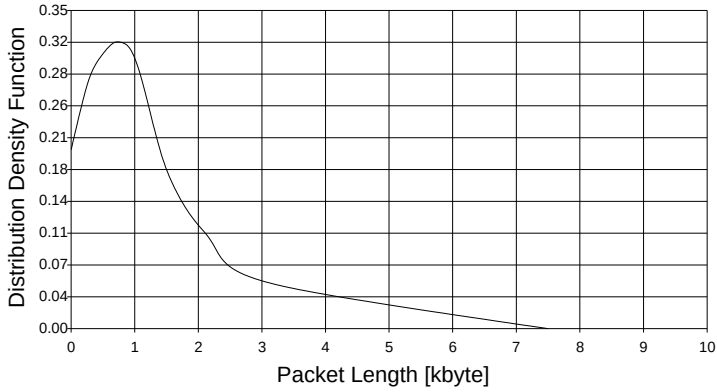


Figure 3.76: The distribution density function of the FUNET load model

FUNET (Finnish University NETwork Operations Centre) load model This model is based on a statistical evaluation of the distribution of the length of electronic messages (e-mail), which is approximated by a Cauchy distribution density function with a maximum message length of 10 kbytes and a mean value of 100 bytes (see Figure 3.76):

$$\text{Cauchy}(0.8.1) = f(x) = \frac{1}{\pi[1 + (x - 0.8)^2]} \tag{3.8}$$

Mobitex load model This model is based on a statistical evaluation of the fleet management application in the Ericsson Mobitex System. Data packets with a uniform distribution length of 30 ± 15 bytes are sent on the uplink. The packet length on the downlink are at 115 ± 57 bytes.

Railway load model This model describes the anticipated distribution of packet lengths in train control applications through a curtailed negative exponential distribution with an average packet length of 256 bytes and a maximum length of 1000 bytes:

$$F(x) = 1 - e^{-x/256} \tag{3.9}$$

Service types The PLMN network operator is responsible for the transmission of data between the respective service access points in the fixed network and the mobile station. Two service categories are available:

Point-to-point (PTP) With this service individual message packets can be transmitted between two users. The PTP service will be offered in

Table 3.34: Service type according to service initiator

	Point-to-point		Point-to-multipoint	
Message flow	PTP-CONS	PTP-CLNS	PTM-M	PTM-G
Fixed network – MS	×	×	×	×
MS – MS	×	×	×	×
MS – Fixed network	×	×	—	×

connection-oriented mode (*Connection-Oriented Network Service*, PTP-CONS) as well as in connectionless mode (*Connectionless Network Service*, PTP-CLNS). The applications drafted on the PTP service can be divided into the following groups based on their communication characteristics:

- *Non-dialogue traffic* There is no relationship between the individual data packets.
- *Dialogue traffic* A logical relationship exists between the service users for a certain period of time, which can extend from several minutes to several hours.

Point-to-multipoint (PTM) This service allows data packets to be transmitted between a service user and a group specified by this user within a particular geographical area. The PTM service is subdivided into

- *Multicast (PTM-M)* A multicast multipoint communication combines the calls that are broadcast in an entire area defined by the call initiator, whereby either all users or only a group is addressed.
- *Group call (PTM-G)* The messages are addressed exclusively to a specific group, and are only sent to areas in which the group members are located.

Table 3.34 shows that the services, with the exception of the multicast service, can either be initiated by the mobile station (*mobile-originated*) or by the fixed network (*mobile-terminated*).

Quality of service Different service profiles are being planned within the framework of GPRS and will be established on the basis of the quality of service parameters *QoS class* and *throughput*. Table 3.35 presents an overview of the defined quality of service classes and the corresponding delay characteristics.

A packet delay is defined as the transmission time between the GPRS quality of service points. Delays outside the system, e.g., through transit networks, are not considered. Along with the quality of service class, the

Table 3.35: Quality of service parameters

Packet length:	128 bytes		1024 bytes	
QoS class	Av. delay [s]	95 % [s]	Av. delay [s]	95 % [s]
1 (predictable)	0.5	1.5	2	7
2 (predictable)	5	25	15	75
3 (predictable)	50	250	75	375
4 (best effort)	Uncertain		Uncertain	

maximum throughput (*peak bit rate*) as well as the average throughput (*mean bit rate*) can be agreed in bit/s; see Table 3.35.

Data services generally require a low residual bit error ratio. Faulty data is usually worthless, whereas speech transmitted with errors gives the impression of poor reception. Unlike the quality of service parameters just mentioned, the residual bit error ratio is not something that can be selected; it is dependent on the current receive situation of the MS.

For group calls, error ratios of 10^{-9} are defined for packets that are duplicated or lost or arrive out of order. The residual bit-error ratio for the multicast service varies between 10^{-4} and 10^{-5}.

Parallel use of service Circuit-switched services (voice, data) can be initiated and used during a GPRS session. There will also be a provision for other services, such as sending/receiving GPRS data during a telephone conversation. The parallel use of these services is planned for PTP as well as for PTM services, but requires varying transfer rates dependent on load and quality of service.

It is also possible to use the *mobile-originated* and *mobile-terminated* Short-Message Service (SMS). However, depending on the particular load situation, the SMS message is delayed or transmitted at a lower data rate. An SMS cell broadcast message can take place simultaneously with GPRS services but not with a circuit-switched connection.

Three equipment or user classes offering end users a packet data service with different characteristics are being proposed:

User class A Simultaneous use of all services according to the service profile and with uniform quality of speech.

User class B Restricted simultaneous use of services with a lower transfer rate and with reduced quality of speech.

User class C No simultaneous use of services.

However, the network will provide a possibility for user class C to receive SMS messages at any time.

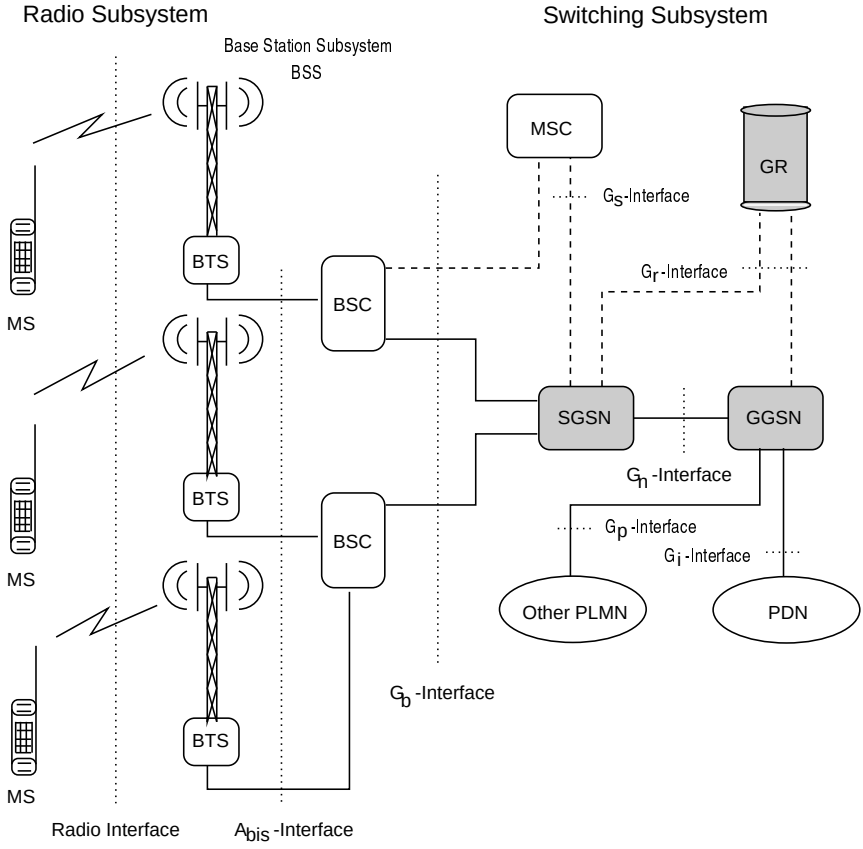


Figure 3.77: The logical architecture of the GPRS

3.11.3.2 The Logical Architecture

The familiar GSM network architecture is expanded by three network elements for the packet data service (see Figure 3.77). The *Gateway GPRS Support Node* (GGSN) serves as the interface to external networks. This is where the packet data protocol addresses are analysed and converted to the `IMSI` of the respective mobile station. The data packets are decapsulated and, according to the options offered by the network protocol, sent to the next entity of the network layer.

The *Serving GPRS Support Node* (SGSN) provides the mobile stations with functional support. This is where the addresses of the subscribers of a group call are requested from the *GPRS registers* (GR). The functions of SGSN and the GGSN can be implemented in one unit.

All GPRS-related data is stored in a *GPRS register* (GR), which is regarded as part of the GSM-HLR.

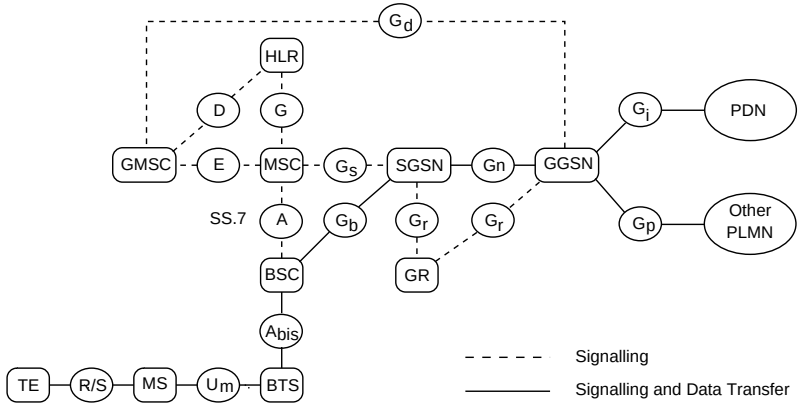


Figure 3.78: Interfaces and reference points in GPRS

The interfaces The expansion of the existing GSM network to include the GPRS-specific blocks has also resulted in new interfaces and reference points being defined. Each GPRS PLMN has two access points: the radio interface U_m for access by the mobile stations, and the reference points R and S for transmitting and receiving messages. The interfaces relevant for GPRS are shown in Figure 3.78. The dashed lines indicate that only signalling data can be exchanged between the corresponding blocks. The continuous lines, on the other hand, mean that packet data may also be transmitted.

The actual data traffic is handled over the SGSN in GPRS. The MSC is used for signalling only. The relationship between SGSN and MSC has not yet been fully defined, but it is conceivable that each SGSN will be assigned to an MSC. Another possibility is to supply the area of several MSCs—in other words to define separate coverage areas for the SGSNs that are largely different from those of the MSCs. Economic considerations play a large role in the division of functionality between MSC and SGSN.

Routing and mobility management The key planning decisions for a packet radio network depend on the implementation of mobility management and path selection (*routing*).

In transmission initiated by an MS, the SGSN decapsulates the arriving packets, evaluates the address information and routes them to the appropriate GGSN which then initiates the routing to the correct *packet data network* (PDN) (see Figure 3.79). The network-specific routing procedures are then used by the relevant PDN for routing the packets to the partner entity.

The packets of the partner entities are routed through the PDN to the GGSN after the destination address has been evaluated. The GGSN checks the routing context to which the destination address is assigned and queries

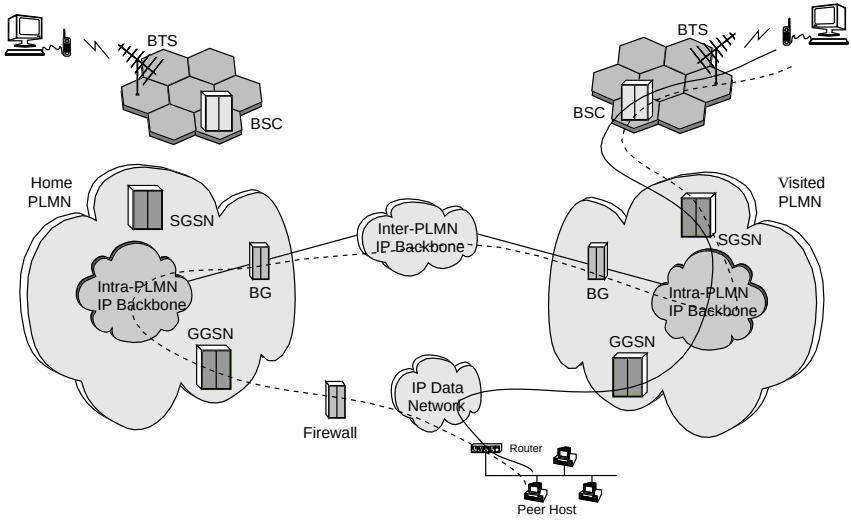


Figure 3.79: Simplified example of routing

Table 3.36: GPRS context

Parameter	Storage location
Mobile station status (`Active`, `Idle`, `Standby`)	MS + SGSN
Authentication (ciphering key)	MS + SGSN
Compression support (yes/no)	MS + SGSN
Routing data (`TLLI`, RA, `Cell ID`, PDCH)	MS + SGSN
Mobile station identity (`IMSI`, `IMGI`)	SGSN
Gateway GSN address (IP address)	SGSN
Call charging parameters (byte number)	SGSN

the corresponding SGSN and associated tunnel information. The packet is then encapsulated and tunnelled to the SGSN, which routes it to the MS.

Two different packet-encapsulating schemes are used. Between the GGSNs, the packets are encapsulated through the use of a GPRS-network-wide standard tunnel protocol. This allows any *packet data protocols* (PDP) to be used, even if they are not supported by all SGSNs. The encapsulation between MS and SGSN separates layer-2 protocols from network-layer protocols.

Mobility management Before a mobile station can send data, it must check into the packet data service. This so-called *attachment procedure* executed between MS and SGSN establishes a logical connection context. The result is that the MS is allocated a unique *temporary logical link identity* (`TLLI`). After successful check-in one or more routing contexts for one or more PDPs

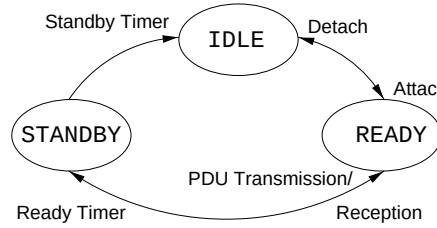


Figure 3.80: State model for mobility management

can be negotiated with the SGSN (see Table 3.36). Equivalent to the current GSM, a *ciphering key sequence number* indicates how the user data is encoded. The GPRS register is interrogated to check whether an MS has access to the respective PDN. Information on contracted services also contains the related GGSN addresses. If access is allowed, the GGSN is requested to update its routing context. All functional GPRS entities thereby agree on each entry to the context before a packet data service can be requested.

Agreements are constantly updated during a GPRS session. The location of an MS is monitored on the basis of the state diagram shown in Figure 3.80. Whereas the MS informs the SGSN of each cell change when it is in state `Ready`, the location information in state `Standby` is only updated if there is a change in *routing area* (RA). This area represents a subset of the location areas defined in GSM. The number of associated cells can be freely defined by the network operator.

A possible cell change is implicitly managed at the level of a logical connection. The area information is updated through the transmission of a Routing_Update_req to the SGSN. This message contains the designation of the new and the original cell as well as the new and the old RA.

In the event that the SGSN is managing the new RA as well as the old RA (*intra-RA update*), there is no need to inform other network elements (GGSN or GR), because the routing context has not changed.

If the original RA is allocated to a different SGSN, an *inter-RA update* must be carried out: the new SGSN sends a request to the old SGSN to transmit the mobility information and routing context for the MS. After the updating has been completed, the old SGSN arranges for the outdated context information to be deleted from the GR and the GGSN can be informed of the new context.

3.11.3.3 The Protocol Architecture

The functions of the packet data service are shown in the protocol architecture in Figure 3.81. Depending on the network environment, existing standard protocols are used as the network layer protocol in the mobile station or in

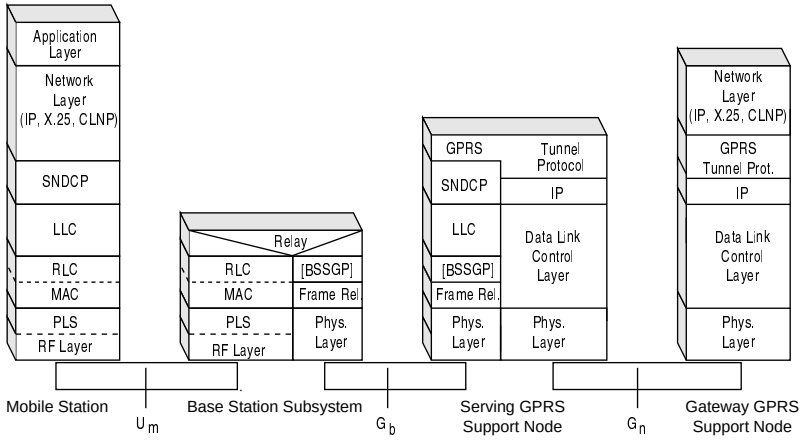


Figure 3.81: GPRS stack of protocols

the GGSN. An interworking with TCP/IP, ISO 8348 CLNP and X.25-based networks is planned for GPRS.

The general transmission principle is based on all data packets from and to external networks within the GPRS backbone network being transmitted in encapsulated form: with the use of a *GPRS tunnel protocol* (GTP), the encapsulated data packets and relevant signalling information are transmitted between SGSN and GGSN. This carries out the interworking function between the GSM and fixed network-specific network protocols. The *Internet Protocol* (IP) is being proposed as the GPRS backbone protocol in Version 6.

The fixed network protocols for the data link and physical layers at the G_n-interface between SGSN and GGSN will use standard protocols.

Between MS and SGSN in the lowest sublayer of the GSM network layer, the *Subnetwork Dependent Convergence Protocol* (SNDCP) is used; this can multiplex several network layer connections on a virtual data link layer connection, and offers coding and data compression algorithms.

The *Base Station Subsystem GPRS Application Protocol* (BSSGP) at the G_b-interface corresponds to BSSMAP familiar from GSM, and essentially provides the same functions. A modified frame-relay technique is applied at the G_b-interface on the data link layer to minimize delays.

The basic idea of the packet data service is to multiplex data of one or more users on a physical channel in order to use its capacity efficiently. It is therefore necessary to divide the data link layer at the U_m-interface into two sublayers: *radio link control* (RLC) and *medium access control* (MAC).

The RLC sublayer provides services that guarantee a reliable logical connection between MS and BSS, whereas the MAC sublayer controls access to the radio medium. The general approach of integrating the packet data service into GSM consists of reserving certain channels from the pool of available

physical GSM channels for the packet data service and dividing them into logical channels.

The security of the user data between MS and SGSN is handled by a separate *logical link control* (LLC) sublayer.

The LLC sublayer The LLC sublayer is responsible for the transport of data packets of the network layer between the mobile station and SGSN. Point-to-point as well as point-to-multipoint communication is supported. The key functions are flow control and error correction using the familiar ARQ and FEC mechanisms. The protocol for the LLC sublayer will be strongly based on the *Link Access Procedure on the D-channel* (LAPD_m) used in GSM and—following current terminology—will be identified as LAPG, for *Link Access Procedure on the G-channel*.

The key modifications can be summarized as follows:

Variable frame length The GPRS protocol architecture with transmission protected at the radio interface by the RLC/MAC protocol allows a variable frame length at the LLC level. Therefore frame delimiters and bit stuffing are not necessary. An additional field is required in the frame header for specification of the frame length.

Variable address types New address types of different lengths are being introduced with the packet data service. The address field must therefore have a variable length, which is controlled by an expansion bit in the address field. Each *Data Link Connection Identifier* (`DLCI`) consists of a *Service Access Point Identifier* (`SAPI`) and a *Terminal Endpoint Identifier* (`TEI`), which can contain addresses such as the `TLLI`, `TMSI`, `IMSI` and `IMGI` *(international mobile GPRS identity)*. The address types are allocated during the checking-in process and are controlled by the management entities.

Prioritized `SAPIs` The priority classes in GPRS are considered in the introduction of new `SAPIs` with priorities. Corresponding to the four proposed classes, at least four `SAPIs` are required. In accordance with the priority scheme based on the expanded voice services (eMLPP) (see Section 3.11.1.3), it is conceivable that other `SAPIs` will be reserved for signalling or emergency calls. With the introduction of a priority system, it will also be necessary for flow control and error correction to be controlled on a priority basis.

Expanded operation mode The protocol supports duplex communication for the acknowledged point-to-point service, and also operates in *asynchronous balanced mode* (ABM). Since LAPD only supports PTP communication, the protocol state machine for the point-to-multipoint service must be expanded. Duplex communication is not possible, so the operation mode being introduced for acknowledged data transfer is the

asynchronous unbalanced mode (AUM). Non-acknowledged data transfer is provided through the exchange of unnumbered information frames.

New service primitives Because of priority, it may be necessary to suspend a logical connection in the short term and resume at a later time. This is supported by two new service primitives (LAPG_suspend, LAPG_resume) and requires an appropriate expansion of the service primitive for connection establishment SABM. Communication with the RLC layer likewise requires one or two new primitives, which are specified as LAPG_data and LAPG_ptm_data.

The medium access control (MAC) layer Along with the management of the mobility of a user and the associated routing functions, the specification of the MAC protocol is central to GPRS. The GPRS MAC sublayer defines procedures for allowing a larger number of users access to the physical radio channel. It contains the key functions of collision resolution, multiplexing and a reservation strategy with consideration of agreed quality of service. A peculiarity of the GPRS MAC protocol is that several physical (TCH) channels can be allocated to a mobile station at the same time (*multi-slot assignment, MSA*) and consequently the data transfer rate can be dynamically changed to $n \cdot 9.6$ kbit/s ($n < 8$) during a connection.

Between the MAC and LLC sublayers is the *radio link control* (RLC) sublayer (see Figure 3.81), which controls the connection between mobile station and BSS at the radio interface, provides bit map-based selective request mechanisms (SREJ) for packets transmitted with errors, and undertakes the necessary segmentation of the LLC protocol data units.

The protocols for both sublayers are briefly explained below.

The physical layer The physical layer at the radio interface is subdivided into the *physical link layer* (PLL) and the *physical radio-frequency layer* (RF). The specifications for these sublayers appear in the Series 05 GSM Recommendations. In view of the special features of the packet data service (e.g., regarding cell change procedures), no further adaptations to specifications are being planned. Whereas the RF sublayer deals with the modulation and demodulation of radio waves, the PLL layer provides services for bit-by-bit data transmission in the form of bursts over the radio interface. It is also responsible for forward error correction and interleaving, and contains functions for time-oriented synchronization (*timing advance*) of the respective mobile station and base station and evaluation of radio signal quality. Cell selection and power control are also embedded in this layer.

3.11.3.4 The RLC/MAC Protocol

This section presents the status of the standardization of the data link layer and access protocol to the GPRS radio interface as of 1997.

Access concept The RLC/MAC protocol controls access by packet data users to the radio medium through algorithms for collision resolution, multiplexing and channel reservation, taking into account the quality of service requested. A peculiarity of the protocol is its *multi-slot* mode, in which a mobile station is allocated several logical channels at the same time. Channel access is based on Slotted-ALOHA access procedures.

As soon as an LLC frame is availabile for transmission in the RLC/MAC protocol, it will be segmented depending on a coding scheme and distributed over the RLC/MAC protocol data units (PDU). An RLC/MAC-PDU in turn will be interleaved into four normal bursts, which in the same number of time slots will be transmitted in consecutive TDMA frames.

Based on a frame sequencing number, the ARQ/LLC-protocol controls the selective repeat request of blocks received with errors. An LLC frame will not be routed to the LLC layer until it has been reassembled correctly.

The base station organizes when and on which channel a mobile station is allowed to send data. On one hand this is determined by the channel reservation (time slot number), and on the other hand also by the value of a special bit combination, the `uplink state flag` (`USF`). On the basis of a temporary `USF` value, an MS detects whether a channel is available to it for transmission so that capacity can be provided dynamically to several mobile stations on one channel. In [35] a short description of the functions and the traffic performance of the GPRS RLC/MAC protocol has been published using the MSDRA (*master-slave dynamic rate access*) protocol, a precursor version of the current design.

3.11.3.5 The Logical Channel Concept

The mapping of *packet data channels* (PDCH) to the physical channels follows the current GSM principle. A PDCH is therefore determined by a physical time slot and the TDMA number. The uplink and the downlink represent separate logical resources; i.e., on the same PDCH data can be sent to an MS (downlink) and data can be received by another MS (uplink) at the same time. In line with the requirement for flexibility in adapting to the traffic volume of a cell, the allocation of a PDCH is dynamically controlled.

To check-in into the GPRS, a PDCH that is indicated on the PBCCH is used as a bidirectional signalling channel. All the necessary signalling takes place on this so-called *master packet data channel* (MPDCH). The MPDCH is subdivided logically into the following channels (see Table 3.37):

- *Packet broadcast control channel* (PBCCH), which indicates control information specific to the GPRS.
- *Packet random-access channel* (PRACH) for random access.
- *Packet paging channel* (PPCH), which indicates which data is to be sent on the downlink.

Table 3.37: The logical packet data channels in the master–slave concept

Group	Channel	Channel identification	Direction
MPDCH	PBCCH	*Packet Broadcast Control Channel*	(MS ⇐ BS)
	PRACH	*Packet Random Access Control Channel*	(MS ⇒ BS)
	PPCH	*Packet Paging Channel*	(MS ⇐ BS)
	AGCH	*Packet Access Grant Channel*	(MS ⇐ BS)
SPDCH	PACCH	*Packet-Associated Control Channel*	(MS ⇔ BS)
PTCH	PTCH	*Packet Traffic Channel*	(MS ⇔ BS)

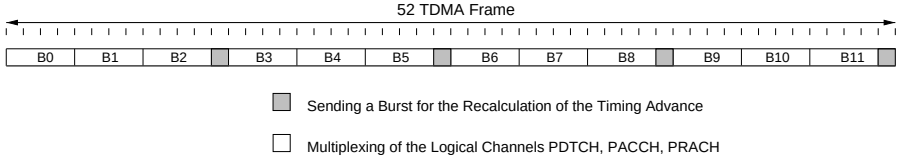


Figure 3.82: Multiframe cycle on the downlink

- *Packet access grant channel* (PAGCH), on which channel reservations are indicated.

Data transfer and connection-related signalling run on the so-called *slave packet data channel* (SPDCH). This includes:

- *Packet traffic channels* (PTCH), which are allocated temporarily to individual MSs for data transfer.
- *Packet-associated control channels* (PACCH), on which connection-related signalling data is transmitted.

This concept makes it possible to report information on the BCCH only about the MPDCH and to save channel capacity on the BCCH. In addition, this procedure facilitates the dynamic changing back and forth between PDCHs and TCHs.

3.11.3.6 Channel Structure

A multiframe structure, as already familiar from Figure 3.14, is required for mapping logical packet data channels to physical GSM channels. At the same time it must be considered that, along with the mobile subscriber data and GPRS-specific signalling information, a mobile station operating in packet data mode must be able to receive the signalling channels of GSM (e.g., BCCH or SCH) at regular intervals.

A 52-frame multiframe, produced as the result of two 26-frame multiframes (see Figure 3.82), is defined for the RLC/MAC protocol. The specified *mul-*

Table 3.38: Coding scheme

Coding scheme	Coding rate	USF	I-frames or USF/BCS	BCS	Tail Bit	Cod. Bit	Punct. Bit	Data rate [kbps]
1	1/2	3	181	40	4	456	0	9.05
2	$\approx 2/3$	6	266	16	4	584	128	13.3
3	$\approx 3/4$	6	314	16	4	680	224	15.7
4	1	12	428	16	0	456	0	21.4

tiframe cycle consists of four 52-frame multiframes; however, it has not yet been specified exactly how the logical channels are to be mapped.

Procedures for adapting transmitting power and establishing the BSIC are carried out in the shaded IDLE frames. In conformance with an interleaving depth of four used to scramble an RLC/MAC-PDU, the same four time slots of consecutive TDMA frames are always combined into a block B_i $(i = 0, 1, \ldots, 11)$.

3.11.3.7 Error Protection

The RLC/MAC protocol uses the punctured convolutional code of the GSM 1/2-convolutional coding. The decoding is carried out with the Viterbi "soft decision" decoder familiar from GSM.

The scheme used for the GSM-SDCCH involving a convolutional code with a code ratio 1/2 and a 40-bit Fire code is used as standard for transmitting signalling information. For user information further code ratios are used, the redundancy of which adapts dynamically to the current signal quality. Table 3.38 presents the four different options.

The decoding of the uplink status flag (USF) is simplified whereby for coding schemes 2–4 a 12-bit long code word, which is not punctured, is generated for the USF. With schemes 2 and 3 the USF is precoded with 6 bits before the frame is convolutionally coded, with the first 12 bits not being punctured. When coding scheme 1 is used, the entire frame is coded and the USF must be decoded as part of the information data. Figure 3.83 shows an example of the coding of an information frame for coding scheme 3.

Both *stealing bits* of a GSM normal burst are used to indicate the coding scheme to be followed. Since four consecutive bursts accompany an RLC/MAC frame, the scheme can be protected against transmission error by an 8-bit block code of the Hamming distance 5.

3.12 Interworking Function (IWF)

Interworking functions allow GSM-supported services to be connected to those of the fixed networks and vice versa (see Figure 3.60).

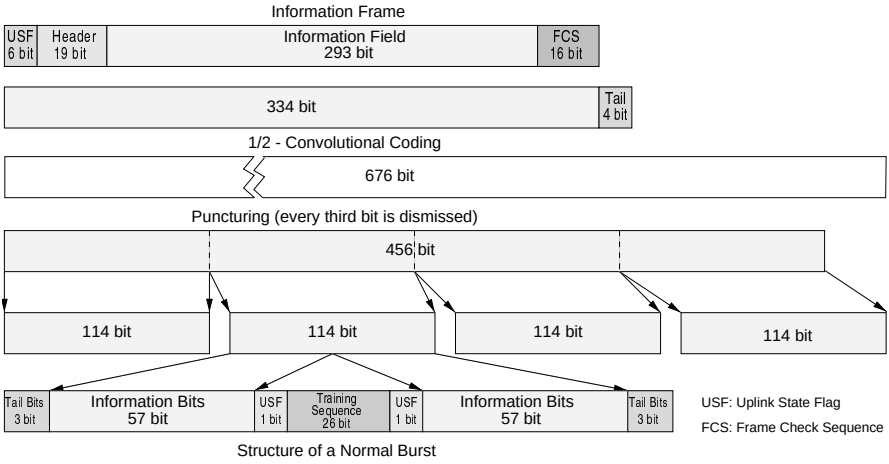


Figure 3.83: Coding of an information frame according to scheme 3

3.12.1 Gateway to the Public Switched Telephone Network

The *public-switched telephone networks* (PSTN) in the CEPT countries have developed in a variety of different ways over a period of many years. As a result, there is a strong reliance on the respective PSTN for the interworking functions needed. The GSM recommendations for the PLMN side of the IWFs are very detailed, whereas only general requirements have been formulated for the PSTN side. The problem is that a switching centre is unable to specify the service desired for calls coming from PSTNs. Therefore a technique is required to differentiate between the types of connection to enable the mobile radio network to select the correct IWFs. Two methods have been selected (GSM Rec. 09.07):

1. Allocation of separate numbers (MSISDN) for each service (optional) to each mobile user, with each number representing a particular bearer service or teleservice, e.g., voice, facsimile group 3, transparent synchronous data transfer at 9.6 kbit/s, and each service requiring special IWFs. The associated information is stored in the HLR under the `MSISDN` together with the MSRN, is established when the mobile user checks-in, and can only be changed through administrative channels.

2. The called mobile user conveys the request for a specific service in his *call confirmation message*. This method is obligatory for all networks and mobile stations because it is also applied to ISDN-terminated calls. The mobile user here is assigned only one `MSISDN` for all services. Based on GSM 03.04, all calls are routed by the querying switching centre. The mobile user is then sent a *call setup message* but without any bearer service information included. The information relating to the service

desired is sent by the mobile user in his call acknowledgement. Only then does the MSC/IWF select the appropriate resources and establish a connection.

3.12.2 Gateway to ISDN

Three bearer services are available for the interworking between Integrated Services Digital Network (ISDN) and GSM-PLMN:

Circuit-switched transmission (3.1 kHz audio) The same solutions being applied to PSTN gateways are being used for this service. The main difference is in the available signalling capabilities of the fixed network, which can allow the selection of IWFs and indication of connection type and teleservice. However, this possibility depends on how this service is currently implemented in the fixed network.

Circuit-switched unrestricted digital bearer service The IWF permits access to circuit-switched data transmission from and to ISDN in accordance with the ISDN standard rate adaptation based on ITU-T X.30 and V.110. This means that a connection can be established with any ISDN subscriber with an asynchronous or synchronous terminal, with transmission rates up to 9.6 kbit/s. Conventional terminals with a V.24 or an X.21 interface or ISDN terminals with an S-interface can be used (see Figure 3.3). Only a minimum number of IWFs are needed for transparent services, because 64 kbit/s are already offered by the MSC. Specifications are still needed for terminal-to-terminal synchronization and status bit filtering. Non-transparent services require additional IWFs. The RLP must be properly terminated and the standard ISDN block structure restored.

Packet-switched transmission Because in ISDN access to the packet services takes place over the S-interface either over a B-channel (64 kbit/s) or a D-channel (16 kbit/s), but GSM only has B_m- or L_m-channels, the data rates do not fit. Rate adaptation will also be carried out in this area.

3.12.3 Gateway to the Public Switched Packet Data Network

There are two different bearer service for the interworking between the GSM network and the *Public Switched Packet Data Network* (PSPDN):

1. Access to the PAD service, duplex, asynchronous with transmission rates of 300–9600 bit/s, with two possibilities for implementation:
   (a) *Basic PAD access* is carried out over an asynchronous bearer service, e.g., a PSTN, to an existing PAD unit in the wired network.

Existing IWFs to the PSTN can be used for data services, and additional functions such as billing are not required. The disadvantage is that users are committed to the options and transmission rates of the wired network (in some countries only 1.2 kbit/s). The mobile user must be registered by the PSPDN of each country visited.

(b) With *dedicated PAD access* the GSM-PLMN has its own PAD unit with the service quality parameters specified by the network operator, e.g., a maximum data rate of 9.6 kbit/s. No registration with a PSPDN is required for the mobile subscriber. The disadvantage is cost because of the need to implement additional functions, particularly for billing.

2. Access to a PSPDN with the rates 2.4/4.8/9.6 kbit/s, duplex, synchronous. There are two possibilities:

   (a) Access over the X.32 interface and the PSTN as the transit network. There is a problem because of incompatible X.32 implementations in Europe. Access to a PSPDN is through dedicated *access units*. A PSPDN regards them as packet-switching and communicates with them over an X.75 protocol. This procedure requires close cooperation with the PSPDN operator.

   (b) Access via ISDN using the X.31 protocol. The IWFs GSM/ISDN can then be used.

3.12.4 Gateway to the Public Switched Data Network

There is a gateway between the *public switched data network* (PSDN) and GSM; most of the procedures used are those already available for PSTN and ISDN.

3.12.5 Interworking Functions for Teleservices

GSM-SMG intentions to support the terminals existing in PSTNs have caused considerable problems. The two-wired telephone interface (a/b) of this equipment must be adapted to the digital mobile radio interface; support of the S- and R-interfaces is also required (see Section 3.10.3).

- Access to videotex: PSTN gateways or PAD access can be used minimizing the need for IWFs to videotex. Roaming mobile users face problems because of the different videotex profiles in different countries (see GSM 03.43).

- Access to facsimile, group 3: This has turned out to be a particularly complex service gateway. There are two different scenarios:

  (a) Support of existing terminals using a two-wire or four-wire analogue interface. In this case the equipment termination must contain

a modem function for converting speech band data signals into data streams, thereby also increasing the complexity of the mobile network termination.

(b) Use of terminals with a data interface, e.g., V.24. This would offer the advantage that existing PSTN gateways for data transmission could be used. The disadvantage is that these terminals are not yet available in the market. The PC interface PCMCIA (*Personal Computer Multiplexing Communications Interface Adapter*) is currently the preferred one in the market.

3.13 Security Aspects

Open access to the network over the radio interface leads to the danger that communication between mobile subscribers could be open to eavesdropping by third parties or that network resources could be subject to unauthorized use at the cost of registered subscribers. As a precautionary measure, access to and use of the mobile communication network is protected by security procedures that are essentially based on proof of identity of the mobile subscriber through authentication, enciphering of message transmission (including the signalling data) as well as anonymization of the identity of the mobile subscriber through a time change of the user identification (TMSI). GSM digital message transmission provides for the use of cryptographic procedures.

With GSM the storage of secret communication data is carried out with the mobile user in a non-manipulable module (SIM), which can be in the form of either a *plug-in* module or an IC card and is also protected against unauthorized use by a *personal identity number* (PIN). Identification of a mobile subscriber to GSM is only possible with a valid SIM. The module gives the user the advantage of being able to have unrestricted access to any agreed services using any other mobile radio unit and to be charged for them. The data stored in the SIM includes the subscriber ciphering key K_i as well as information that is necessary for using the system flexibly and efficiently (see Section 3.2.1).

3.13.1 Authentication

The identity of a mobile subscriber is proved to the network through the process of authentication.

A subscriber joining a mobile radio network is allocated an international unique subscriber identity (IMSI), a secret authentication key K_i and a secret algorithm A3, which are stored in the SIM. If the subscriber is to identify himself, he is provided with a random number `RAND` from the network, which is used by the mobile station to calculate the authenticator $\texttt{SRES} = K_i(\texttt{RAND})$ with K_i and A3 (see Figure 3.84). It sends the result to the VLR, where it is checked against the internal value `SRES` for agreement. If the result is

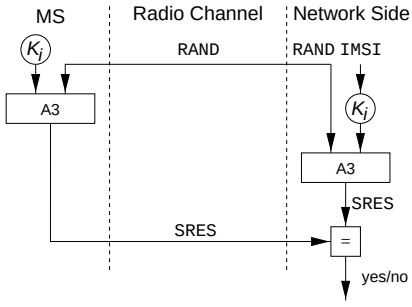


Figure 3.84: Authentication of a mobile subscriber

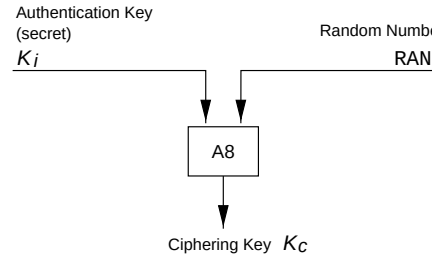


Figure 3.85: Generation of a ciphering key

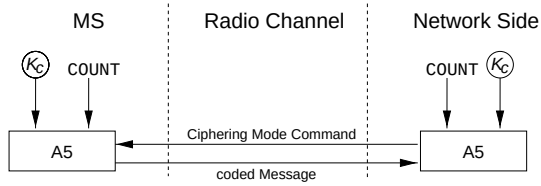


Figure 3.86: Ciphering mode

positive, the mobile subscriber is authorized to proceed; otherwise all existing transactions are immediately terminated.

3.13.2 Confidentiality of User and Signalling Data

All messages with subscriber-related information are transmitted in protected mode. A ciphering key K_c is generated from a random number `RAND` with the authentication key K_i and A8 algorithm (see Figure 3.85).

K_c is not transmitted over the radio path, but is stored in the mobile station and recalculated with each authentication procedure. Synchronization between the mobile station and the network is ensured whereby the ciphering key is allocated a key number (`Count`), which is also stored in the mobile station and supplied to the network each time a new message is transmitted. If the network recognizes that data requiring protection is being transmitted, it initiates a ciphering procedure whereby it sends a Ciphering_Mode_Command message to the mobile station. The mobile station in turn encodes or decodes on the basis of a stream cipher procedure using the A5 algorithm and the key K_c (see Figure 3.86).

3.13.3 Confidentiality of Subscriber Identity

The temporary mobile subscriber identity (`TMSI`) is changed periodically to guarantee the confidentiality of the information transmitted by a mobile user.

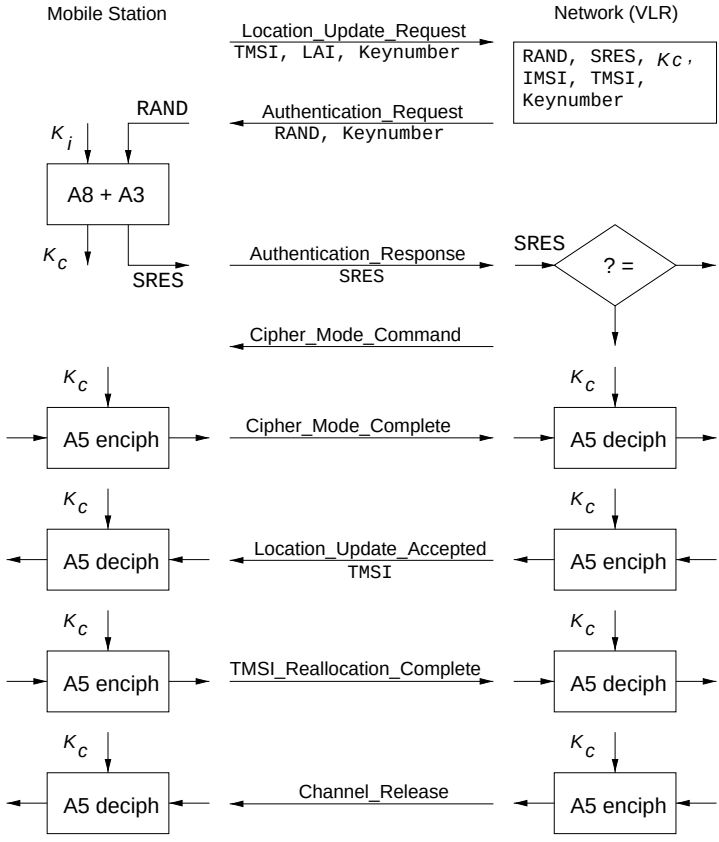


Figure 3.87: Location area update from the standpoint of security

This prevents a mobile radio connection from being allocated to a certain mobile subscriber as a result of eavesdropping. The TMSI is always assigned temporarily by the current VLR and is transmitted coded to the mobile station. A `TMSI` is changed no later than when there is a change in the location area controlled by a VLR. However, it can also be changed even if there is no change in VLR. Figure 3.87 shows a diagram of the procedure location area update from the standpoint of security.

3.13.4 The Transport of Security-Related Information between MSC, HLR and VLR

The authentication centre (AuC) is set up separately with special protection or integrated into the HLR, and guarantees the security of the network. It carries out the following tasks:

- Generation of key K_i as well as its allocation to `IMSI`.

- Generation of records RAND/SRES/K_c per IMSI for transfer to the HLR.

For updating the location area a VLR requires security-related information, which it obtains as follows:

- If a mobile station identifies itself using an IMSI, the VLR sends a request to the HLR for five records RAND/SRES/K_c, which are allocated to this IMSI.
- If a mobile station identifies itself using a TMSI and momentary location area identification LAI, the new VLR requests the IMSI as well as the records RAND/SRES/K_c from the old VLR that have not been used up.

If a mobile station roams in the area of a VLR for a longer period of time, after several authentications the VLR requires new RAND/SRES/K_c records. It receives them by sending a request to the HLR of the respective mobile user. Authentication itself takes place in the VLR whereby it sends the RAND to the MSC. The MSC establishes the authenticator SRES and transmits it to the VLR, which compares the value of the authenticator with that received from the mobile station. If the authentication is successful, the VLR allocates a TMSI to the IMSI. The ciphering key and the TMSI are sent to the MSC.

3.14 Closing Remarks

GSM 900 (the addition of the 900 underlines the frequency of operation) incorporates two important technical advances: it has implemented digital transmission technology and introduced a large number of new services and supplementary services, and it has replaced existing national standards with international ones. Although the new services offered by GSM during the initial phase were mainly attractive to business users, by 1998 there were already more than 45 million mobile subscribers in Europe, and by 1 August 1998 there were 100 million GSM subscribers worldwide, with the numbers rising sharply.

The role GSM will play in future market penetration initially depends on the current stage of development of mobile communication in different countries and regions.

Markets that were opened up without predecessor networks have shown that GSM subscribers grew relatively quickly in number.

In existing markets GSM is able to eliminate national capacity bottlenecks, making it sufficiently attractive to entice subscribers from analogue networks to the GSM system. In developed markets GSM particularly offers the possibility of supporting personal communication, thereby creating new communications opportunities. Table 3.39 shows the extent of current GSM coverage (without claiming completeness).

In the USA many regional mobile radio network operators have decided to operate GSM networks—albeit in the 1900 MHz frequencies, which means that

special radio front ends are required for the licenced bands. These systems are named GSM 1900 systems. Operators of digital mobile radio systems with a different radio interface use the services and protocols of GSM in network and switching subsystems (see Figure 3.1).

Since the introduction of Dual-Band GSM phones it makes sense to list GSM 900 and GSM 1800/1900 operators together. 1900 MHz is used in the US, Canada and Japan, the rest of the world uses 900 MHz (usually first two operators) and 1800 MHz (usually third and fourth operator).

Table 3.39: GSM coverage at 900, 1800 and 1900 MHz (as of November 1998)

Country	Operator Name	Network Code
Albania	AMC	276 01
Andorra	STA-Mobiland	213 03
Argentina		
Armenia	Armentel	283 01
Australia	Optus	505 02
	Telecom/Telstra	505 01
	Vodafone	505 03
Austria	Mobilkom Austria	232 01
	max.mobil.	232 03
	Connect Austria	232 05
Azerbaijan	Azercell	400 01
	JV Bakcell	
Bahrain	Batelco	426 01
Bangladesh	Grameen Phone Ltd	470 01
	TM International	
	Sheba Telecom	
Belgium	Proximus	206 01
	Mobistar	206 10
	KPN Orange	
Bosnia	Cronet	218 01
	PTT Bosnia	218 19
Botswana	Mascom Wireless	
Brunei	DSTCom	528 11
	Jabatan Telekom	528 01
Bulgaria	Citron	284 01
Burkina Faso	OnaTel	
Cambodia	CamGSM	456 01
	Cambodia Samart	
	Cambodia Shinawatra	
Cameroon	PTT Cameroon Cellnet	624 01
Cape Verde	Cabo Verde Telecom	
Canada	Microcell	302 37
Chile	Entel Telefonia	
China	Guangdong MCC	460 00
	Beijing Wireless	
	China Unicom	460 01
	Zhuhai Comms	
	DGT MPT	
	Jiaxing PTT	
	Tjianjin Toll	
	Liaoning PTTA	460 02
Congo	African Telecoms	

Continued

Table 3.39: GSM coverage (as of 11/98) (continued)

Country	Operator Name	Network Code
	Congolaise Wireless	
Croatia	HR Cronet	219 01
Cyprus	CYTA	280 01
Czech Rep.	Eurotel Praha	230 02
	Radio Mobil	230 01
Denmark	Sonofon	238 02
	Tele Danmark Mobil	238 01
	Mobilix	238 30
	Telia	238 20
Egypt	Arento	602 01
Estonia	EMT	248 01
	Radiolinja Eesti	248 02
	Q GSM	248 03
Ethiopia	ETA	636 01
Faroe Isl.	Faroese Telecom	
Fiji	Vodafone	542 01
Finland	Radiolinja	244 05
	Sonera	244 91
	Alands Mobiltelefon	244 05
	Telia	244 03
	Finnet	244 09
	Lännen Puhelin	
	Helsingin Puhelin	
France	Itineris	208 01
	SFR	208 10
	Bouygues Telekom	208 20
Fr. Polynesia	Tikiphone	547 20
Fr. W. Indies	Ameries	340 01
Georgia	Superphone	
	Geocell	282 01
	Magticom	282 02
Germany	D1, DeTeMobil	262 01
	D2, Mannesmann	262 02
	E-Plus Mobilfunk	262 03
	Viag Interkom	262 07
Ghana	Franci Walker Ltd	
	ScanCom	620 01
Gibraltar	GibTel	266 01
Great Britain	Cellnet	234 10
	Vodafone	234 15
	Jersey Telecom	234 50
	Guernsey Telecom	234 55
	Manx Telecom	234 58
	One2One	234 30
	Orange	234 33
Greece	Panafon	202 05
	STET	202 10
	Cosmote	202 01
Greenland	Tele Greenland	
Guinea	Int'l Wireless	611 ??
	Space Tel	611 ??
	Sotelgui	611 02

Continued

Table 3.39: GSM coverage (as of 11/98) (continued)

Country	Operator Name	Network Code
Hong Kong	HK Hutchison	454 04
	SmarTone	454 06
	Telecom CSL	454 00
	P Plus Comm	454 22
	New World PCS	454 10
	Sunday	454 16
	Pacific Link	454 18
	Peoples Telephone	454 12
Hungary	Pannon GSM	216 01
	Westel 900	216 30
Iceland	Post & Simi	274 01
	Icelandic Mobile Phone	274 02
India	Airtel	404 10
	Essar	404 11
	Maxtouch	404 20
	BPL Mobile	404 21
	Command	404 30
	Mobilenet	404 31
	Skycell	404 40
	RPG MAA	404 41
	Modi Telstra	404 14
	Sterling Cellular	404 11
	Mobile Telecom	
	Airtouch	
	BPL USWest	404 27
	Koshika	
	Bharti Telenet	
	Birla Comm	
	Cellular Comms	
	TATA	404 07
	Escotel	404 12
	JT Mobiles	
	Evergrowth Telecom	
	Aircell Digilink	404 15
	Hexacom India	
	Reliance Telecom	
	Fascel Limited	
Indonesia	TELKOMSEL	510 10
	PT Satelit Palapa	510 01
	Excelcom	510 11
	PT Indosat	
Iraq	Iraq Telecom	418 ??
Iran	T.C.I.	432 11
	Celcom	
	Kish Free Zone	
Ireland	Eircell	272 01
	Digifone	272 02
	Meteor	272 03
Israel	Partner Communications	
Italy	Omnitel	222 10
	Telecom Italia Mobile	222 01
	Wind	222 88

Continued

Table 3.39: GSM coverage (as of 11/98) (continued)

Country	Operator Name	Network Code
Ivory Coast	Ivoiris	612 03
	Comstar	612 01
	Telecel	612 05
Japan		
Jordan	JMTS	416 01
Kenya	Kenya Telecom	
Kuwait	MTCNet	419 02
Kyrgyz Rep.	Bitel Ltd	437 01
La Reunion	SRR	647 10
Laos	Lao Shinawatra	457 01
Latvia	LMT	247 01
	BALTCOM GSM	247 02
Lebanon	Libancell	415 03
	Cellis	415 01
Lesotho	Vodacom	651 01
Liechtenstein	Natel-D	228 01
Lithuania	Omnitel	246 01
	Bite GSM	246 02
Luxembourg	P&T LUXGSM	270 01
	Millicom Lux' S.A.	270 77
Lybia	Orbit	
	El Madar	
Macao	CTM	455 01
Macedonia	PTT Makedonija	294 01
Madagascar	Sacel	646 03
	Madacom	646 01
	SMM	
Malawi	TNL	650 01
Malaysia	Celcom	502 19
	Maxis	502 12
	My BSB	502 02
	MRTEL	502 13
	Adam	502 17
	Mutiara Telecom	502 16
Malta	Telecell	278 01
Marocco	O.N.P.T.	604 01
Mauritius	Cellplus	617 01
Monaco	Itineris	208 01
	SFR	208 10
	Office des Telephones	
Mongolia	MobiCom	
Montenegro	Pro Monte	220 02
Mozambique	Telecom de Mocambique	634 01
	T.D.M. GSM1800	
Namibia	MTC	649 01
Netherlands	PTT Netherlands	204 08
	Libertel	204 04
	Telfort Holding NV	204 12
New Caledonia	Mobilis	546 01
New Zealand	Bell South	530 01
Nigeria	EMIS	
Norway	NetCom	242 02

Continued

Table 3.39: GSM coverage (as of 11/98) (continued)

Country	Operator Name	Network Code
	TeleNor Mobil	242 01
Oman	General Telecoms	422 02
Palestine	Palestine Telecoms	
Pakistan	Mobilink	410 01
Papua	Pacific	310 01
Philippines	Globe Telecom	515 02
	Islacom	515 01
Poland	Plus GSM	260 01
	ERA GSM	260 02
	IDEA Centertel	260 03
Portugal	Telecel	268 01
	TMN	268 06
	Main Road Telecoms	
	Optimus	268 03
Qatar	Q-Net	427 01
Reunion		
Romania	MobiFon	226 01
	MobilRom	226 10
Russia	Mobile Tele... Moscow	250 01
	United Telecom Moscow	
	NW GSM, St. Petersburg	250 02
	Dontelekom	250 ??
	KB Impuls	250 99
	JSC Siberian Cellular	250 ??
	BM Telecom	250 07
San Marino	Omnitel	222 10
	Telecom Italia Mobile	222 01
	Wind	222 88
Saudi Arabia	Al Jawal	420 01
	EAE	420 07
Senegal	Sonatel	608 01
Seychelles	SEZ SEYCEL	633 01
Serbia	Serbian PTT	220 03
Singapore	Singapore Telecom	525 01
	MobileOne	525 03
Slovak Rep.	Eurotel	231 02
	Globtel	231 01
Slovenia	Mobitel	293 41
South Africa	MTN	655 10
	Vodacom	655 01
Sri Lanka	MTN Networks Pvt Ltd	413 02
Spain	Airtel	214 01
	Telfonica Spain	214 07
	Retevision	
Sudan	Mobitel	634 01
Swaziland		
Sweden	Comviq	240 07
	Europolitan	240 08
	Telia	240 01
Switzerland	Swisscom 900	228 01
	Swisscom 1800	228 01
Syria	SYR MOBILE	417 09

Continued

Table 3.39: GSM coverage (as of 11/98) (continued)

Country	Operator Name	Network Code
Tahiti		
Taiwan	LDTA	466 92
	Mobitai	466 93
	TransAsia	466 99
	TWN	466 97
	Tuntex	466 06
	KG Telecom	466 88
	FarEasTone	466 01
Tanzania	Tritel	640 01
Thailand	TH AIS GSM	520 01
	Total Access Comms	520 18
Tunisia	Tunisian PTT	605 02
Turkey	Telsim	286 02
	Turkcell	286 01
UAE	UAE ETISALAT-G1	424 01
	UAE ETISALAT-G2	424 02
Uganda	Celtel Cellular	641 01
Ukraine	Mobile comms	255 01
	Golden Telecom	255 05
	Radio Systems	
	Kyivstar JSC	
USA	Bell South	310 15
	Sprint Spectrum	310 02
	Voice Stream	310 26
	Aerial Comms.	310 31
	Omnipoint	310 16
	Powertel	310 27
	Wireless 2000	310 11
Uzbekistan	Daewoo GSM	434 04
	Coscom	434 05
	Buztel	434 01
Vatican City	Omnitel	222 10
	Telecom Italia Mobile	222 01
	Wind	222 88
Vietnam	MTSC	452 01
	DGPT	452 02
Yugoslavia	Mobile Telekom	220 01
	Pro Monte	220 02
Zaire	African Telecom Net	
Zimbabwe	NET*ONE	648 01
	Telecel Zimbabwe	

3.15 ETSI/DCS 1800 Digital Mobile Radio Network

The ETSI/DCS 1800 (*Digital Cellular System at 1800 MHz*) mobile radio network is regarded as an extension as well as a competitor to the GSM 900 network. The DCS 1800 standard developed by ETSI is based on the GSM recommendations but applied to the higher frequency range at 1800 MHz. This

is the reason why nowadays DCS 1800 is termed GSM 1800. The frequency ranges assigned in Europe are 1710–1785 MHz (uplink) and 1805–1880 MHz (downlink) (see Appendix C). A total of 374 carrier frequencies are therefore available to the system.

The entire technical infrastructure of the DCS 1800 system is directed at the mass market for personal mobile communication—a so-called *Personal Communication System* (PCS) planned to be comparable to the system licensed for the USA/FCC (*Federal Communication Commission*, the frequency regulator in the USA) frequency band at 1900 MHz, with:

- low communication costs similar to those of the fixed network
- very high network capacity
- lightweight, compact radio telephone
- radio coverage also in buildings

The DCS 1800 system meets the following requirements of PCS systems:

- high traffic density of 500 Erl./km^2
- lower power consumption of 250 mW up to 2 W
- use of GSM half-rate codec
- cost-effective implementation of the network

The DCS 1800 system follows the GSM recommendations, except for the frequency-specific specifications, which is evident from the common architectural components (see Figure 3.88) with network elements which for the most part have already been shown in Figure 3.1: *Base Transceiver Station* (BTS), *Base Station Controller* (BSC), *Mobile Services Switching Centre* (MSC), *Operation and Maintenance Centre* (OMC), *Service Creation and Accounting Centre* (SCAC), *Home Location Register* (HLR), *Visitor Location Register* (VLR) und *Equipment Identification Register* (EIR).

The deviations of DCS 1800 from the GSM 900 specifications were published in 11 Delta recommendations (see Appendix E). The changes affect the radio interface definition, which differs owing to the low transmitter power required (250 mW up to 2 W in DCS 1800 compared with 5 up to 10 W in GSM). Because of the low transmitter power and the transmission attenuation, which on average is 10 dB higher at 1800 MHz than at 900 MHz, the cell radii in DCS 1800 are smaller than in GSM 900. With DCS 1800, cell radii of a maximum of 1 km are in use in urban areas, whereas in open country radio cells of up to 8 km (macrocells) are supported.

There will be two important different cell sizes in urban areas: microcells with a radius of over 150 m and picocells for the support of mobile radio communication inside buildings. Smaller cells have the advantage of higher traffic capacity, thus providing better support of personal communication.

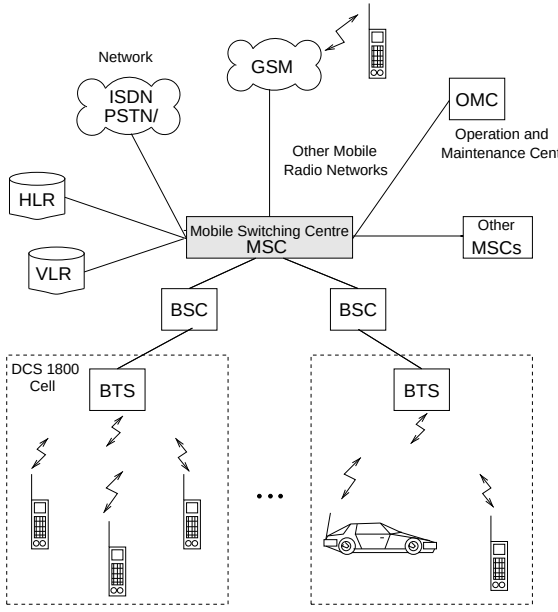


Figure 3.88: The architecture of the DCS 1800 system

However, DCS 1800 must plan for considerably more cells than are required in GSM, resulting in a higher investment in the development of the network infrastructure.

So that there is no great deviation between the GSM 900 and the DCS 1800 specifications, it was agreed that during the second phase of DCS 1800 standardization ETSI would combine the different documentations into a joint range of specifications. Other aspects to be examined in the second phase cover local routing and roaming between GSM 900 and DCS 1800, extension of the DCS 1800 service to include coin-operated telephones, and interworking with Inmarsat and satellite services [29].

Great Britain was considered to be the trailblazer in the introduction of the DCS 1800 system, which is called a PCN there and was put into operation in early 1993.

3.16 GSM Abbreviations and Acronyms

A3	Authentication Algorithm
A5,A8	Encryption Algorithm
ABM	Asynchronous Balanced Mode
ADM	Asynchronous Disconnected Mode
ACF	Autocorrelation Function

ARQ	Automatic Repeat Request
AuC	Authentication Centre
AGCH	Access Grant Channel
ASCI	Advanced GSM Speech Call Items
ASE	Application Service Element
BCCH	Broadcast Control Channel
BCF	Base Station Control Function
BSC	Base Station Controller
BSIC	Base Station Identity Code
BSS	Base Station System
BTS	Base Station Transceiver System
CC	Country Code
CCCH	Common Control Channel
CCITT	Comité Consultatif International de Télégraphes et Téléphones
CDM	Code-Division Multiplex
CE	Connection Endpoint
CEPT	Conférence Européene des Administrations des Postes et des Télécommunications
CI	Cell Identity
CRC	Cyclic Redundancy Check
DCCH	Dedicated Control Channel
DLCI	Data Link Connection Identifier
DS	Direct Sequencing
DSU	Data Service Unit
DTAP	Direct Transfer Application Part
DTMF	Dual-Tone Multiple Frequency
DTX	Discontinuous Transmission
EIR	Equipment Identity Register
eMLPP	enhanced Mult-Level Precedence and Pre-Emption
ETSI	European Telecommunications Standards Institute
FACCH	Fast-Associated Control Channel
FCCH	Frequency Correction Channel
FDM	Frequency-Division Multiplexing
FEC	Forward Error Correction
FH	Frequency Hopping
FPLMTS	Future Public Land Mobile Telecommunications System
GGSN	Gateway GPRS Support Node
GMSC	Gateway MSC
GMSK	Gaussian Minimum Shift Keying
GPRS	General Packet Radio Service
GSM	Groupe Spéciale Mobile
HDLC	High-Level Data Link Control
HDRCH	High Data Rate Channel
HLR	Home Location Register

HSCSD	High Speed Circuit Switched Data
IMEI	International Mobile Equipment Identity
IMSI	International Mobile Subscriber Identity
ISDN	Integrated Services Digital Network
ISP	Intermediate Services Part
IWF	Interworking Function
K_c	Transmission Key
K_i	Authentication Key
LAI	Location Area Identity
LAPD	Link Access Procedure D-Channel
LPC	Linear Predictive Coding
MAP	Mobile Application Part
MCC	Mobile Country Code
MM	Mobility Management
MoU	Memorandum of Understanding
MS	Mobile Station
MSC	Mobile Services Switching Centre
MSRN	Mobile Station Roaming Number
MSISDN	Mobile Subscriber ISDN
MT	Mobile Termination
MTP	Message Transfer Part
NDC	National Destination Code
OACSU	Off-Air Call Set-Up
OMC	Operation and Maintenance Centre
OSI	Open Systems Interconnection
PAD	Packet Assembler Disassembler
PCH	Paging Channel
PCM	Pulse Code Modulation
PIN	Personal Identity Number
PLMN	Public Land Mobile Network
PN	Pseudo Noise
PSDN	Public Switched Data Network
PSPDN	Public Switched Packet Data Network
PSTN	Public Switched Telephone Network
PUK	PIN Unblocking Key
RACH	Random Access Channel
RAND	Random Number
RLP	Radio Link Protocol
RR	Radio Ressource Management
SACCH	Slow-Associated Control Channel
SAP	Service Access Point
SAPI	Service Access Point Identifier
SCCP	Signalling Connection Control Part
SCH	Synchronization Channel
SDCCH	Stand-Alone Dedicated Control Channel

SDM	Space-Division Multiplex
SGSN	Serving GPRS Support Node
SI	Subscriber Identity
SIM	Subscriber Identity Module
SMS	Short Messages Service
SNR	Signal-to-Noise Ratio
SRES	Authenticator
SS	Supplementary Service
SS 7	Common Channel Signalling System No. 7
TA	Timing Advance
TCAP	Transaction Capabilities Application Part
TCH	Traffic Channel
TDM	Time-Division Multiplexing
TKN	Transmission Key Number
TMSI	Temporary Mobile Subscriber Identity
TMN	Telecommunications Management Network
TRAU	Transcoder/Rate Adaptor Unit
TRX	Transceiver (Transmitter/Receiver)
UMTS	Universal Mobile Telecommunications System
USDC	US Digital Cellular
VAD	Voice Activity Detection
VCC	Visited Country Code
VEA	Very Early Assignment
VLR	Visitor Location Register
VMSC	Visited Mobile Switching Centre
VNDC	Visited National Destination Code
VSN	Visited Subscriber Number

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4
Other Public Mobile Radio Systems

4.1 Airline Telephone Network for Public Air–Ground Communication

In 1993 ETSI RES 5 submitted a standard for the *Terrestrial Flight Telephone System* (TFTS), specifying the radio interface and the interfaces to public telecommunications networks. At the same time the *European Airlines Electronic Committee* (EAEC) specified the airline equipment and interfaces to cabin facilities. Commercial operations began in 1994. In July 1994, after inviting international tenders, the Ministry of Post and Telecommunications granted a licence to DeTeMobil for the operation of TFTS. DeTeMobil was to supply radio coverage to all airspace up to an altitude of 4500 m. The service was available by 1996.

Thirteen network operators in Europe have signed an MoU for the introduction of TFTS and an agreement on a cooperation with the major European airlines in order to resolve related commercial, organizational, technical and operational issues [1, 2].

4.1.1 TFTS Cellular Network

TFTS is a cellular system that uses direct radio links to ground stations (GS) that are connected to the fixed network to provide public communication services for air passengers (see Figure 4.1).

There are three types of ground stations differentiated by area covered (cell) and related transmitter power:

- *En-route* (ER) GS for altitudes from 13 to 4.5 km, with cell radii up to 240 km
- *Intermediate* (I) GS for altitudes below 4.5 km, with cell radii up to 45 km
- *Airport* (AP) GS, with cell radii of 5 km

Handover between areas is part of the system. According to WARC'92, two 5 MHz wide bands have been specified for operation of the TFTS:

- 1670–1675 MHz for uplink (*ground-to-air*)

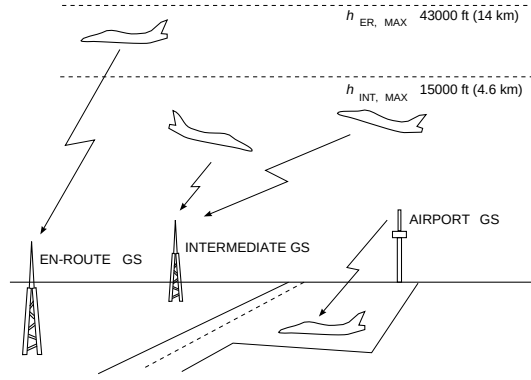


Figure 4.1: Coverage areas and ground stations

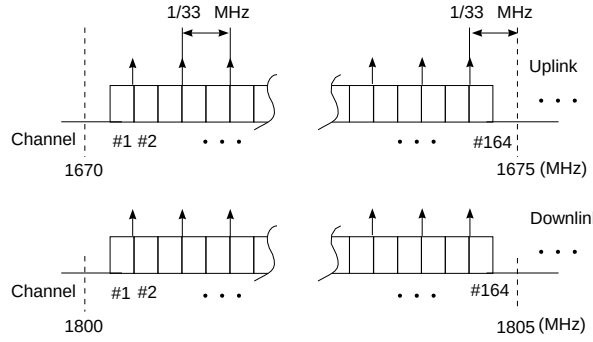


Figure 4.2: TFTS channel map

- 1800–1805 MHz for downlink

The system offers automatic dialled connections to PSTN/ISDN without any limitation on target subscribers, with the same quality of service as customary in PLMNs.

In addition to speech, data services such as facsimile, data transfer at 4.8 kbit/s and DTMF signalling are supported. Calls from the ground to an aircraft are only allowed to be made for operational purposes or for paging.

The user is billed directly by (credit) card for services used.

4.1.2 Frequency and Time-Multiplexing Channels

Each 5 MHz band is divided into 164 FDM channels (each 30.45 kHz wide); see Figure 4.2.

Each FDM channel transmits at 44.2 kbit/s gross. On the uplink this capacity is divided into 17 time channels based on the TDM method, and on

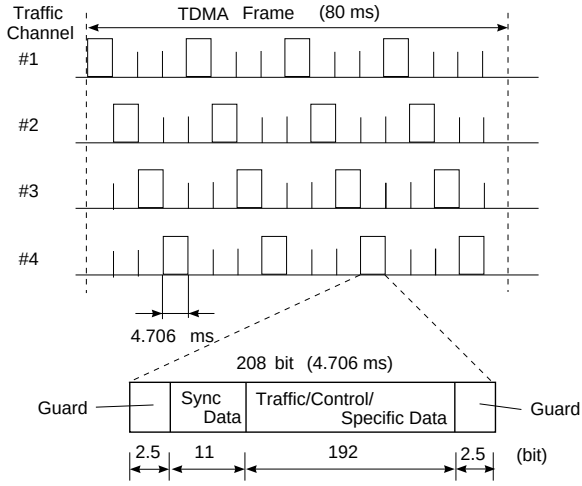


Figure 4.3: Frames and time slots

the downlink into 17 time channels based on the TDMA method. Each FDM channel carries four voice channels.

According to Figure 4.3, 17 time slots are combined into a frame of 80 ms duration, and 20 frames form a superframe of the duration of 1.6 s. Each time slot contains 208 bits and has a duration of 4.706 ms.

4.1.3 Voice and Data Transmission

Voice signals are digitally coded into blocks of 192 bits and transmitted at 9.6 kbit/s in time slots. A 9.6 kbit/s voice channel occupies 4 of the 17 time slots of an FDM channel; the 17th slot is used for network control.

As soon as voice codecs are available for a 4.8 kbit/s transmission rate, the number of voice channels will be doubled. Data services at 4.8 kbit/s require 2 time slots per frame, therefore each FDM channel carries 8 data channels.

4.1.4 Functional Characteristics

Each aircraft has transmitting and receiving facilities (*transceiver*), which can be tuned selectively to one of the different FDM channels. Four communications can be carried out on the same FDM channel at the same time. Ground stations can transmit to different aircrafts simultaneously (on different time channels) on each one of their FDM channels.

Signals are transmitted digitally with linear $\pi/4$-DQPSK (*Differential Quadrature Phase Shift Keying*) modulation, and require a simple non-coherent receiver.

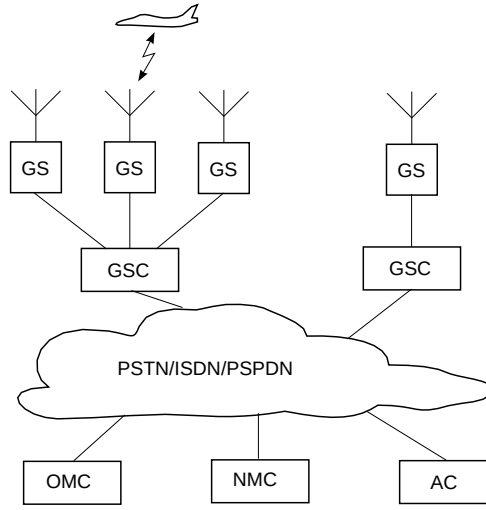


Figure 4.4: Architecture of a complete TFTS network

Handover can be initiated by the mobile or the ground station, and is controlled by signal quality, distance and flight state. The particular ground station selected as a target is the one towards which the mobile station is moving.

The distance between mobile and ground stations is estimated on the basis of signal propagation delay time. This information also determines the network synchronization for the ground stations capable of receiving. Ground stations are linked to the fixed network through *ground switching centres* (GSC) (see Figure 4.4).

The GSC has responsibility for all the ground stations linked to it, and its tasks include mobility management, connection establishment to mobile subscribers, handover control and dynamic frequency management. The TFTS fixed network additionally contains three management components, namely:

- Operations and maintenance centre (OMC)
- Network management centre (NMC)
- Administration centre for billing (AC)

The MoU group produced a coordinated introduction plan for the TFTS ground network to enable the system to be introduced throughout Europe. This effort required a cooperation between telecommunications network operators and airlines.

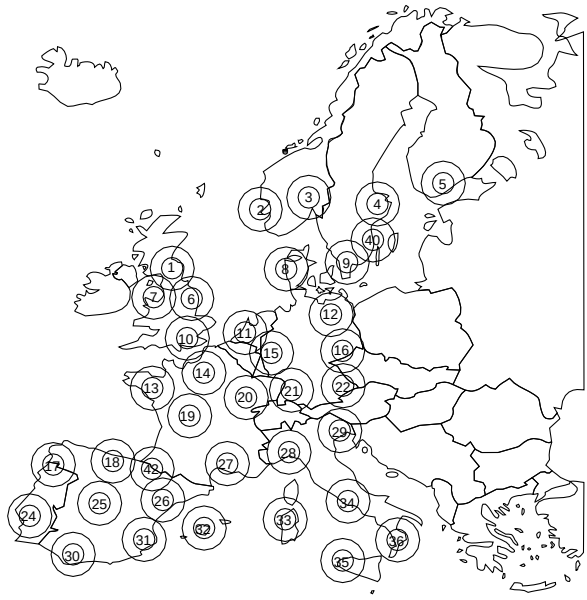


Figure 4.5: Cellular coverage through en-route ground stations in Europe

4.1.5 Ground Stations and Frequency Plan

En-route ground stations are spaced approximately 380 km apart according to a hexagonal grid, with a nominal range of approximately 240 km, which cannot be exceeded for signal propagation reasons (see Figure 4.5).

Cochannel ground stations are planned at a distance of 760 km, and neighbouring channel cells at a distance of at least 600 km. Cell planning is more difficult compared with terrestrial cellular networks because of the need to incorporate flying altitudes.

4.2 The US Digital Cellular System (USDC)

During the 1980s there was an impressive increase in the number of subscribers to the public cellular mobile radio network in the USA. Because approval for the installation of new base stations and antennas is expensive and difficult to obtain in larger cities, only a portion of this increased need for capacity could be accommodated through a reduction in cell sizes. A permanent solution turned out to be the development of a digital system capable of coping with increased capacity without the need for new base stations.

In March 1988 the *Telecommunication Industries Association* (TIA) set up the *TR-45.3* subcommittee to develop the standard for a cellular digital system. This digital system, the *American Digital Cellular System* (ADC),

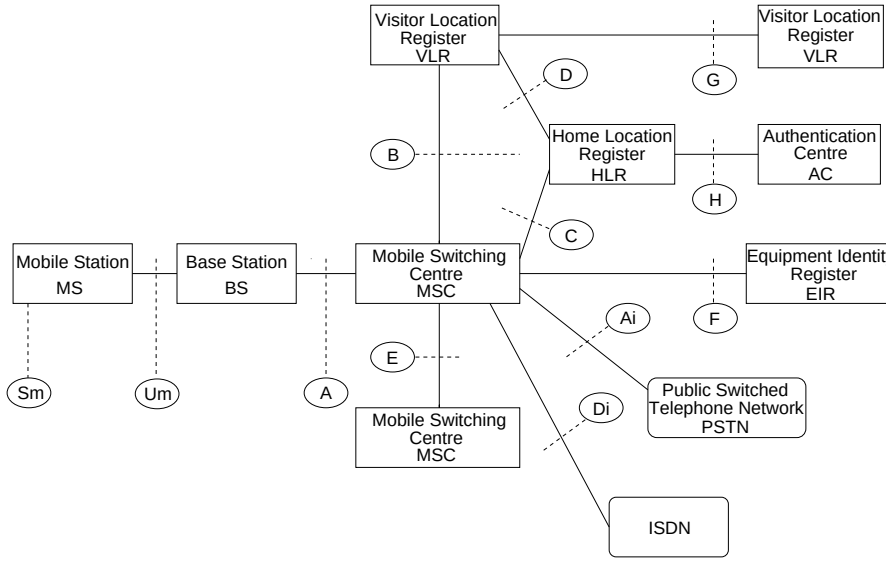


Figure 4.6: Functional architecture of the USDC system

was to support and be compatible with the existing analogue mobile radio network, the *American Mobile Phone System* (AMPS); see [3]. The digital system operates in the frequency range of the analogue AMPS system at the same time, which allows individual channels to change over gradually to digital technology. A characteristic of this system is that terminal equipment can be used for analogue as well as for digital operation (dual-mode). In addition to increased capacity, the ADC standard enables the introduction of new services, such as authentication, a data service and a short-message service, which were not supported by AMPS.

In 1990 the digital standard was accepted by industry as Interim Standard 54 (IS-54). The North American digital system with the architecture illustrated in Figure 4.6 is now called US Digital Cellular (USDC). In addition, a number of standards have been accepted by FCC for the *Personal Communication System* PCS 1900 market, e.g., IS-134.

4.2.1 Technical Data on the USDC System

The USDC system uses the 824–849 MHz frequency band for transmission between mobile station and base station (uplink), and in the reverse direction (downlink) the 869–894 MHz band. The duplex separation between the transmit and receive frequency is therefore 45 MHz. The frequency bands are divided into FDM channels with a 30 kHz bandwidth, thereby providing 832 frequency carriers.

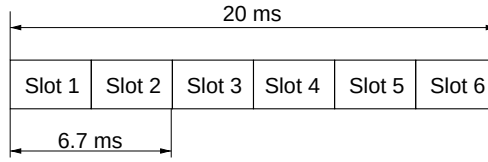


Figure 4.7: Structure of a TDMA frame in the USDC system with half-rate channels

The modulation technique used is $\pi/4$-DQPSK (*Differential Quadrature Phase Shift Keying*), a four-level scheme that, although it produces higher spectral efficiency than GMSK, places a heavy demand on the linearity of the output amplifier. In addition, for optimal detection at the receiver input, filters with a transmission function capable of describing the root of the Nyquist transmission function are required, and this is something that inexpensive filters can only approximate. In contrast to GMSK, $\pi/4$-DQPSK contains different amplitude components. The eight different phase states in $\pi/4$-DQPSK modulation are all in one circle, but the four allowed phase transitions from one phase to another do not run in the circle. This means that not only the phase but also the amplitude is covered in the specifications for modulation.

Like the GSM system, the USDC system operates in time-division multiplexing (TDM) and multiple access (TDMA) mode, albeit with three voice channels being transmitted over one carrier. The length of the TDMA frame is 20 ms and is divided into three time slots each of 6.7 ms duration. The modulation data rate per FDM channel (3 time slots, 30 kHz) is 48.6 kbit/s. After the development and introduction of a half-rate codec, a TDMA frame will contain six time slots (see Figure 4.7) [5].

The USDC system uses a VSELP speech codec (*Vector Sum Excited Linear Prediction*) which, compared with GSM, results in lower source rates. With a full-rate codec, voice coding together with error-protection coding produces an overall transmission rate of 13 kbit/s, whereas the total rate on the SACCH is 0.6 kbit/s.

4.3 CDMA Cellular Radio According to US-TIA/IS-95

TIA Interim Standard 95 was developed by QUALCOMM. Unlike IS-54, which guarantees compatibility of a digital system with analogue, the IS-95 standard defines a CDMA transmission system. It includes the lowest three levels of the OSI reference model. The transmission system of the LEO system Globalstar will be based on the IS-95 standard with modifications (see Section 14.3.3). The physical layer is described below. However, only the

modulators have been standardized but not the demodulators; these can be specified by the manufacturer.

4.3.1 Forward-Link

Forward-link uses coherent QPSK modulation in which transmitter and receiver must be phase-synchronized for demodulation. Walsh sequences are used for channel separation (see Section 2.6.4). A short PN sequence is used for each in-phase and quadrature-phase for the spreading. A long PN sequence individually assigned to the user is used for the traffic channel. Demodulation is carried out through a pilot tone that is also transmitted.

4.3.1.1 Modulator

Figure 4.8 shows the modulator for the forward link. A number of physical channels are available for establishing a connection. The first thing that must be carried out when a mobile station is switched on is synchronization. Phase synchronization and frame synchronization are achieved through the transmission of a pilot tone. The network synchronization is then carried out over the synchronization channel. This involves transmitting the paging channel data rate and power control information. Data for channel allocation is sent over the paging channel. Information is transmitted over the traffic channel.

Pilot channel The all-one Walsh sequence W_0 is combined with a short code and transmitted to the modulator. With a value set of $(0, 1)$ the two codes are added modulo 2, or with a bipolar $(-1, 1)$ approach they are multiplied.

The Walsh sequences are the lines of the Hadamard matrix, and are formed according to the following recursion:

$$H_1 = 0 \quad \text{and} \quad H_{2N} = \begin{pmatrix} H_N & H_N \\ H_N & \bar{H}_N \end{pmatrix} \tag{4.1}$$

in which N must be a power of two and $\bar{H}_N$ is the negation of H_N. The next two matrices are formed in the same way:

$$H_2 = \begin{pmatrix} 0 & 0 \\ 0 & 1 \end{pmatrix} \quad \text{and} \quad H_4 = \begin{pmatrix} 0 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 \\ 0 & 0 & 1 & 1 \\ 0 & 1 & 1 & 0 \end{pmatrix} \tag{4.2}$$

All Walsh sequences of the same matrix are orthogonal to each other. The IS-95 standard uses $2^6 = 64$ Walsh sequences. The Globalstar system will probably use $2^7 = 128$ sequences.

In IS-95 the short code is formed with two irreducible polynomials (the polynomials 121 641 and 117 071 are primitive. Note that because code sequences can be produced with a polynomial and its reciprocal polynomial,

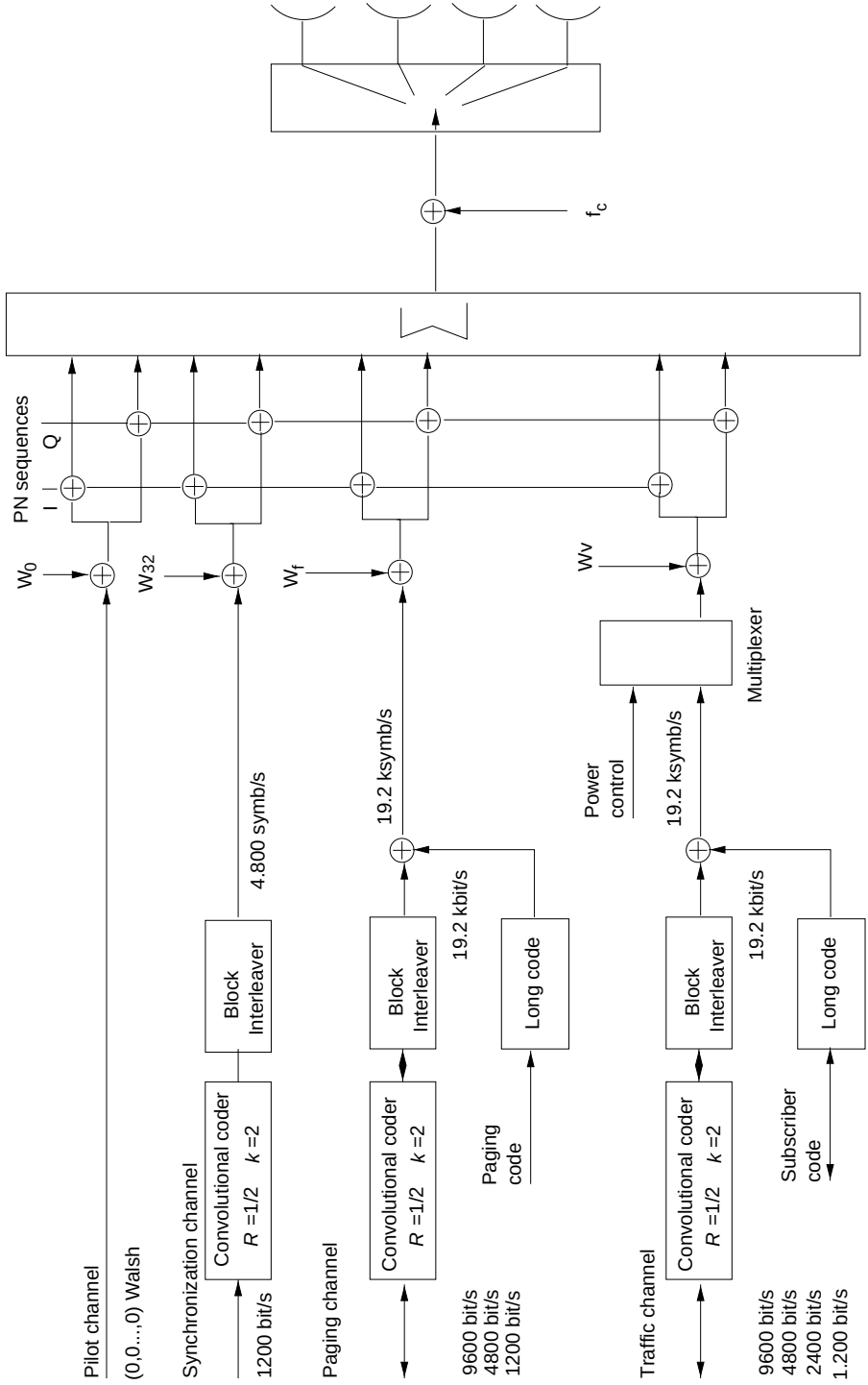


Figure 4.8: Modulator for forward-link

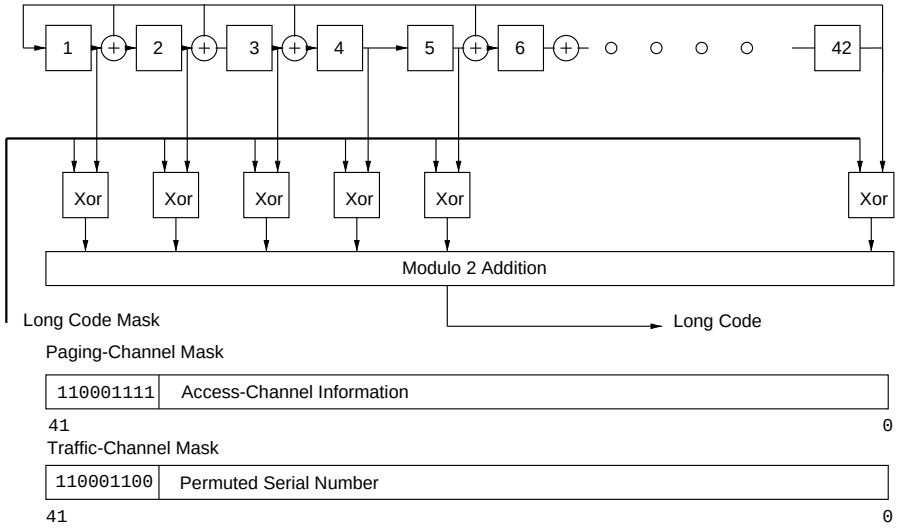


Figure 4.9: Long code generator

only one polynomial is given in the tables [7]). In IS-95 the grade is $n = 15$; in the Globalstar system the grade will probably be $n = 17$.

The polynomials for the in-phase components and the quadrature-phase components in IS-95 are

$$P_I = x^{15} + x^{13} + x^9 + x^8 + x^7 + x^5 + 1 \quad (4.3)$$

$$P_Q = x^{15} + x^{12} + x^{11} + x^{10} + x^6 + x^5 + x^4 + x^3 + 1 \quad (4.4)$$

The short code is the same for the whole system. In Globalstar a code misalignment (different misalignment in the shift register) is used to provide unique identification of the gateway, the satellite and the beam.

The Walsh sequence is spread with the short code at a 1.23 MHz clock-pulse rate over the entire bandwidth and QPSK-modulated.

Synchronization channel The synchronization channel produces data flow at a rate of 1200 bit/s. The data is channel-coded with a $(R = 1/2,\ K = 9)$ convolutional coder, then interleaved and combined with the Walsh sequence W_{32}. The signal is then spread with the short code and QPSK-modulated.

Paging channel Data is channel-coded with a $(R = 1/2, K = 9)$ convolutional coder, then interleaved and spread with a long code. For the channel separation the signal is combined with the W_p Walsh sequence allocated to the paging channel. The signal is then spread with the short code and QPSK-modulated. Figure 4.9 shows the structure of a long-code generator.

This involves setting up a shift register with 42 delay elements, with the outputs linked by a 42-bit long mask. The outputs are added modulo 2 and generate the long code.

Traffic channel The vocoder (standardized in accordance with IS-96), which is capable of producing different data rates as required, delivers the data to the channel coder and the interleaver. Each user has a personal secret key number which forms part of the long code mask for the traffic channel. The long code is linked to the output of the interleaver. On this basis, power control data and traffic channel data are alternatively spread using a user Walsh sequence W_u. The data flow is combined with the short code and QPSK-modulated.

4.3.1.2 Power Control on the Forward-Link

In IS-95 power control is carried out in a closed loop on the forward-link. This requires a periodic reduction in the transmitted power of the base station. The reduction continues until the user notices an increase in the frame error ratio. The user then sends a command for the power to be increased. The measurement increments of power control are relatively small and in the area of 0.5 dB. The dynamics covers an area of ±6 dB. The power changes occur every 20 ms.

4.3.2 Return-Link

Non-coherent orthogonal 64-correlated Walsh modulation is used in the return-link in IS-95. This modulation can be interpreted as FSK modulation with the Walsh sequences corresponding to different frequencies.

The long code is used here for the channel separation. In forward-link Walsh sequences are used for channel separation, whereas here the Walsh sequences are used for modulation. The data is spread with the short code and transmitted using QPSK.

4.3.2.1 Modulator

There are two physical channels: an access channel and a traffic channel, differentiated only by the long code mask. Figure 4.10 shows the modulator for the return-link of the traffic channel.

Access channel A base station receives access requests on the access channel. First a preamble of three frames of 96 zeros per frame is transmitted. Then the user's long code is transmitted. The data rate is always 4.8 kbit/s. Another eight bits containing only zeros are added after each net data frame. The data is channel-coded with a ($R = 1/3$, $K = 9$) convolutional coder, scrambled by the interleaver and modulated orthogonally. The long-code generator ($n = 41$), which is combined with a paging mask, and the short-code generator

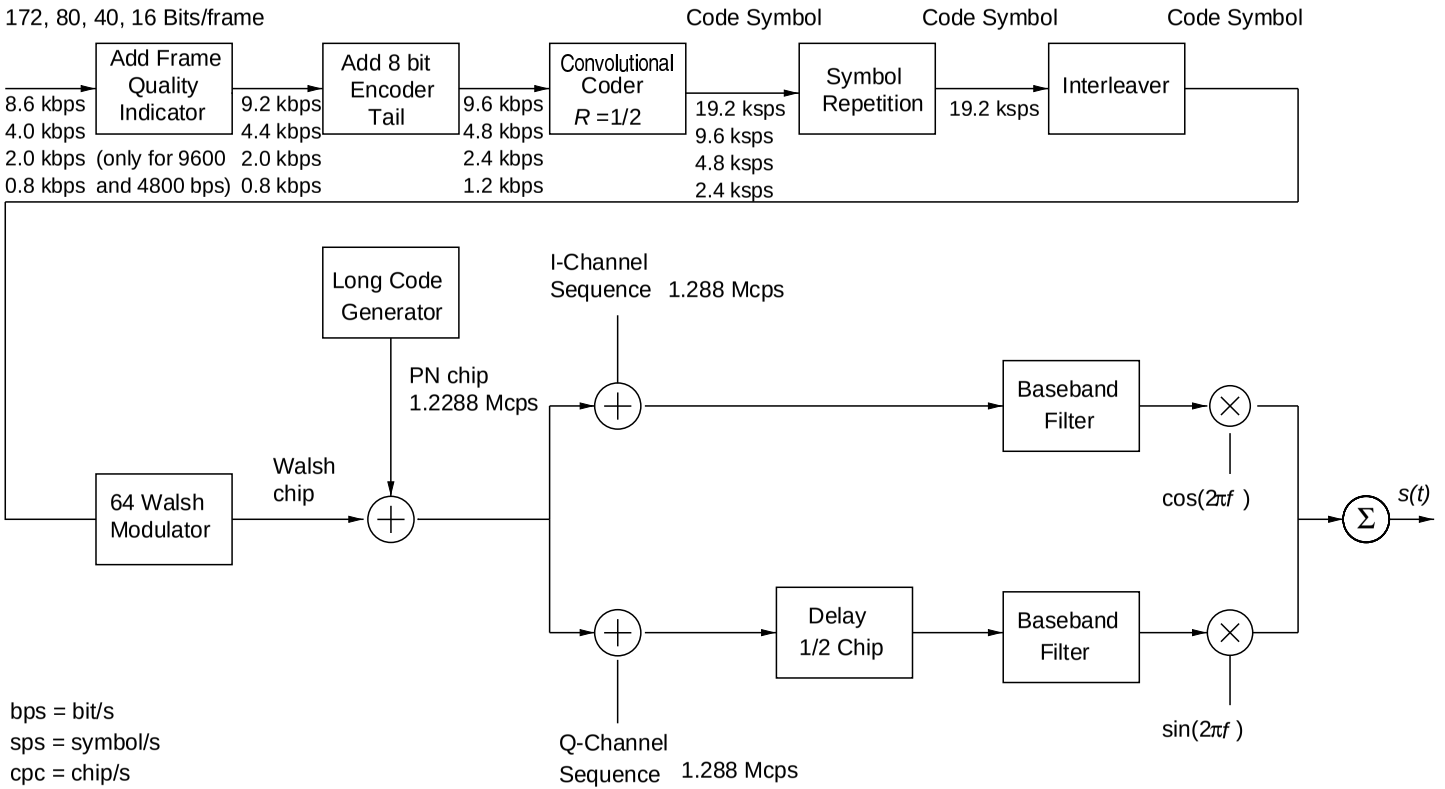


Figure 4.10: Modulator for the return-link

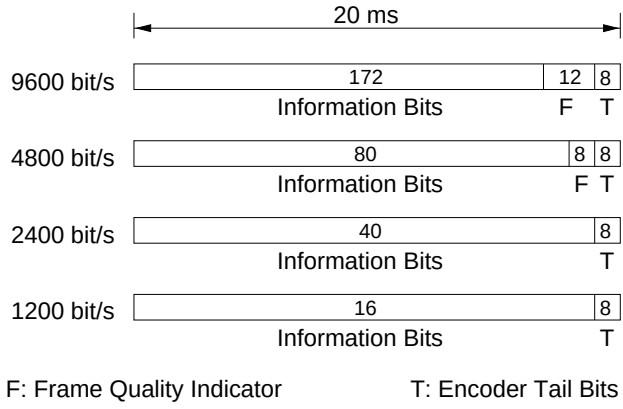


Figure 4.11: Frame structure for different data rates of the IS-95 vocoder

($n = 15$) are used in the spreading. With a bandwidth of 1.2388 MHz the signal is digitally band-pass filtered. At the same time it is sampled at four times the data rate.

Traffic channel Figure 4.11 illustrates the frame structure for the different data rates of the vocoder. The data rate can be altered dynamically in order to adjust to the data volume. For example, with voice transmission the data rate is reduced to 1200 bit/s during intervals when there is no speech.

A CRC check sum is generated at transmission of 9600 bit/s or 4800 bit/s. It has two functions:

1. To protect the net data
2. To provide support in establishing the data rate at the receiver

The CRC polynomials are given in [8]. The data is channel-coded with a ($R = 1/3$, $K = 9$) convolutional coder, interleaved und modulated orthogonally. The user-specific long-code generator ($n = 41$) and the short-code generator ($n = 15$) are used in the spreading.

4.3.2.2 Power Control on the Return-Link

The power is controlled through an interplay between two mechanisms:

- Power control with *open loop*
- Power control with *closed loop*

Each mobile station attempts to estimate free-space attenuation. With IS-95 a tone is sent from the base station on the pilot channel. The power of the pilot tone is measured by the mobile station, and is also used by it to

Table 4.1: Number of subscribers as of June 1997 [9]

	Worldwide	USA
GSM	44 Million	0.5 Million
IS-95 CDMA	5.5 Million	0.6 Million

estimate its own transmitted power. If a strong signal is measured, this means that the mobile user is relatively close to the base station or has an unusually good connection. If there is a quick improvement in the channel state, *open-loop power control* is operating. This can react to very sudden changes (in the Microsecond range). The power of the transmitted signal is controlled analogously, corresponding to the receive power. The power gradation is in the area of 85 dB. However, only the mean value of the transmitted power can be computed. Despite the speedy reaction to power changes, it is not possible to compensate for *Rayleigh fading* with open loop. The reason is that forward-link and return-link are in different frequency bands, and Rayleigh fading occurrences of the two connections are statistically independent of one another. A closed loop is used to suppress the fast Rayleigh fading.

With closed-loop power control the input power is measured at the base station and compared with the desired level of power. Should there be a discrepancy between the two, the mobile station is instructed to carry out a discrete change in transmitter power. The power changes are in the area of 0.5 dB and are transmitted every 1.25 ms. Field tests have shown that this time is sufficient to combat multipath fading.

4.3.3 Experiences Gained with IS-95 CDMA Systems

Seen from today, IS-95 is a narrowband CDMA system when compared with the forthcoming UMTS; see Chapter 5.

Since mid-1992, systems according to TIA/IS-95 standard developed by QUALCOMM have been introduced in the USA by many cellular operators, i.e., just half a year later than GSM in Europe. IS-95 systems have also been established in South Korea and Hong Kong. Seen from June 1997, GSM and IS-95 had both reached a quite different acceptance in terms of the number of subscribers; see Table 4.1 [9].

Until 1997 it had been claimed by QUALCOMM that, owing to the much larger radio distance that could be covered with CDMA signals, the coverage of a given area would only need less than half of the base stations compared with GSM to provide the same quality of service. This has not been proven in the field; e.g., investigations in the number of base stations deployed to serve the same area by different systems (and operators) have resulted in the statistics shown in Table 4.2 [9].

One reason for this result is that radio engineering for a given traffic load is much more difficult with CDMA systems compared with FD/TDMA/FH

Table 4.2: Number of base stations in Tampa/Florida [9]

Operator	Aerial GSM	AT&T D-AMPS	GTE CDMA	Prime Co CDMA
Number of base stations	16	21	23	19

systems (GSM, D-AMPS). With CDMA systems the cells are shrinking with increased traffic load (so-called cell breathing). Even after six years of experience, optimization of CDMA systems in the field, which combines power control, frequency re-use, and error protection (besides others), is still considered a very complex task requiring a lot of time with measurements and adustments to be performed in the running system.

There is no seamless handover available across frequency bands with IS-95 systems, and a number of value-added services attracting many GSM subscribers are still not offered [9]. It has been found that the GSM infrastructure cost is only one-third of that of IS-95.

In spite of this experience, CDMA has been selected as the transmission technique for the radio interface of the UMTS; see Chapter 5. There the situation is somewhat different: much more spectrum is available for UMTS, allowing higher spreading factors, e.g., for speech transmission, and a lot of experience will be available from IS-95 systems when the UMTS is introduced around the year 2003.

Anyway, it appears worth repeating what has been stated in [9] as the result of an extensive investigation of the reasons for deciding to operate IS-95 CDMA systems:

"CDMA is a religion... You get: I believe..."

4.4 The Personal Digital Cellular System (PDC) of Japan

A study of the public mobile radio market in Japan shows a high concentration of users in metropolitan areas such as Tokyo (50%) and Osaka (22%). To eliminate the bottlenecks caused by a lack of frequencies in these areas, the Ministry of Post and Telecommunications (MPT) made the decision in April 1989 to develop a digital mobile radio standard. Compared with existing analogue systems, this digital mobile radio system was to be more cost-effective and offer a higher level of capacity and security, as well as new services.

The new system, formerly called *Japanese Digital Cellular* (JDC) but now referred to as the *Personal Digital Cellular* (PDC) system, was specified by the *Research & Development Center for Radio Systems* (RCR). In addition, there is the *Personal Handyphone System* (PHS), which is a PCS system for the cordless mass-market; see Chapter 11.

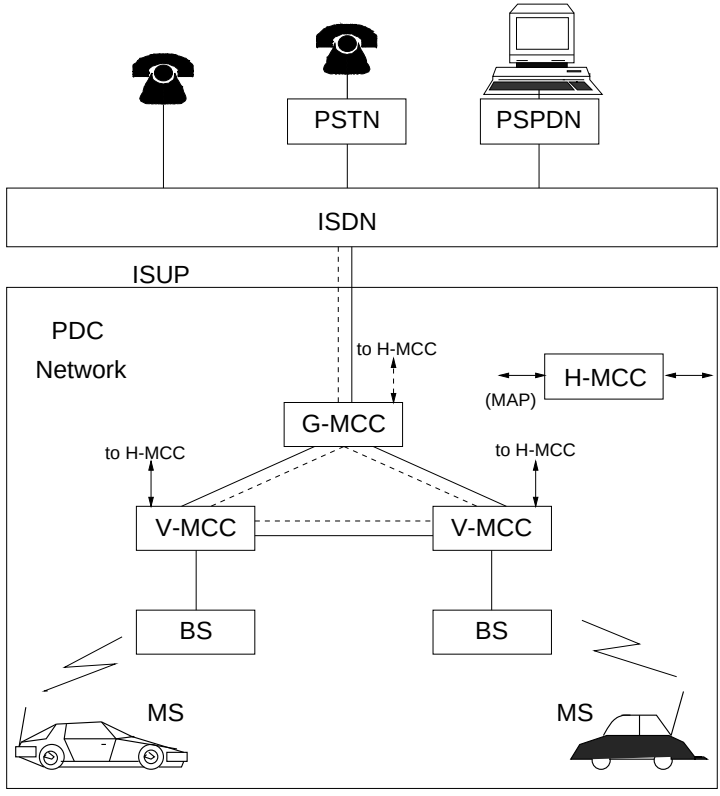


Figure 4.12: The functional architecture of the PDC system

As can be seen in the system architecture shown in Figure 4.12, the *Mobile Communication Control Centers* (MCC) are divided into *Gate-MCC*, *Visit-MCC* and *Home-MCC*, with only the G-MCCs being connected to the fixed network to save on the cost of infrastructure [4].

4.4.1 Technical Data on the PDC System

The technical parameters of the PDC standard are similar to those of the American USDC system, albeit with some important differences.

In Japan the digital system does not directly replace the existing analogue system, because the frequency bands for PDC are above and below those of the analogue system. In contrast to the USDC and GSM systems, mobile stations transmit on a higher frequency (940–960 MHz, uplink) than the base station (810–830 MHz, downlink). Furthermore, additional frequencies were provided in the 1500 MHz frequency band for the PDC system. The duplex separation is 130 MHz in the 800 MHz band and 48 MHz in the 1500 MHz band. The frequency bands themselves are divided into 25 kHz channels.

Table 4.3: Comparison of the technical parameters in the GSM, USDC and PDC systems (assuming GSM full-rate traffic channels)

Parameter	GSM	USDC (ADC)	PDC (JDC)
Frequency range [MHz]			
MS-BS	890–915	824–849	940–960
BS-MS	935–960	869–894	810–830
MS-BS (partial)			1477–1513
BS-MS (partial)			1429–1465
Access method	TDMA	TDMA	TDMA
Duplex method	FDD	FDD	FDD
Duplex separation [MHz]	45	45	130
			48
Channel grid [kHz]	200	30	25
No. of channels	8 (16)	3 (6)	3 (6)
Frequency carrier	124	832	800
No. of traffic channels	$124 \cdot 8$	$832 \cdot 3$	$800 \cdot 3$
Modulation	GMSK	$\pi/4$-DQPSK	$\pi/4$-DQPSK
Speech codec	RPE-LTP	VSELP	VSELP
TCH transmission rate [kbit/s]	22.8	13	11.2
Data rate [kbit/s]	270.8	48.6	42
Min. C/I [dB]	> 9	> 12	> 13
Max. speed [km/h]	250	100	100
User data rate [kbit/s]	9.6	4.8	4.8
User cap. [Erl./km^2/MHz]	[1.1...1.6]	[1.64...2.99]	[2...3.63]

Similarly to the American USDC system, the PDC system uses four-level $\pi/4$-DQPSK modulation [10].

With the use of a full-rate codec, three voice channels are transmitted over one carrier using time-division multiplexing and multiple access (TDMA); once the half-rate codec is introduced, it will be six. The TDMA frame has a duration of 20 ms, and each of the three time slots comprises 280 bits. Since two bits per symbol are transmitted with the $\pi/4$-DQPSK-modulation, the total transmission rate for the three time slots is 42 kbit/s.

Because a VSELP codec is used, the transmission rate (speech coding together with error-protection coding) amounts to a total of 11.2 kbit/s. The signalling data is transmitted over the SACCH at a rate of 0.75 kbit/s.

4.5 Comparison of some Second-Generation Cellular Systems

Table 4.3 provides a comparison of the technical parameters for the radio interfaces of the digital public mobile radio systems (GSM, USDC and PDC).

Studies show that the user capacity in the PDC system is almost double that of the GSM system and that the USDC system is about 1.5 times that of the GSM system [6]. The high spectral efficiency in the PDC and USDC systems is essentially attributed to the different modulation techniques and different speech codecs used in these systems.

The advantages of GSM over USDC and PDC are higher user data rates and a lower minimal carrier-to-interference ratio. $\pi/4$-DQPSK transmitters produce considerable broadband noise. Because of an efficient speech codec, the channel and source coding, and interleaving, the GSM system achieves a higher level of voice quality than the other two systems. Comparisons of the spectral efficiency of mobile radio systems are not appropriate, mainly because the systems are designed for different qualities of service yet this parameter is usually ignored in any comparative analysis.

Unlike the GSM system, the PDC system only uses antenna diversity and no equalizers. In multipath propagation antenna diversity can be more advantageous than an equalizer, which has a high power consumption because of the complexity of the calculations.

The GSM recommendations have been successfully accepted in Europe as well as internationally. What was important for the success of GSM was European union, open standardization and early availability of the system.

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5
Third-Generation Cellular: UMTS*

Dramatic developments have been taking place in the mobile radio area all over the world during the last couple of decades. Mobile communications is one of the fastest growing markets in the telecommunications area. According to projections, there will be a linear increase in the number of subscribers to the major GSM networks operated in Europe by the end of the decade.

The political environment in Europe is the main reason for the rapid development. Without a free exchange of information, the concept of an internal market striving for a free flow of goods between EU states would be inconceivable. This was the line of thinking behind the liberalization and deregulation of the telecommunications industry, which promoted and accelerated competition and opened up the markets.

Another reason for this rapid development is the advances being made in the microelectronics, microprocessor and transmission technology areas. These advances are enabling the use of ever smaller terminal equipment, with computing power previously only possible with mainframes, and with low power consumption—factors that have improved customer acceptance.

In Europe the development of uniform standards, the introduction of European-wide radio systems and the participation of industry in the standardization process through the establishment of ETSI have further contributed to the widespread success of mobile communications.

The chronological development of different kinds of mobile radio networks which conform to different user needs is presented in Figure 1.2 [23].

The systems that fall into the category of first-generation mobile communications systems in which mobility is only ensured within a specific network area are the different analogue cellular systems (e.g., C-Netz, NMT), cordless systems (CT1/CT2) and various national paging systems.

The second generation includes the digital systems such as GSM, DCS 1800, USDC, PDC, IS-95 and ERMES, which underwent further development and were expanded or were first introduced during the first half of the 1990s.

Along with these public cellular systems that provide PSTN/ISDN services at a mobile terminal, there are other systems that fall into the second-generation or the transitional category between the second and third generations, and cater specifically to mobile or moving applications. These include trunked radio (ETSI/TETRA, see Section 6.3), cordless communica-

*With the collaboration of Arndt Kadelka, Matthias Lott and Peter Seidenberg

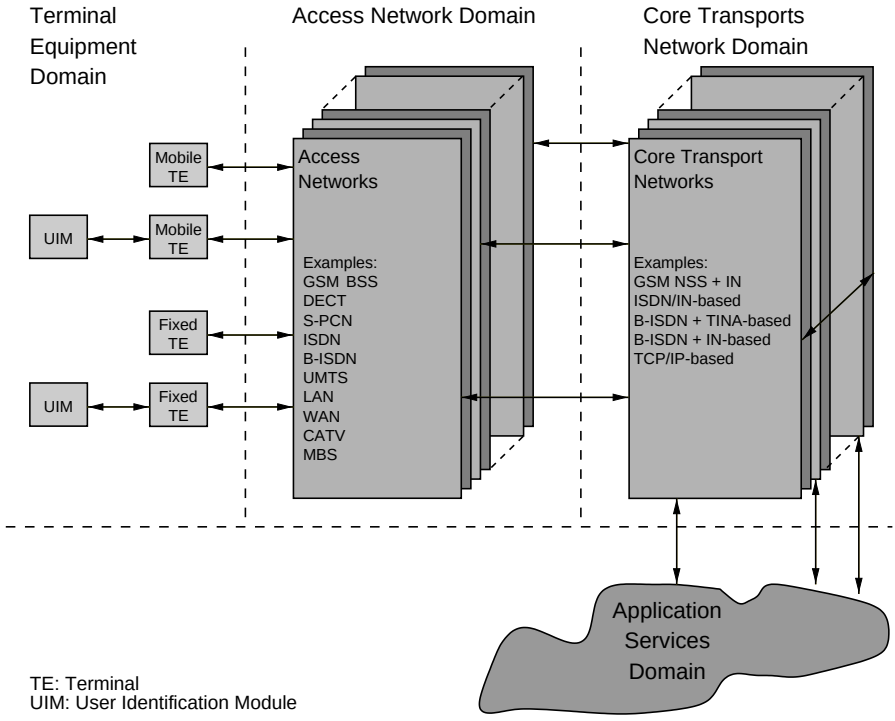


Figure 5.1: Global multimedia mobility architecture

tions (ETSI/DECT, see Chapter 9, and the *Personal Handyphone System*, PHS, see Chapter 11), local broadband communications (ETSI/HIPERLAN 1, see Section 13.1, IEEE 802.11, see Section 13.9), wireless ATM systems (ETSI/BRAN, see Section 12.1.5), mobile personal satellite radio (IRIDIUM, Globalstar, see Chapter 14) and other systems integrating aspects of these systems.

Third-generation mobile radio systems, which use intelligent networks to incorporate public mobile radio services that previously were operated separately, are already being developed today. Under the designation *Global Multimedia Mobility* (GMM), ETSI is developing an architecture that defines mobile radio networks as the access networks to an integral transport platform that is based on broadband (B)-ISDN and provides mobility-supported value-added services (see Figure 5.1). It is planned that these future standard mobile communications networks (UMTS and FPLMTS or IMT 2000), which aim to support the services of the terrestrial broadband ISDN, will lead to a universal worldwide public mobile radio system, which is expected to be operational by the year 2003.

The main characteristics of third-generation mobile radio systems are [1]:

- Support of all features currently being offered by different systems.
- Support of new services with high quality of service, the same as in the fixed network.
- High capacity, which will support high market penetration.
- High spectral efficiency.
- Lightweight, small (pocket-sized) and inexpensive handheld equipment for mobile telephone use.
- High security, comparable to that of the fixed network.

High demands are being placed on the third-generation systems, e.g.:

- Services (voice and data, teleservices, bearer services, supplementary services).
- Different bit rates (low bit rates for voice; data rates up to 2 Mbit/s).
- Variable bit rates and packet-oriented services.
- Use of different sized cells (macro, micro, pico) for *indoor* and *outdoor* applications, with seamless handover between indoor and outdoor base stations.
- Operation in non-synchronous base station subsystems.
- Advanced mobility characteristics (UPT, see Chapter 15; roaming, handover, etc.).
- Flexible frequency management.
- Flexible management of radio resources.

5.1 UMTS (Universal Mobile Telecommunications System)

In Europe work continues to be carried out on the development of a third-generation mobile radio system called UMTS (*Universal Mobile Telecommunications System*) in the EU programmes RACE (1989–1994) (*Research and Development in Advanced Communications Technologies in Europe*) and ACTS (1995–1998) (*Advanced Communication Technologies and Services*) in cooperation with ETSI. Work on UMTS is also being done in COST (*European Cooperation in the Field of Scientific and Technical Research*) projects [20].

The technical subcommittee (STC) SMG 5 at ETSI has been given the responsibility for producing the UMTS standard. Other SMG subcommittees that are currently still working on the GSM 2+ standard will eventually

become involved in the standardization of UMTS, e.g., SMG 2. SMG 5 will then take over the creation of the UMTS standard and the coordination of the standardization activities. There is also the UMTS Forum, comprising the European signatories to the *UMTS–Memorandum of Understanding of the Introduction of UMTS* defined in 1996.

The main tasks of SMG 5 are [2, 15]:

- Study and definition of services, system architecture, the air interface and the network interfaces for UMTS.
- Generation of basic technical documentation for UMTS.
- Coordination of ETSI and of SMG regarding UMTS.
- Cooperation and coordination with the ITU for the definition of a worldwide standard on the basis of UMTS/FPLMTS/IMT 2000.
- Cooperation with European research programmes.

The aim of the UMTS concept is to provide users with a handheld terminal that will cover all areas of application—at home, in the office, en route by car, in a train, in an aircraft and as a pedestrian. UMTS will therefore offer a common air interface that will cover all fields of application and have the flexibility to integrate worldwide the different mobile communications systems available today, such as mobile telephone and telepoint, trunked radio, data radio, and satellite radio systems, into one system.

What will play an important role in UMTS is the concept of intelligent networks (IN) that will provide call charging and mobility management for the localization and routing of calls across networks operated by different service providers and operators. UMTS will be the first system to offer mobile users roaming during an existing connection, with handover between networks with different applications and different operators [17].

UMTS will offer transmission capacity comparable to ISDN for services such as video telephony and wideband connections, and will support the service concept *Universal Personal Telecommunications* (UPT) [4]; see Chapter 15. With UMTS it will be possible to transmit voice, text, data and images over one connection, and subscribers will be assigned a personal telephone number that will allow them to be reached anytime, anywhere in the world.

The first series of standards for UMTS has been completed in March 1999. The projection is that UMTS, which according to Appendix D will use the frequency band between 1.885 and 2.2 GHz, will be introduced around 2003. However, the UMTS Forum has a preference for the frequencies indicated in Figure 5.2, staggered timewise as shown, and is promoting the *refarming* of bands previously used for other purposes (see Appendix D) and working towards including asymmetrical bands along with the symmetrical ones. The planned frequency allocations to IMT 2000/UMTS are shown in Figure 5.3.

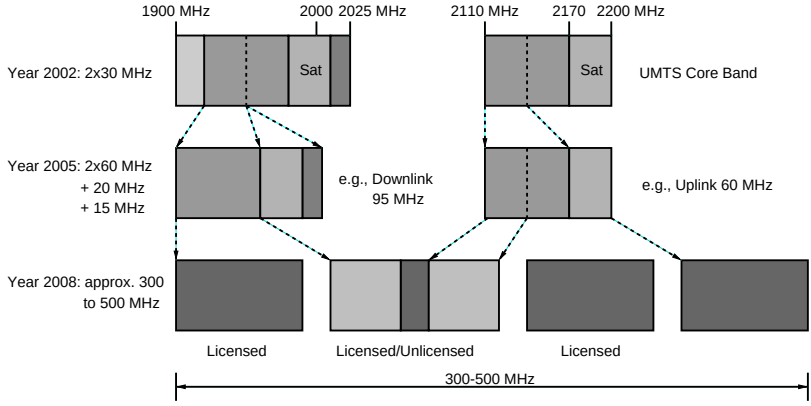


Figure 5.2: UMTS frequency spectra, UMTS Forum's perception of timetable for development

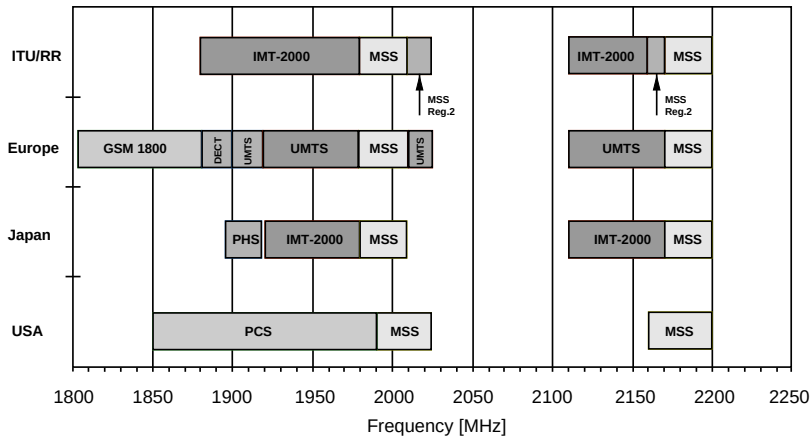


Figure 5.3: Spectrum Allocation

Originally it was planned to specify one air interface only able to cover all the different services and applications aimed at. From the decision made in January 1998 (see Section 5.7.5), it is now clear that at least two air interfaces will be specified—one based on paired bands with *frequency division duplexing* (FDD) transmission, and another air interface operating in a single band with *time division duplexing* (TDD) transmission.

Both standards will use DS-CDMA for radio transmission and channel access, and are addressed as FDD-CDMA and TDD-CDMA systems respectively. From the viewpoint of *intellectual property right* (IPR), there are still those in Europe proposing to stay with F/TDMA as the basis for UMTS and to make no use of CDMA, since QUALCOMM is the owner of some CDMA key patents but is not willing to license their use under the so-called fair rules

established by ETSI. In fact, the EDGE proposal (see Section 3.11) submitted by Ericsson to ITU-R is capable of providing wideband services compatible to GSM 2+.

The main driving force towards UMTS at present comes from manufacturers aiming to introduce new products into the market and operators aiming to get under the label UMTS access to more bandwidth for voice services only. Mobile data was still only a few percent of business in 1998. The European Commission has issued guidelines for the licensing of UMTS bands to operators demanding that 50 % of the services offered should be data services for multimedia applications.

The demand for more bandwidth can of course easily be covered by assigning UMTS frequency bands to be used by GSM networks, and does not need the introduction of a new air interface.

5.2 FPLMTS (Future Public Land Mobile Telephone System); IMT 2000 (International Mobile Communications at 2000 MHz)

In 1985 the CCIR (see Annex B.1.2) set up a working group, the Task Group 8/1 (previously IWP 8/13), for the purpose of specifiying all the requirements and system parameters for a future public land mobile telecommunication system (FPLMTS). The following requirements for an FPLMTS were drawn up by the working group [5, 13, 19]:

- Small, lightweight handheld equipment.
- Worldwide use of terminal equipment, i.e., uniform frequencies worldwide.
- Integration of different mobile radio systems and international roaming.
- Integration into the fixed telephone networks (ISDN compatibility).
- Integration of mobile satellite radio.
- Use of terminal equipment on land, in the air and at sea.

As with the UMTS, the aim with the FPLMTS is to integrate all existing services (mobile telephony, cordless telephony, paging, trunked radio, etc.) into one service. Many of the aspects of FPLMTS are the same as those of UMTS; however, since the ITU activities are globally based, there are some differences between the two systems. For example, FPLMTS defines several air interfaces for dealing with the different requirements of densely populated areas (e.g., in Europe) versus sparsely populated areas (third world countries) [22]:

- R1: radio interface between *mobile station* (MS) and *base station* (BS)
- R2: radio interface between *personal station* (PS) and *personal base station*
- R3: radio interface between satellite base station and mobile earth station (MES)
- R4: additional air interface for paging FPLMTS terminals

The plan is to use FPLMTS as a temporary or permanent substitute for fixed networks in developing countries and in rural areas where it is not economically feasible to set up fixed networks.

At W(A)RC 1992 a spectrum of 230 MHz in the frequency bands 1885–2025 MHz and 2110–2200 MHz was allocated to the FPLMTS system worldwide. These frequency bands were not exclusively reserved for FPLMTS, and can also be used in other systems. So in Europe, for example, the lower part of the allocated frequency band is occupied by GSM 1800 and the DECT system. The UMTS Forum is now requesting that 500 MHz starting from 1900 MHz be reserved for symmetrical and asymmetrical connections (see Figure 5.2).

The earliest date being envisaged for the operation of FPLMTS is sometime between 2000 and 2005, the same as for UMTS. Since about 1995, FPLMTS has often been referred to as IMT 2000, but both designations refer to the same system operating around 2000 MHz.

5.3 Services for UMTS and IMT 2000

ETSI has published a preliminary list of services [10] that are to be supported by UMTS and are based on the ITU-R/CCIR recommendations for FPLMTS and the specifications of various European research projects of the RACE 2 programme. These UMTS-supported services are described below.

5.3.1 Carrier Services

UMTS should be able to support ISDN as well as broadband ISDN bearer services. The following services are to be integrated [10]:

- Circuit-switched services:
  - Transparent 64, 2·64, 384, 1536 and 1920 kbit/s with user data rates of 8, 16 and 32 kbit/s
  - Voice transmission
  - 3.1, 5 and 7 kHz audio transmission
  - Alternative voice or transparent data transmission with user data rates of 8, 16, 32 and 64 kbit/s

- Packet-switched services:
  - Virtual calls and permanent virtual channels
  - Connectionless ISDN
  - User signalling

Broadband (B) ISDN services with a transmission rate of 2 Mbit/s (so-called *wideband services*) are also to be offered by UMTS to mobile users. According to CCITT, these services will be classified as interactive or distribution services.

Interactive services fall into the category of conversational services, message services or interrogation services. Conversational services are implemented through end-to-end connections, which can be either symmetrical bidirectional, asymmetrical bidirectional or unidirectional. Message services offer communication between users that is not time transparent. Interrogation services are used for the inquiry and receipt of centrally stored data.

With distribution services information can be transmitted continuously from one central location to any number of users, with the users unable to influence the start or the end of a transmission. Another distribution service offers users the possibility of influencing the start of the information transmission.

Asynchronous Transfer Mode (ATM) was specified by ETSI as the transmission technology for these B-ISDN services in the fixed (core) networks. In order to derive requirements for the radio interface from the bearer services being supported, ETSI, in accordance with the functional descriptions of B-ISDN and the ATM adaptation layer (AAL) (see Section 12.2.4), divided the bearer services into four classes [11]. These four classes of bearer services differ from each other in their time responses, bit rates and types of connection. Maximum bit ratio, maximum bit-error probability and maximum delay time are specified within each class of bearer service for the different communication scenarios.

5.3.2 Teleservices

The teleservices to be supported by UMTS are divided into three classes [10]:

1. Teleservices that already exist in the fixed network in accordance with ITU-T/CCITT recommendations of the E, F and I series:
   - Telephony:
     - Voice
     - Inband facsimile (telefax groups 2 and 3)
     - Inband data transmission (using modem)
   - Teleconferencing:
     - Multiparty, added value services
     - Group calls
     - Acknowleged group calls
     - Multiple calls

2. UMTS teleservices and applications, e.g.:
   - Audio and video transmission
   - Paging
   - Broadcast services
   - Database inquiries
   - Data transmission
   - Directory services (e.g., telephone book)
   - Mobility services (e.g., navigation or localization)
   - Electronic mail
   - Emergency calls
   - Emergency call broadcasts
   - Short-message services:
     - Initiated by user
     - Terminated by user
     - Voice messages
     - Facsimile
     - Electronic mail
   - Teleaction services (e.g., remote control)
   - Teleshopping
   - Video monitoring
   - Voice messages
3. The services with the largest need for bandwidth are multimedia (MM) and interactive multimedia (IMM), such as data, graphics, images, audio and video, and combinations thereof. With UMTS it should be possible to use more than one of these media at the same time. Multimedia allows the transmission of more than one type of information, e.g., video and audio information. No further specifications exist yet for this service [10].

5.3.3 Supplementary Services

In the standardization of supplementary services a differentiation has principally been made between traditional non-interactive PSTN/ISDN services and personalized interactive supplementary services. The service provider has the option of making these services accessible to user groups or to individual users. The following classes of supplementary services have been proposed in accordance with the GSM and ISDN standards:

Number identification, e.g., abbreviated dialling, protection against undesirable calls, calling party identification

Call offering, e.g., call forwarding

Call termination, e.g., call holding

Multiparty communication, e.g., conference call

Group communication, e.g., communication in closed user groups

Billing, e.g., credit balance

Additional information, e.g., user-to-user signalling

Call rejection, e.g., blocking all incoming calls

A list of different service attributes is available in [10].

5.3.4 Value-Added Services

Personal mobility Using a smart card, subscribers are able to transfer their telephone numbers to any terminal.

Virtual home environment (VHE) and service portability This allows the users to set up their own personalized service portfolios and use them in any other network. VHE emulates those services that are not actually offered in the visited network, so that users notice nothing differently from their own home network environments. Moreover, this is how the preliminary UMTS services are provided.

Bandwidth-on-Demand This offers an efficient use of resources for services that have heavily varying requirements for transmission bandwidth, such as short-message services and video. Furthermore it allows users the independent option of selecting between a higher bandwidth for a maximum quality of service or a lower bandwidth for more favourable costs.

5.3.5 Service Parameters

A service is characterized by different parameters, some of the most important being:

- Net bit rate
- Usage level
- Symmetry of a service
- Coding factor
- Maximum bit-error ratio based on channel decoding
- Maximum delay allowed in data transmission

The net bit rate is the product of the average number of bits that have to be transmitted within a certain period of time.

The delay parameter describes how long a waiting time is allowed in the transmission of these bits. For example, a voice service requires a small delay whereas a packet-data transmission has minimal requirements for the delay times of individual packets. However, data transfer requires a considerably lower bit-error ratio than a voice service, because the redundancy of the voice codec can be fully utilized. A higher coding factor is needed for achieving a

Table 5.1: Quality of service parameters

Service	Call duration	Data rate [kbit/s]	Residual bit-error ratio	Delay [ms]
Telephony				
– Voice	2 min	8–32	10^{-4}	40
– Teleconferencing	1 h	32–128	10^{-4}	40
Video telephony	2 min	64–384	10^{-7}	40–90
Video conferencing	1 h	384–768	10^{-7}	90
Message services				
– SMS and paging	cl	1.2–9.6 (1.2–2.4 type)	10^{-6}	100
– Voice mail	2 min	8–32	10^{-4}	90
– Facsimile mail	1 min	32–64	10^{-6}	90
– Video mail	tbd	64	10^{-7}	90
– e-mail	cl	1.2–64	10^{-6}	100
Distribution services	tbd	1.2–9.6 (2.4 type)	10^{-6}	100
Database use	tbd	2.4–768	10^{-6}	200+
Teleshopping	tbd	2.4–768	10^{-6}	90
Electronic mail	tbd	2.4–2000	10^{-6}	200
Message dist.	cl	2.4–2000	10^{-6}	300
Tele-action services	tbd	1.2–64	10^{-6}	100–200

tbd to be defined cl connectionless

lower bit-error ratio in order to protect data during transmission over a radio channel.

The usage level parameter describes how often a connection is being used to transmit data. For example, the usage level of a voice service is less than 0.5 because a user is generally either listening or speaking.

A service is also defined by its symmetry. This value determines which bandwidth is required for a connection in one or the other direction. The voice service is an example of a symmetrical service, because the same bandwidth is used for both speaking and listening. Internet browsing (e.g., world wide web, WWW) is a typical example of an asymmetrical service, because it requires considerably less bandwidth for requesting than for receiving data. Table 5.1 lists the characteristics of some of the services.

5.3.6 Service-Specific Traffic Load

The effective service bandwidth can be calculated from the data of the service parameters net bit rate, symmetry and coding factor [12]. The service bandwidth describes the bandwidth used to provide a particular service.

The traffic generated by the use of a service is calculated by taking the average duration of this usage and the frequency of usage. The effective call

duration T_{eff}, which is calculated on the basis of the usage level N and the average call duration T_{call}, is produced as

$$T_{eff} = NT_{call} \tag{5.1}$$

For the systems being planned, the frequency of usage of a service can only be estimated. It is measured in BHCA (*busy hour call attempts*), and indicates the average frequency of the usage of a service by a user during the peak traffic hour.

If it is known which portion of the overall usage of services is an individual service then it is possible to calculate the effective bandwidth needed by a user. The share of the service in the overall usage is then indicated with the penetration D. The penetration varies with different operating environments (see Section 5.5.1). The traffic produced by a user utilizing a service is calculated in *Equivalent Telephony Erlang* (ETE) [12]:

$$\frac{\text{ETE}}{user} = T_{eff} \times \text{BHCA} \times D \times \frac{service\,bandwidth}{telephony\,bandwidth} \tag{5.2}$$

An ETE therefore corresponds to an Erlang of voice service with a transmission bandwidth of 16 kbit/s. This equation was used to produce an example of the traffic load for voice telephony, video telephony and the facsimile service.

The throughput represents the speed at which the user data is transmitted. This data quantity is increased by a constant factor through the coding used for error detection and correction. Finally consideration must be given to the form of symmetry. For example, with the telephony services data is transmitted in both directions, whereas with the facsimile service it is mainly in one direction.

5.3.6.1 Voice Telephony

Voice telephony is a symmetrical service with a usage level of 0.5 or less. The net bit rate of the voice codec is 16 kbit/s. Since the requirements for bit error ratio are low, a coding factor of 1.75 is sufficient. For the average call duration 120 s is assumed. This equates to an effective service bandwidth of 56 kbit/s and an effective call duration of 60 s (see Table 5.2).

The estimated values for penetration D and for the frequency of calls during a busy hour produce the ETE/user values shown in Table 5.3 for the voice service in different communications environments (see Section 5.5.1).

5.3.6.2 Video Telephony

Video telephony is a symmetric service and has a usage level of one, i.e., transmission is always in both directions of a connection. The effective service bandwidth for the video telephony service is 384 kbit/s and the effective call duration 2 min (see Table 5.2).

The estimated values for the penetration D and for the BHCA produce the ETE/user values in Table 5.4 for different communications environments.

Table 5.2: Service bandwidth and effective call duration of some services

	Voice telephony	Video telephony	Fax service
Throughput [kbit/s]	16	64	64
Coding factor	1.75	3	3
Symmetry	2	2	1.1
⇒ Service bandw. [kbit/s]	56	384	211.2
Usage level	0.5	1	1
Call duration (average) [s]	120	120	156
⇒ Eff. call duration [s]	60	120	156

Table 5.3: Calculation of traffic load for voice telephony

Operating environment	D	BHCA per user	ETEs per user
Business use indoors	0.5	1.0	$8.33 \cdot 10^{-3}$
Residential area	0.3	0.13	$6.50 \cdot 10^{-4}$
City centre, in vehicle	0.4	0.5	$3.33 \cdot 10^{-3}$
City centre, pedestrian outdoors	0.4	0.5	$3.33 \cdot 10^{-3}$
Aircraft	0.4	0.5	$3.33 \cdot 10^{-3}$
Local high bit rate	0.5	1.0	$8.33 \cdot 10^{-3}$

Table 5.4: Calculation of traffic load for video telephony

Operating environment	D	BHCA per user	ETEs per user
Business use indoors	0.13	1.0	$2.97 \cdot 10^{-2}$
Residential area	0.08	0.13	$2.67 \cdot 10^{-3}$
City centre, in vehicle	0.04	0.5	$4.57 \cdot 10^{-3}$
City centre, pedestrian outdoors	0.04	0.5	$4.57 \cdot 10^{-3}$
Aircraft	0.04	0.5	$4.75 \cdot 10^{-3}$
Local high bit rate	0.13	1.0	$2.97 \cdot 10^{-2}$

5.3.6.3 Facsimile

A throughput of 64 kbit/s is assumed for the facsimile service, which corresponds to the transmission rate of the facsimile service currently being offered by ISDN. The facsimile service is an asymmetrical service with a coding factor of 3. This produces an effective service bandwidth of 211.2 kbit/s. The effective call duration is 156 s (see Table 5.2).

The estimated values for penetration D and for BHCA produce the ETE/user values in Table 5.5 for different communications environments.

5.3.6.4 Resultant Overall Traffic Loads

The procedures presented in the sections above can be used to calculate the traffic generated by a user in each of the services listed. Table 5.6 gives the total traffic generated by a user in the different communications environments

Table 5.5: Calculation of traffic load for facsimile services

Operating environment	D	BHCA per user	ETEs per user
Business use indoors	0.3	0.06	$2.94 \cdot 10^{-3}$
Residential area	0.15	0.03	$7.35 \cdot 10^{-4}$
City centre, in vehicle	0.1	0.002	$3.27 \cdot 10^{-5}$
City centre, pedestrian outdoors	0.1	0.002	$3.27 \cdot 10^{-5}$
Aircraft	0.15	0.002	$4.90 \cdot 10^{-6}$
Local high bit rate	0.3	0.06	$2.94 \cdot 10^{-3}$

Table 5.6: Resulting total traffic loads

Operating environment	ETE/user Voice+ video+fax	Total ETE/user	User density (per km^2)	Total traffic density [ETE/km^2]
Business use indoors	$4.10 \cdot 10^{-2}$	$4.92 \cdot 10^{-2}$	180000	$8.85 \cdot 10^{3}$
Residential area	$3.76 \cdot 10^{-3}$	$4.52 \cdot 10^{-3}$	380	1.72
City centre car telephone	$7.94 \cdot 10^{-3}$	$9.52 \cdot 10^{-3}$	2050	$1.95 \cdot 10^{1}$
City centre pedestrian	$7.94 \cdot 10^{-3}$	$9.52 \cdot 10^{-3}$	730	6.95
Aircraft	$7.91 \cdot 10^{-3}$	$9.49 \cdot 10^{-3}$	0.24	$2.28 \cdot 10^{-3}$
Local high bit rate	$4.10 \cdot 10^{-2}$	$4.92 \cdot 10^{-2}$	108000	$8.85 \cdot 10^{3}$

(see Section 5.5.1). If a specific user density is assumed for each communications environment [10] then it is always possible to arrive at a value for the traffic load.

This traffic load describes in ETEs the traffic originating from an area. The requirement for frequency spectrum can be calculated if an assumption is made on the efficiency of the radio interface (see Section 5.4).

5.4 Frequency Spectrum for UMTS

This section presents the UMTS Forum assessments on the frequency spectrum required for UMTS [12]. They are based on estimates of market penetration, future user density, service characteristics and characteristics of the radio interface.

In determining the bandwidth needs the UMTS Forum makes its assumptions based on the breakdown of different categories of service shown in Figure 5.4. In addition, assumptions are made on anticipated user numbers in relationship to the communications environment. The figures for the year 2010 are given in Table 5.7.

The service characteristics compiled in Table 5.8 are also taken into account, with a 16 kbit/s voice codec assumed. The voice service is a symmet-

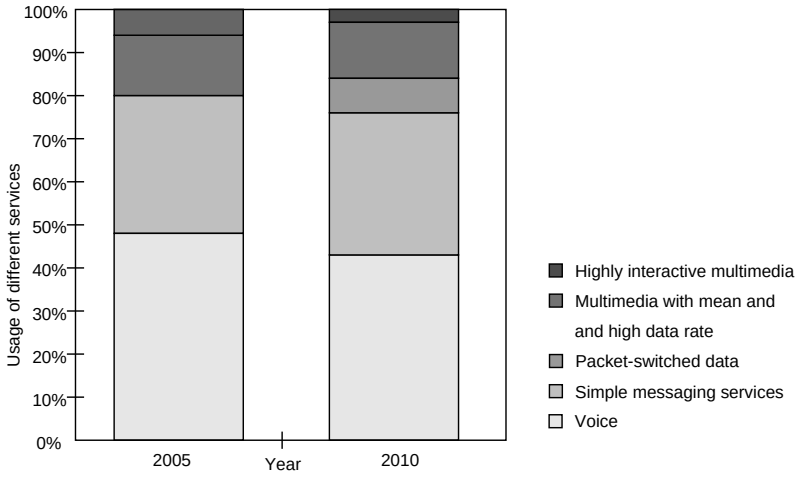


Figure 5.4: Anticipated service spectrum for UMTS

Table 5.7: User density in the year 2010

Environment	User density [per km^2]
City (indoors)	180 000
Suburbs (indoors or outdoors)	7 200
City, pedestrian	108 000
City, auto	2 780
Rural areas (total)	36

rical service with the same transmission rates on the uplink and the downlink. Simple message services are those services that are similar to the SMS (*Short-Message Service*) in GSM. The asymmetric multimedia services (MM) represent typical Internet services (WWW using the http protocol), whereas the interactive multimedia service represents a symmetrical connection such as is required for video conferencing.

Together with the ratio of the average number of active users to the overall number, measured during the busy hour (see Table 5.9), the bandwidth requirements for UMTS can be calculated from the information supplied in the service characteristics and user density (see Table 5.10).

The projected bandwidth requirements for each service for the years 2005 and 2010 are presented in Figure 5.5.

The maximum requirement for bandwidth projected for the year 2010 is 554 MHz for traffic bands and 28 MHz for guard bands. The basic standards for UMTS have been completed in March 1999, and UMTS itself is expected to be introduced in about 2003. The UMTS Forum has a preference for the frequencies given in Figure 5.2, staggered timewise as shown, with bands

Table 5.8: Overview of service characteristics

Service	Net rate [kbit/s]	Coding factor	Symmetry	Eff. call duration [s]	Service bandwidth [kbit/s]
High interactive MM	128	2	1/1	144	256/256
High data rate MM	2000	2	0.005/1	53	20/4000
Med. data rate MM	384	2	0.0026/1	14	20/768
Packet-sw. data	14	3	1/1	156	43/43
Simple mess. serv.	14	2	1/1	30	28/28
Voice	16	1.75	1/1	60	28/28

Table 5.9: Number of calls in a city during busy hour

	2005			2010		
Service	Business	Ind.	Outd.	Business	Ind.	Outd.
High interact. MM	0.12	0.06	0.004	0.24	0.12	0.008
High data rate MM	0.12	0.06	0.004	0.12	0.06	0.004
Med. data rate MM	0.12	0.06	0.004	0.12	0.06	0.004
Packet-sw. data	0.06	0.03	0.002	0.06	0.03	0.002
Simple mess. serv.	0.06	0.03	0.002	0.06	0.03	0.002
Voice	1	0.06	0.06	1	0.85	0.85

Table 5.10: Bandwidth requirements for the years 2005 and 2010 [MHz]

Year	2005	2010
High interactive multimedia	22	82
Multimedia with medium and high data rates	113	241
Packet-switched data	12	9
Simple message services	2	2
Voice	220	220
Overall	339	554
Overall (with guard bands)	406	582

previously used for other purposes set aside for *refarming*, and is aiming to have asymmetrical as well as symmetrical bands.

The need for frequency spectrum in individual countries can vary depending on population density and economic development. The UMTS Forum is initially planning the use of a so-called *core* band. Since UMTS is being interpreted as a third-generation system within the IMT 2000 family, either part or all of the core band is to be available for UMTS/IMT 2000 worldwide. This band is therefore earmarked for mobile applications. The 1900–1980 MHz and 2010–2015 MHz as well as the 2110–2170 MHz bands are being provided for terrestrial applications. The 1980–2010 MHz and 2170–2200 MHz bands are to be used for satellite-supported applications.

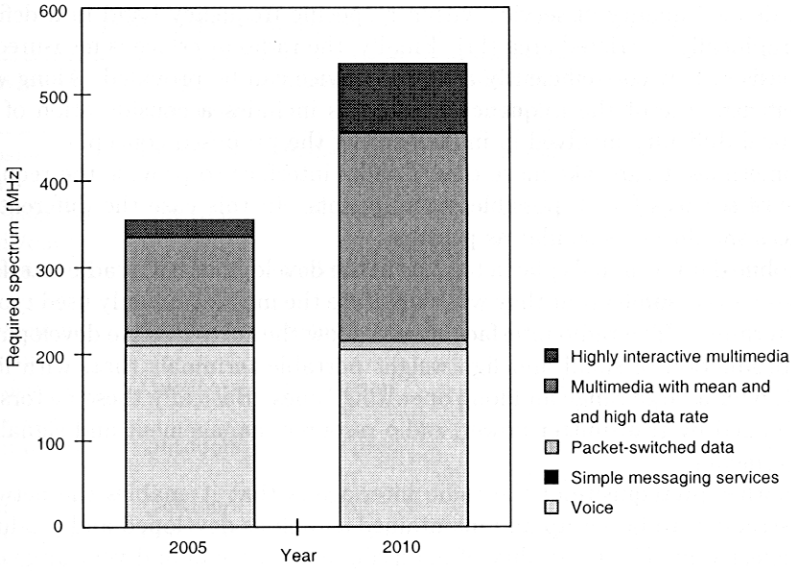


Figure 5.5: Anticipated requirements for frequency spectrum for UMTS services

Two times 30 MHz in the 1900–1980 MHz and 2110–2170 MHz bands are to be made available by the year 2002. This should be at least doubled by the year 2005. Depending on how needs develop, a bandwidth of from 300 to 500 MHz should be available by 2008.

5.5 Demands on the Radio Interface

UMTS is to offer a number of telecommunications services to stationary and mobile users in different communications environments. Access to these services will be through different mobile terminals over public or private networks based on the UMTS standard. Furthermore, international roaming and in particular roaming with FPLMTS, which is standardized by the ITU, should be possible from an UMTS terminal. These goals have determined the requirements of the UMTS radio interface.

A radio interface must be able to provide a variable quality of service in different communictions environments. In this context there will be a strong dependence on the standardization of FPLMTS. Since a large number of telecommunications services are already being offered by existing systems, the UMTS radio interface is to enable the further development of the interfaces of current second-generation systems to meet the requirement of UMTS. Furthermore, UMTS should also be developed further. Utilization of frequency spectrum, i.e., spectral efficiency, is particularly important in the design of a radio interface. Spectral efficiency relates to the traffic that can be carried

for a defined quality of service within a specific frequency band in a defined geographically restricted area [11]. Finally, the radio interface is measured on the basis of how cost-efficiently a bearer service can be provided. Along with the efficient use of the frequency band, this includes a consideration of the technical difficulty involved in implementing the proposed concepts.

Sometimes it can take more than a radio interface to provide the required range of services for all possible environments. In this case the different interfaces should be as similar as possible.

Mobile data terminals play a big role in the development of a radio interface, because it is assumed that they will constitute the most commonly used type of data terminal. The radio interface should allow the cost-effective development and production of small and lightweight portable terminals that, with little effort, offer a large range and long operating times. Basically these factors are determined by transmitter power, radio resource management and signalling requirements.

Another prerequisite of the radio interface is that it enables the network infrastructure to be set up and maintained as well as developed and produced cost-effectively. The possibility of maintaining very small and very large cells should be considered in this context.

The radio interface should also be flexible enough so that different levels of coverage can be realized. Along with wide-area coverage, it should also be possible to supply coverage to regionally restricted areas.

5.5.1 Operating Environment

The radio interface should be flexible enough to support the communications scenarios listed in Table 5.11.

The traffic is given in ETE units (*Equivalent Telephony Erlang*) and describes the traffic normalized to a 1-Erlang voice service. The relative speeds between base and mobile station presented in Table 5.12 are assumed. Three different types of cell are defined: macro cells serve as *umbrella cells* in areas with low traffic density, micro cells carry heavy traffic in outdoor areas, and pico cells are primarily used within buildings.

5.5.2 Services

The radio interface is defined by the bearer services that are to be provided (see Section 5.3). It should ensure that future teleservices and supplementary services and new techniques for improving quality of service can be introduced. In principle, the network capacity should be adequate to provide different levels of quality of service. Therefore any evaluation of proposals for a radio interface must take into account the different multiple-access procedures with their specific transmission delays and interference characteristics as well as the performance of the proposed handover and power control protocols.

Table 5.11: Different communications environments [11]

Environm.	Prop. conditions	Mobility	Traffic density [ETE/km²]	Cell type	Service environm.
Business use	Indoors	Pedestrian	> 1000	pico	office
Residential ind./outd.	Out to in	Pedestrian	< 1000	micro	resid.
Residential	Out to in	Pedestrian	< 10	micro pico	resid.
Vehicle in a city	Outdoors	Vehicle	< 1000	macro	public
Pedestrian in a city	Outdoors	Pedestrian	< 10	macro micro	public
Rural area	Outdoors	High-speed vehicle	< 1 up to < 1000	macro	public
Air	Outdoors	Aircraft	< 1	macro	tbd
Fix. station outdoors	Outdoors	Stationary	< 1000	macro	resid.
High bit rate	Out to in	Pedestrian	> 1000	pico	resid.
City area via satellite	Satellite	Satellite to vehicle	< 10 (< 1000)	satel.	public
Rural area via satellite	Satellite	Satellite to aircraft	< 10	satel.	public
Fix. station via satellite	Satellite	Satellite to fix. station	< 10	satel.	resid.
Within satell. area	Satellite	Satellite to pedestrian	< 10	satel.	public
Vehicle with mobile BS	Indoors	Pedestrian	> 1000	pico	tbd

Table 5.12: Relative speeds [km/h]

Fixed	0	High-speed vehicle	up to 500
Pedestrian	up to 10	Aircraft	up to 1 500
Vehicle	up to 100	Satellite	up to 27 000

5.5.3 Mobility

The mobility of mobile users is determined by the ability of the network to support this function. Mobility should not only extend to one of the operating environments described in Section 5.5.1 but should also be possible between different operating environments. This mobility is to be implemented through handover procedures, which should be unnoticeable to the users. The selection of a handover procedure depends on which service is being supported. For example, the voice service requires speedy handover and allows a cer-

tain amount of data loss, whereas the data service requires a high level of protection against bit errors but accepts a slower handover speed.

5.5.4 Protocols

The protocols of the UMTS radio interface are to fulfil the following requirements [11]:

- Support a broad spectrum of services and operating environments.
- Conform with the ISO/OSI model.
- Develop maximum communality among different radio interfaces across the protocol stack.
- Provide efficient exchange of information between entities.
- Support different multiple-access procedures, in particular CDMA, TDMA, FDMA and hybrid methods.
- Consider new types of radio technologies, such as packet-data transmission, dynamic adaptation of the interface parameters, soft handovers.
- Support service-specific response times through the protocols.
- Carry out signalling over the radio interface.
- Provide parameterizable protocols to allow new characteristics of the radio interface to be implemented through modification of the protocol parameters.

The protocol stack for the radio interface for UMTS should also include protocols of the so-called UMTS adaptation layer, which should be imbedded above the network layer. This adaptation layer should map the teleservices to the UMTS bearer services. Since this functionality depends on the respective bearer service, a separate protocol stack must be available for the adaptation layer for each bearer service.

5.6 Basics of the UMTS Radio Interfaces

UMTS has two different air interfaces, both of which rely on wideband transmission with a bandwidth of 4.4–5 MHz per frequency channel. A characteristic of mobile radio channels is that they are time-selective as well as frequency-selective. A statistical change to the channel parameters causes the received signal to experience fading at certain time intervals, which in turn causes transmission errors, thereby necessitating the use of effective channel coding procedures. Frequency-selectivity causes strong linear distortion of

the receive signal, which must be offset by equalizers. The effects of time-selectivity increase as the bandwidth narrows, whereas those of frequency-selectivity decrease because the bandwidth of a signal is inversely proportional to the symbol duration.

A compromise must therefore be found between coding and equalizing requirements, i.e., between bandwidth and symbol rate, when radio interfaces are designed. Furthermore, appropriate procedures for multiple access and protocols for power control and handover must be developed. The high data rates demanded by the third generation of mobile radio systems require large bandwidths up to 2 Mbit/s. Thus the effectiveness of these proposals is also determined by how efficiently low transmission rates, e.g., for the voice service, are integrated. Services with different rates in different environments should be realized with the same radio interface.

Depending on the duplexing method (FDD or TDD), two different multiple access schemes are defined as shown in Figures 5.9 and 5.12. Both operating modes have a common structure in the frequency and time domain, with a carrier spacing between 4.4 MHz and 5.0 MHz and a frame duration of 10 ms. Each frame is split into 16 slots, each 0.652 ms in length. A super-frame corresponds to 72 consecutive frames. The physical channels are subdivided into dedicated physical (DPCH) and common physical (CPCH) channels.

5.6.1 Wideband CDMA

Code-Division-Multiple-Access (CDMA) is based on a separation of transmission channels through codes. A characteristic of this technique is that the narrowband radio signal is transmitted in a wide frequency spectrum in which the narrowband signal is spread to a wideband signal through one code of a (pseudo) orthogonal codes family.

Each user of the radio communications system is assigned an appropriate spreading code, which is used to spread the signal spectrum being transmitted into a multiple of its original bandwidth. The signals obtained in this way are then sent by the transmitters simultaneously in the same frequency band.

The coding instructions used by the transmitters must be selected in such a way that the interference experienced by the receivers is minimal despite the simultaneous transmission. The use of an orthogonal *Pseudo-noise* code (PN code) for carrier modulation of the information being transmitted meets this requirement; see Section 2.6.4.

The receiver, which must know the spreading rules applied by the transmitter, searches for the wideband signal according to the bit pattern of the PN sequence of the transmitter. By setting up the autocorrelation function (ACF), the receiver is able to synchronize with the coding channel of the transmitter and despread the signal to its original bandwidth. The respective signals of the other transmitters whose codes do not agree with the selected PN sequence are not despread back to the original bandwidth, and there-

fore only contribute to the noise level of the receive signal, i.e., co-channel interference is effective as noise only.

When the number of code channels on the same frequency channel exceeds a certain number, the *signal-to-noise ratio* (SNR) can exceed the value required for reception with the correlator.

An advantage of CDM is the coding, which ensures the confidentiality of user data, thereby cancelling out the need for any cryptographic procedures to protect transmitted data. Systems that use CDM are less prone to interference than FDM or TDM systems.

A further advantage compared with TDM is that special measures for time synchronization of the different transmitters are not required in CDM systems. Because of the code, they are self-synchronizing. A disadvantage of systems using CDM is the need of transmitters and receivers to generate synchronous pseudorandom code sequences. When several stations are transmitting at the same time, random statistical superpositions can occur, leading to errors and requiring appropriate measures for error detection and correction.

5.6.1.1 Direct Sequence Spread Spectrum Transmission

Direct-Sequence (DS) is a spreading technique in which the binary signals being transmitted are added modulo two to the binary output signal of a pseudo-noise generator and then, for example, used in the phase modulation of the carrier signal. Attaching the data bits to the pseudorandom bit sequence (*chip sequence*) transforms the narrowband information signal to the large bandwidth of the PN signal, thereby producing a code channel. The spreading of different orthogonal chip sequences produces orthogonal code channels with the sum signal zero. In practice the orthogonality is only achieved approximately, because the chip sequences are derived from the same pseudo-noise generator through the allocation of different start values.

5.6.1.2 Receiver

In the receiver the user signal has to be regenerated from the noise-similar receive signal through a correlation with the code sequence of the transmitter. According to [21], the optimal maximum-likelihood receiver for a CDMA system with synchronous transmission in K code channels consists of K matched filters followed by a correlator that calculates the 2^K possible information sequences. The one most likely to be transmitted is then selected from these sequences. In the case of asynchronous transmission the correlation is increased to the calculation of 2^{NK} correlation metrics in which N equals the number of information bits per code sequence. In practice this kind of receiver is too complex, and therefore simpler receiving techniques are being used, although they are suboptimal in terms of bit-error ratio.

Two basic techniques available for simplifying the complexity of the receiver are individual detection and joint detection. With individual detection the

receive signal is supplied to an individual decoder in which the non-relevant $K-1$ code sequences and the additive noise have the effect of interference.

Joint detection is oriented towards the optimal receiver; however, correlation and decision are simplified to the extent that the workload is no longer exponential but rather linear with an increase in the number of code channels [6]. Since all the code sequences must be known to the receiver, joint detection is usually only applied in base stations. It reduces the level of neighbouring interference considerably.

An intermediate option between individual and joint detection is provided by the technique of *interference cancellation*, which entails decoding a channel from the received total signal. The transmit signal is then reconstructed from the estimated bit sequence through spreading and subtracted from a version of the received total signal with the same propagation delay. This produces a signal with reduced interference, from which the bit sequence of the next channel is estimated in a further step. This technique is less complicated than joint detection and more efficient than individual detection.

The residual interference is also determined by the number of code channels occupied at the same time. Consequently, depending on the system, the quality of service decreases as the number of active users increases.

The current ETSI UMTS standards are designed to allow the use of single-user detection.

5.6.1.3 Diversity

In principle, diversity can be applied in the following areas (see also Section 2.1.3; especially Figure 2.10):

- in the frequency range through the use of a wide spectrum
- in the time range through the interleaving of time hopping techniques
- through a number of transmitting and receiving antennas
- through the estimation of direction of arrival of the signal
- through the separation of individual multipath components

Because all cells in CDMA systems have a common frequency channel, macro diversity can also be implemented through the multiple supply of mobile stations in co-channel operation. This involves transmitting the traffic and control data of more than one base station on the same channel simultaneously on the downlink. A level of quality comparable to that of a line-of-sight connection can be achieved if the mobile station is in a position of resolving the individual multipath components of the different connections.

RAKE receivers [21] are especially suitable for resolution of multipath components. They have been proposed as the preferred receivers for CDMA-based radio interfaces because of the requirement of wideband transmit signals for a

low coherence time, which necessitates a good resolution of the receiving components. Macro diversity is also used on the uplink to support mobile stations in the boundary areas of a cell by providing the currently best base station or several base stations at the same time, thereby enabling a *soft handover*.

5.6.1.4 Variable Transmission Rates

The radio interfaces of UMTS is able to implement services with transmission rates varying from several kbit/s up to 2 Mbit/s. Variable transmission rates are especially important for the efficient implementation of voice services, which generally have an activity level of less than 0.5.

Using CDMA, variable transmission rates are achieved through different methods. One of these is the variation of spreading level. With a constant chip rate this involves changing the corresponding bit rate by adapting the spreading code (see Figure 5.9) or, if the level of spreading is the same, changing the bandwidth occupied on the channel. Another way of achieving variable transmission rates in CDMA systems is to transmit in code multiplex, i.e., to obtain higher bit rates through the allocation of several code channels to one user. This method has the disadvantage that several parallel receiving components are required, which results in a linear increase in the complexity of the receiver. In the end, different channel coding procedures allow different net bit rates.

The use of the techniques mentioned above can be different for the downlink and for the uplink. A general prerequisite for variable transmission rates is the availability of information in the transmitter and the receiver about current symbol rates, coding and other determining parameters. In the UMTS radio interface the physical channels are divided into equal-length time frames, the transmission rate being maintained constant within a frame (10 ms; see Figures 5.9 and 5.12), thereby allowing it to be changed at regular intervals. The information about current transmission parameters is transmitted within a separate control channel, whose transmission parameters are known a priori. Because a connection is maintained over the control channel even if no data is being transferred over the traffic channel, this principle also allows connection-oriented packet-data transmission.

5.6.1.5 Handover

Because of the cellular structure of mobile radio networks, each cell is allocated a part of the available frequency spectrum. With UMTS, thanks to the use of W-CDMA, the same frequency band is used in all cells with the same cell hierarchy, for example one frequency band in macro cells and a different band in the micro cells below it. In other words, a single cell cluster is used.

The handover of a mobile station from one base station to another one can therefore be carried out without changing frequency. On the assumption that a sufficient number of code sequences are available there is no need for a change in code channel during the transition from one cell to another, a

connection can be set up to a new base station before the old connection has been released, i.e., the mobile station is then maintaining a double connection (see Section 5.6.1.3) for a short period of time. This is called a *soft handover*. There is no need for this process to be limited to two base stations, and it allows a handover to be executed without any interruption to the connection.

If hierarchical cell structures are being supported by a CDMA system through a separation of the individual cell layers by different frequency ranges, a change of frequency must occur when the cell layer changes. To carry out this kind of handover in which there is no interruption to the active connection, the mobile station must be able to receive information about the target cell in another cell layer. This can be effected through the reception of pilot signals from the target cell. Furthermore, the mobile station must be able to maintain quasi two connections at the same time in different frequency bands during handover. This adds to the complexity of the CDMA receiver.

A handover between two cells with different frequencies is referred to as a *hard handover*.

A handover that takes place between two base stations with different frequencies without affecting the quality of service of the connection is called a *seamless handover*.

5.6.1.6 Power Control

With CDMA all signal sequences occurring at the receiver must be at exactly around 1 dB to prevent strong signals from suppressing the weak ones. This effect is referred to as a *near/far problem* and is more pronounced the more removed the receiver is from an optimal structure. The near/far problem necessitates the use of adaptive fast power control. This requirement for speed can be met through *open-loop* power controls. This kind of control is based on the assumption that there is a correlation between the attenuation of the signal on the uplink and on the downlink, and consequently knowledge about the attenuation on the downlink can be used to establish that on the uplink and vice versa. Open-loop power control therefore controls the transmitter power on the basis of knowledge about the receive power. Because it lacks correlation, multipath fading cannot be offset through this kind of control.

Uncorrelated multipath fading can be offset through *closed-loop* power controls. With closed loops a base station, on the basis of the measured receive power, instructs the mobile station to monitor its transmitter power. Conversely, a base station can control its transmitter power using measurements from the mobile station on the downlink.

Because this type of control is always linked to a communication between base and mobile station, it is considerably slower than an open-loop control, but can offset the uncorrelated part of propagation fading. Combinations of open and closed loops are therefore used.

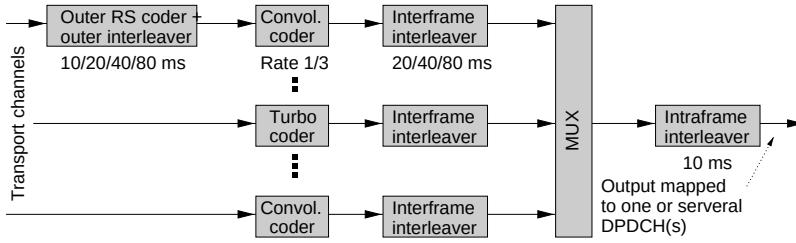


Figure 5.6: UMTS channel coding

5.6.1.7 Channel Coding

Most designs for the radio interface propose block coding, convolutional coding or hybrid versions of the two. Convolutional coding is especially suitable for correcting evenly distributed bit errors, thereby providing a soft decision. Block coding procedures, on the other hand, are mainly used to protect against burst errors. The occurrence of burst errors can be counteracted by interleaving the bits being transmitted. Thus protection against burst errors with minimal block coding is achieved at the expense of increased interleaving and vice versa. However, this statement only applies to a point, since, particularly with hybrid convolutional and block coding techniques, consideration must be given to the overall performance of the coding and interleaving. Variable net bit rates can be achieved through coding ratios and puncturing.

The channel coding for a CDMA system developed in the European RACE-II project CODIT is presented as an example in the following [7]. This system uses different coding procedures for the logical channels for voice, data and the reserved control channel. Table 5.13 presents an overview of the suggested coding.

Every 10 ms the voice codec delivers a frame with a rate between 0.4 kbit/s and 16 kbit/s. The channel coding for this voice data is based on asymmetrical error protection for three different categories of bits according to order of importance. The codes used, which have different coding ratios between 2/5 and 1/2, are based on a rate-compatible punctured convolutional code. The block interleaving includes different interleaving depths and interleaving lengths. The channel coding procedure for the traffic channels of the data services is based on a chain of outer Reed–Solomon codes and an inner convolutional code with the coding ratio 1/2. For the inner and outer coding a separate interleaving is carried out with different parameters for different data rates [7].

Currently the channel coding has been definitively defined. As shown in Figure 5.6, turbo codes and convolutional codes without RS coding are also under consideration.

Table 5.13: Channel coding for CODIT

	Traffic channel voice						
Source bit rate [kbit/s]	0.4	3.2	7.2	8.5	12	12.5	16
Coding conditions							
Category 1	0/0	20/60	16/48	15/45	16/48	15/45	28/70
Category 2	0/0	12/30	40/100	54/135	72/180	78/195	92/184
Category 3	4/40	0/0	16/32	16/32	32/64	32/64	40/60
Category 4 (tail)	0/0	4/10	4/8	4/8	4/8	4/8	4/6
Interleaving[a]	10×4	25×4	47×4	55×4	75×4	78×4	80×4
Format [ms]	10	10	10	10	10	10	10
Transmission rate [kbit/s]	4	10	18.8	22	30	31.2	32
Overall coding ratio	1/10	8/25	2/5	17/44	2/5	2/5	1/2

	Control channel	Traffic channel data		
Source bit rate [kbit/s]	9.6	9.6	64	128
Outer coding				
Code	RS(30,24,8)	RS(60,48,8)	RS(100,80,8)	RS(100,80,8)
Interleaving[a]	—	2×60	6×100	12×100
Format [ms]	—	80	60	60
Inner coding				
Code		Convolutional coder (2,1,7)		
Interleaving[a]	31×16	62×16	606×16	$1\,212 \times 16$
Format [ms]	20	40	60	60
Transmission rate [kbit/s]	24.8	24.8	161.6	323.2
Overall coding ratio	12/31	12/31	40/101	40/101

[a] depth × length

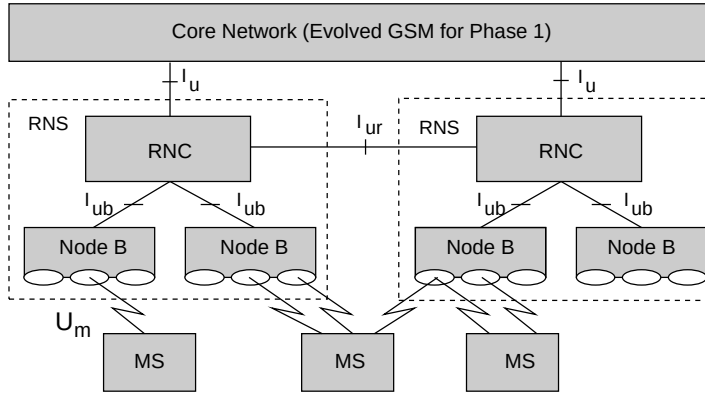


Figure 5.7: UTRAN architecture

5.7 UMTS Terrestrial Radio Access Network Logical Architecture

An UMTS Terrestrial Radio Access Network (*UTRAN*) consists of a set of Radio Network Subsystems (*RNS*) connected to the *Core Network* through the I_u interface. The RNS are interconnected through the interface I_{ur} as shown in Figure 5.7.

A RNS can be either a full network or only the access part of a UMTS network offering the control of specific radio resources to connect a Mobile Station (*MS*) to the UTRAN. The RNS consists of a Radio Network Controller (*RNC*) and one or more abstract *Node B* entities connected to the RNC through the I_{ub} interface. A Node B is in charge of radio transmission/reception in one or more cells to/from a MS connected through the U_m interface. The Radio Network Controller is responsible for the use and the integrity of the radio resources of its cells, e.g., for handover decisions or combining and splitting functions for macro diversity.

5.7.1 Radio Interface Protocol Architecture

In the style of the ISO/OSI Reference Model, the radio interface is layered into the physical (L1), the data link (L2) and the network (L3) layers; see Figure 5.8. The data link layer is split in two sublayers: the Radio Link Control (RLC) and the Medium Access Control (MAC) layer. The network layer is subdivided into Control (C-) and User (U-) planes, whereas the Control plane is further partitioned into the Radio Resource Control (RRC), Mobility Management (MM) and Call Control (CC). MM and CC communicate with the RRC through the General control (GC), the Notification (Nt) and the Dedicated Control (DC) Service Access Points (SAPs).

The RRC provides interlayer services to the MAC and the physical layer.

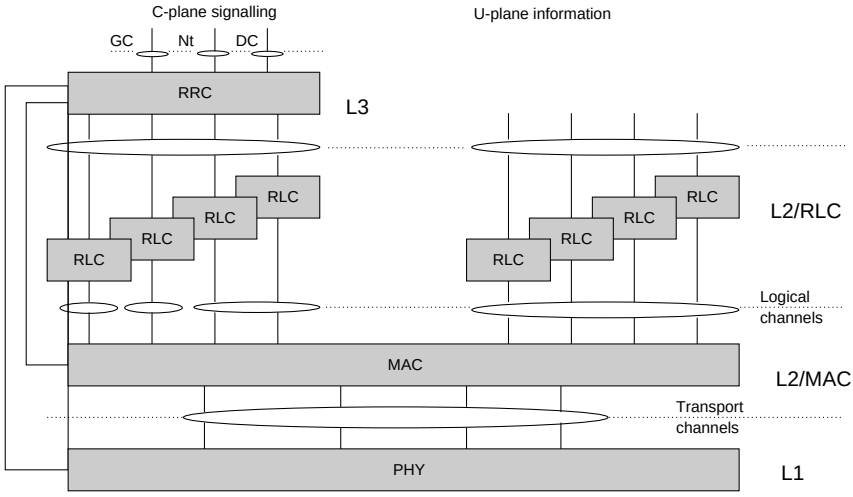


Figure 5.8: Radio interface protocol stack and architecture

5.7.2 FDD Mode

In one version of the UMTS radio interface, called W-CDMA, uplink and downlink are realized on different frequencies spaced by 130 MHz. In FDD mode a physical channel is identified by a code and a frequency. On the uplink, different information may be transmitted on the I- and Q-branch of the QPSK modulation. Therefore in the uplink direction a physical channel is additionally determined by the signal's relative phase. The principle of multiple access in FDD mode is shown in Figure 5.9. Table 5.14 lists the typical parameters for the W-CDMA radio interface of UMTS.

There exist three types of dedicated physical channels—one for the downlink and two for the uplink (see Figure 5.10). Within one downlink DPCH, dedicated data generated at layer 2 and above is transmitted in time-multiplex with control information generated at layer 1. This control information consists of known pilot bits, Power Control commands (PC) and an optional Transport-Format Indicator (TFI) describing the instananeous parameters of the different transport channels multiplexed on the Dedicated Physical Data Channel (DPDCH). The number of bits per downlink DPCH slot depends on the spreading factor of the physical channel, which ranges from 4 up to 256. If the data rate to be transmitted exceeds the maximum bit rate for one DPCH, multicode transmission is employed, i.e., several parallel downlink DPCHs are trasmitted for one connection using the same spreading factor. Then, the control information is put on only the first DPCH. In the uplink, layer-1 and layer-2 informations are transmitted separately in the I- and Q-branch in parallel within the Dedicated Physical Control Channel (DPCCH) and the Dedicated Physical Data Channel (DPDCH) respectively. Multicode

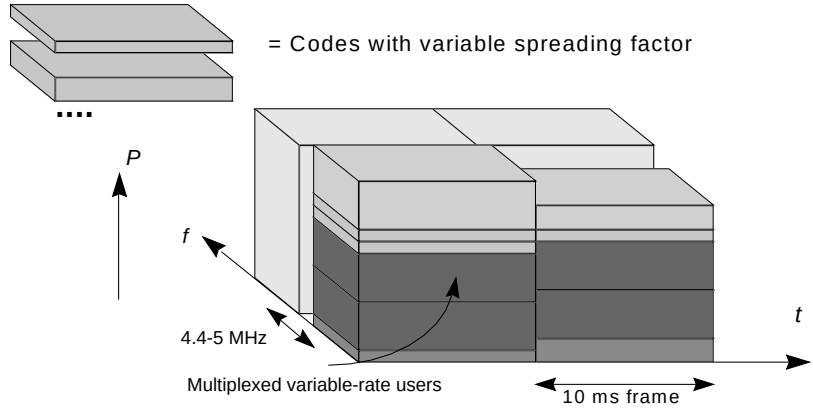


Figure 5.9: Multiple access in FDD mode

Table 5.14: Characteristic parameters of a W-CDMA radio interface

Multiple-access procedure	DS-CDMA
Duplex procedure	FDD
Channel spacing [MHz]	5, 10 and 20
Transmission rate (gross) [MChip/s]	4.096; 8.192; 16.384
Frame duration [ms]	10
Multirate concept	UL and DL: variable spreading level and/or multicode
Channel coding (FEC)	Turbo coding or inner convolutional coding and outer Reed–Solomon code; coding conditions dependent on service
Interleaving	Intraframe, interframe
Spreading factor	4–256
Spreading code	Sequences similar to Gold code, modified Walsh sequences
Modulation	QPSK
Pulse shaping	Root-raised cosine
Detection	UL: coherent through pilot symbol; multi-user detection (joint detection) DL: coherent through pilot channel or symbols; single-user detection with interference cancellation
Diversity	RAKE receiver and antenna diversity (UL)
Power control	Open-loop and closed-loop
Handover	Soft handover
IF handover	Hard handover for hierarchical cell structures
Synchronization of BS	Asynchronous mode
Channel structure	Traffic and control channels Reserved control channels transmitted in same frequency channel as traffic channel

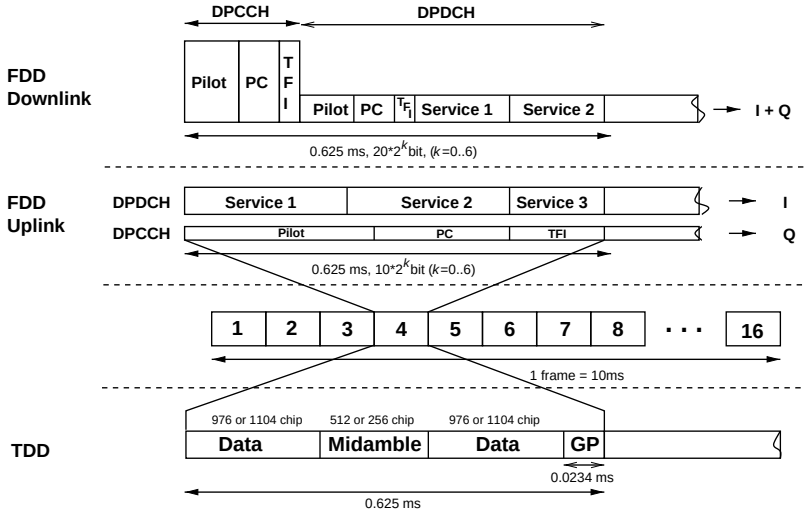


Figure 5.10: Structures of traffic and control channel bursts in FDD and TDD mode

transmission is applied by transmitting several parallel DPDCH with different channelization codes. As in the downlink, there is only one DPCCH per connection.

The FDD mode comprises four common physical channels: the Physical Random Access Channel (PRACH) in the uplink and the Primary Common Control Physical Channel (Primary CCPCH), the Secondary Common Control Physical Channel (Secondary CCPCH) and the Synchronization Channel (SCH) in the downlink. The Primary CCPCH is a fixed rate downlink physical channel used to carry the BCCH (see Section 5.7.4). The Secondary CCPCH is used to carry the FACH and PCH. The PRACH carries the RACH. It is based on a Slotted ALOHA approach where a MS can start the transmission on the PRACH at a number of defined time offsets. These time offsets are evenly spaced 1.25 ms apart. Figure 5.11 shows the structure of a PRACH burst. It consists of a preamble and a message part separated by an 0.25 ms idle period that enables the BS to detect the preamble.

The preamble consists of one of 16 different signature sequences of length 16. Each preamble symbol is spread with a 256-chip Gold code. The message part has the same structure as the uplink DPCH. It encloses the random-access request and an optional user packet for data transmission on the RACH. The spreading factor is limited to 2^k, $k = 5 \ldots 8$. The control part carries pilot bits and rate information using a spreading factor of 256. The rate information indicates which channelization code is used on the data part.

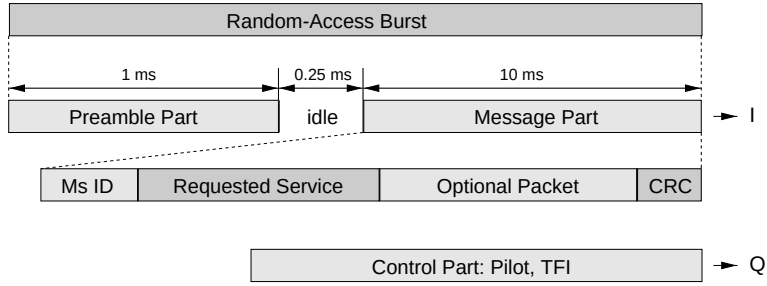


Figure 5.11: Structure of a random-access burst

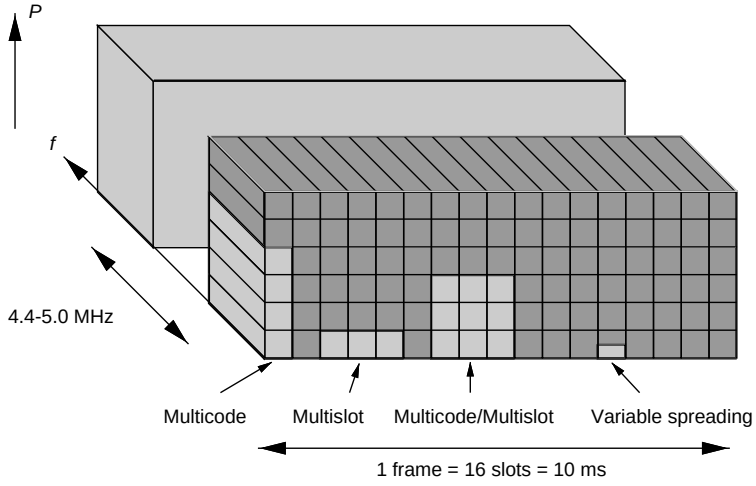


Figure 5.12: Multiple access in TDD mode

5.7.3 TDD Mode

This second variant of the UMTS Terrestrial Radio (UTRA) is based on a combination of the multiple-access techniques TDMA and FDMA. Instead of FDD mode, Time-Division Duplexing (TDD) is used on the same frequency channel. The principle of multiple access on a channel is illustrated in Figure 5.12.

Mobile stations can only access an FDM channel at specific times and only for a specific period of time. If a mobile station is allocated one or more time slots, it can periodically access this set of time slots. The time during which this occurs is called frame duration, i.e., time slots are combined into frames.

User signals within one slot are separated by up to eight different spreading codes. Thus a physical channel is defined by a code, one time slot and one frequency. Each time slot can be allocated to either the uplink or the downlink.

Table 5.15: Characteristic parameters of a TD-CDMA radio interface

Multiple-access procedures	TDMA/CDMA
Resources	Codes/slots
Duplex method	FDD/TDD
Channel spacing [MHz]	4.4–5.0
Transmission rate (gross) [MChip/s]	4.096
TDMA frame structure	8 time slots/frames or dynamic
TDMA frame structure [ms]	4.615
Signal spreading	16 chips/symbol, orthogonal codes
Multiple rate concept	Multislot and multicode
Channel coding	Punctured codes, repetition coding, coding ratio 1/2 and 1/3
Interleaving	Interslot
Modulation technique	QPSK
Pulse shaping	Root-raised cosine
Detection	Coherent detection using Midambel
Power control	Slow, dynamic 50 dB, fast on burst basis (optional)
Handover	Mobile-assisted hard handover
Diversity	Frequency hopping per frame or slot, time hopping
Noise signal reduction	Joint detection
Channel assignment	Fast and slow dynamic channel assignment

For TDD mode, two burst types are defined. Both consist of two data symbol fields, a midamble and a guard period (GP) (see Figure 5.10). The spreading factor ranges between 1 and 16.

Table 5.15 summarizes the key parameters of a wideband TD-CDMA radio interface.

5.7.3.1 Variable Transmission Rates

Users can obtain flexible transmission rates in TDMA systems by occupying several time slots of a frame. This does not require any extra effort on the part of the transceiver hardware. Only when more than one frequency channel is being occupied does the burden on the transceiver increase. However, this is avoided through the wideband transmission.

Variable data rates are realized either by multicode transmission with fixed spreading or single-code transmission with variable spreading. In the former case, multiple spreading codes within the same time slot can be allocated to different users as well as partly or all to a single user. In the latter case the spreading factor of one physical channel can be varied depending on the data rate.

The PRMA++ protocol that controls the occupation of time slots through statistical multiplexing was developed for TDMA systems. This will be discussed further in the next section.

For high transmission rates up to 2 Mbit/s the possibility can be provided for use of a frequency channel with a width that corresponds to a multiple of the FDM width. Furthermore, these high rates, which are typically only available in microcell environments, can also be achieved through multivalued modulation techniques.

5.7.3.2 PRMA++ Protocol

The *Packet-Reservation-Multiple-Access* protocol carries out statistical multiplexing based on TDMA [9, 14, 24]. This protocol originally was called the Reservation-ALOHA protocol [8, 18] that is still the basis of PRMA++. The individual time slots on the uplink are divided into reservation and traffic time slots. On the downlink a distinction is made between acknowledgement, call and traffic time slots. Each time that data is to be transmitted, the mobile station sends a reservation request in a reservation time slot using a Slotted-ALOHA protocol. If access has been successful and if the right resources are free, the base station sends a confirmation and releases a quantity of traffic time slots. If sufficient resources are not available, the reservation request is stored in a queue. The mobile station is notified of this procedure, and then waits until the requested resources are allocated to it.

With this technique a call only occupies resources if data is being transmitted. This is an advantage particularly for the voice service, which only has an activity of less than 0.5. The block-by-block transmission of data also provides each block with channel coding adapted to the current state of the mobile radio channel. Therefore a higher net bit rate can be used to transmit if the level of interference is low, and vice versa.

Priorities can be assigned for different services through the selection of a processing strategy for queues and dynamic channel allocation supported through a central allocation of resources. This statistical multiplexing increases the variance of interference in a system, thereby reducing the mean interference levels by about 1 dB [9]. It is possible to achieve a systematic increase in the variance of interference through the use of frequency hopping techniques.

5.7.3.3 Handover

The handover procedure on TD-CDMA systems is hard handover, which is supported by the mobile stations. This means that a mobile station supplies the base station with information on the transmission qualities of the channel on the downlink. The handover, which involves changing to a different frequency channel, is then initiated by the base station. TD-CDMA systems can therefore support hierarchical cell structures.

5.7.3.4 Power Control

Slow power control has proved to be an advantage for TD-CDMA systems because it reduces co-channel interference. The parameters for this control depend heavily on the fixed frequency channel allocation strategy selected. Fast power control on a burst basis is also being proposed. This kind of power control will only be able to reduce the average transmitter power of stations, and is therefore only being considered as an option.

5.7.4 Transport Channels

Transport channels are the information transfer services offered by layer 1 to MAC and higher layers. There exist two types of transport channels. The common channels provide services where there is a need for inband identification of the MSs when particular MSs are addressed. For dedicated channels the MSs are identified by the physical channel. Common transport channels are:

- Downlink Shared Channel (DSCH, DL)
- Broadcast Channel (BCH, DL)
- Paging Channel (PCH, DL)
- Synchronization Channel (SCH, TDD DL)
- Random-Access Channel (RACH, UL)
- ODMA (Opportunity-Driven Multiple Access) Random-Access Channel (ORACH, TDD UL/DL)
- Forward Access Channel (FACH, UL)

Dedicated transport channels are:

- Dedicated Channel (DCH, UL/DL)
- Fast Uplink Access Channel (FAUSCH, UL)
- ODMA Dedicated Channel (ODCH, TDD UL/DL)

Exept for the FAUSCH, for each transport channel there is an associated transport format or an associated transport format set describing the respective encoding, interleaving, bit rate and mapping onto physical channels.

5.7.5 Agreement Reached on UMTS Radio Interface (UTRA) for Third-generation Mobile System

On the second day of the ETSI SMG 24bis meeting, which was held on 28–29 January 1998 in Paris, France, an agreement was reached by consensus on the radio interface for the third-generation mobile system, UMTS. The solution, called UTRA (*UMTS Terrestrial Radio*), draws on both W-CDMA and TD-CDMA technologies.

This decision follows the inconclusive vote that was conducted on 28 January, where opinions were divided between two principal technologies: W-CDMA and TD-CDMA.

The 316 delegates attending the SMG 24bis meeting, representing telecommunications manufacturers, operators, administrations and research bodies, agreed on the following technical solution:

- In the paired band of UMTS the system adopts the radio access technique formerly proposed by the W-CDMA group, using *Frequency-Division Duplex* (FDD).
- In the unpaired band the UMTS system adopts the radio access technique proposed formerly by the TD-CDMA group, using *Time-Division Duplex* (TDD).

In implementing this solution, ETSI SMG members pursue, together, the specification of UMTS with the objective of providing low-cost terminals, ensuring harmonization with GSM, and providing FDD/TDD dual-mode operation terminals. UTRA should also support operation in a spectrum allocation as small as 2×5 MHz. UTRA should also be designed for broader spectrum allocation to satisfy the capacity requirements and service requirements outlined by the UMTS Forum and the GSM MoU Association.

The parties that made the proposal leading to this new solution included Alcatel, Bosch, Ericsson, Fujitsu, Italtel, Matsushita (Panasonic), Mitsubishi Electric, Motorola, NEC, Nokia, Nortel, Siemens and SONY, as well as Analog Devices, Cegetel, Cellnet, CSEM/Pro Telecom, Deutsche Telekom, France Telecom, Mannesmann Mobilfunk, NTT DoCoMo, Samsung Electronics, SFR, T-Mobil, Telecom Finland, Telia, Texas Instruments, TIM and Vodafone. These companies will work together to provide agreed guidelines for the handling of intellectual property rights essential to the UTRA specification that result in reasonable cost for the manufacturers.

The agreed solution offers a competitive continuation for GSM evolution to UMTS and will position UMTS as a leading member of the IMT 2000 family of systems recommendations being developed in the ITU.

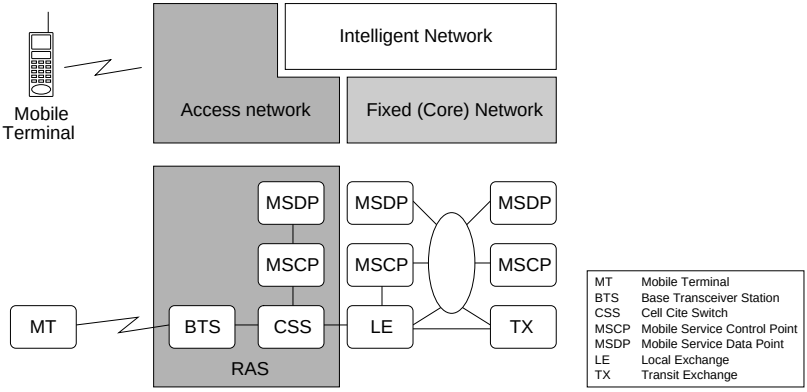


Figure 5.13: UMTS system (generic view)

5.8 Handover in UMTS

Unlike GSM, the UMTS system will be integrated into the fixed network (see Figures 5.1 and 5.13). The fixed network will act as a B-ISDN and consequently use virtual ATM links to transport UMTS information; this will be the basis of a *UMTS Core Network*. The services of mobile radio systems today only offer low data rates. UMTS will provide services with a wider bandwidth of up to 2 Mbit/s [16].

UMTS consists of four system components (see Figure 5.13):

- Fixed core network
- Access network
- Mobile station (MS)
- Functions of intelligent networks

On the user side the mobile terminal (MT) communicates over the UMTS air interface with the base transceiver system (BTS), which is part of the radio access system (RAS). W-CDMA or TD-CDMA are used as the access techniques at the radio interface. The RAS will be able to contain a *cell site switch* (CCS) to transmit information between a BS and the fixed network. The elements of the switching system will use B-ISDN on an ATM basis for transporting user data and signalling information.

The network side is structured similarly to the user side. In addition, the signalling messages must be transported between intelligent networks (IN) consisting of a *Mobile Service Control Point* (MSCP) and a *Mobile Service Data Point* (MSDP), which can be connected to the CSS. An ATM-based signalling network in the core network routes the signalling messages between the participating network elements.

There are no plans in UMTS for the transmission of ATM cells over the air interface, which means that virtual end-to-end connections will not be possible. Instead, the air interface will be designed to be service-specific and

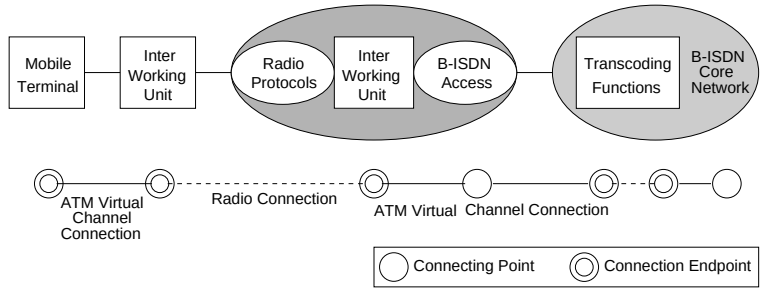


Figure 5.14: Use of virtual ATM connections with UMTS-specific protocols for the radio interface

offer a defined variety of services similar to those of current mobile radio networks. For each service the radio interface will use a specific transmission mechanism in order to guarantee the quality of service (QoS). Integration into B-ISDN requires the provision of interworking functions (IF) in the base transceiver station (BTS/CSS) (see Figure 5.14).

The GSM/UMTS Phase 1 network architecture with its interfaces and including access to the Internet is shown in Figure 5.15. As can be seen, GSM-UMTS MSC/VLR and GSM-UMTS HLR core network elements enable as much use as possible to be made of the GSM switching subsystem components for UMTS. To provide Internet connectivity, a GPRS support node (GSN) is foreseen (see Figure 3.77).

5.8.1 Network-Supported Handover

ATM provides two possibilities for transporting the data of the mobile services. With the first, all the traffic in the *radio access system* (RAS) is transported using ATM transmission technology (see Figure 5.16); with the second, a so-called *UMTS Mobility Server* is integrated into B-ISDN in order to support some of the services [3].

The proposal without the UMTS mobility server only uses standard ATM functions in order to offer basic mobile functionalities. The normal B-ISDN switching must be able to carry out handovers. The following characteristics are necessary for handover support:

Real-time switching This refers to the forwarding of a virtual connection (VC) during the active phase. This process is currently not being supported; however, it should be possible to execute the handover of a VC between two consecutive ATM cells.

Multicasting This is currently being supported by the ATM standards in which the same ATM cell is transmitted to the old and to the new RAS.

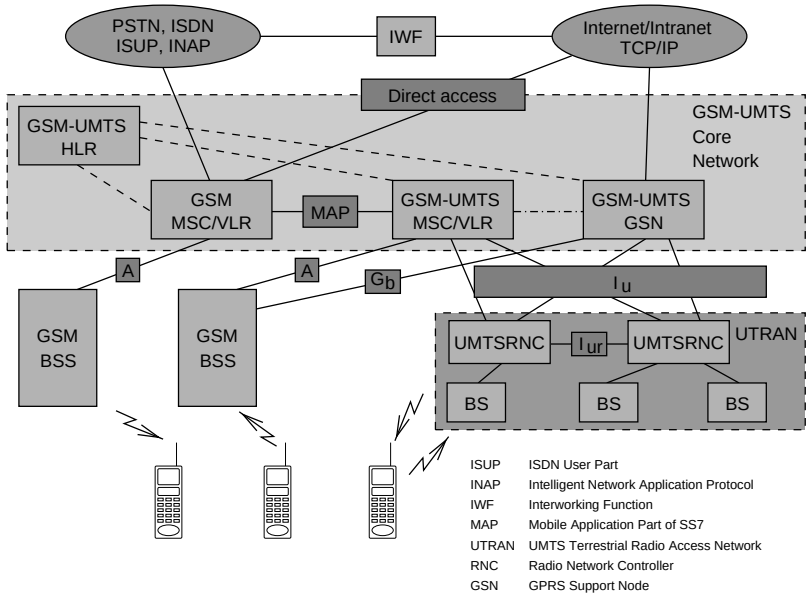


Figure 5.15: GSM/UMTS network architecture

Merging It should be possible to map different incoming VCs to an outgoing VC, which would allow cells to be received from the old and the new RAS at the same time. However, this possibility is also not yet being supported by the ATM standards.

Start/end delimiters These could be used to indicate the first or the last ATM cell of the old or new VC if *multicasting* and *multiplexing* is carried out.

Because merging and start/end delimiters are currently not supported by the ATM standard, handover support is not easy. Moreover, no functionality for the rearrangement of ATM cells or rerouting of VCs is yet being implemented (see Section 12.4).

The UMTS mobility server (see Figure 5.16) would provide precisely those functions that are not easy to implement over an ATM network. The connections between the RAS and the *UMTS mobility servers* will be normal ATM connections. The functions of the mobility server (handover support, macro diversity, transcoding and interworking) will be imbedded above the ATM layer. It will support the following functions:

Handover This will be supported through the setting up of a new VC between the mobility server under the new RAS and the releasing of the old VC. Real-time switching of the VCs in the LE will not be necessary;

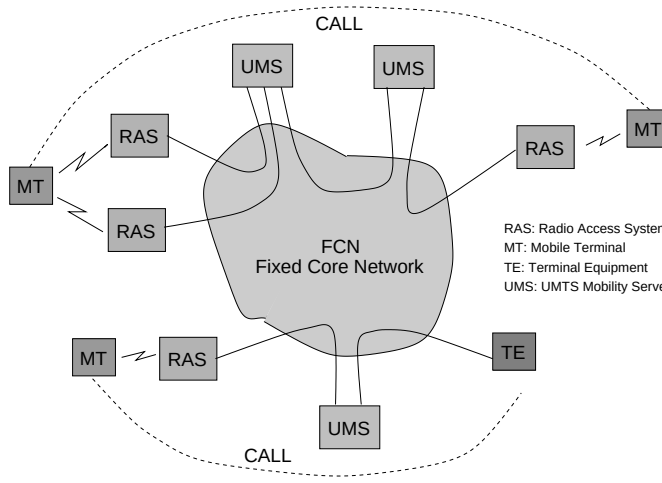


Figure 5.16: Integration of UMTS into B-ISDN with UMTS mobility servers

connections between the mobility server and the RAS can be set up prematurely or disconnected again outside of the active phase.

Macro diversity combining The mobility server is able to combine several *macro diversity streams* on the basis of information about receiving quality. In the simplest case of execution the first correctly received frame is forwarded and all other frames cancelled (*selection combining*). It could also be possible for *Soft Decision Macro Diversity* to be supported.

Macro diversity This also uses multicasting to transmit information to all the base transceiver stations within a macro diversity set.

Transcoding and interworking Interworking between different types of AAL or services of the higher layers (e.g., *transcoding*) could simply be offered in the mobility server.

5.9 Limitations of UMTS

It is clear from the current design of UMTS/UTRA that the data rates are much higher than with second-generation systems, e.g. GSM. The 2 Mbit/s maximum data rate, however, is available only for slow mobile terminals that are close to a base station, e.g., in urban and suburban areas. A data rate of 384 kbit/s can be supported for mobile terminals in urban and suburban areas. Terminals roaming in the country side will get support of data rates of up to 144 kbit/s or below. This would mean that Internet services that temporarily require a high peak bit-rate support cannot be served well countrywide. It

is also clear that W-CDMA/FDD is much better suited to supporting symmetrical than asymmetrical services; Internet services are characterized by a high asymmetry. Because of its limitations to indoor and urban coverage, due to the maximum transmit power allowed, TDD-CDMA is not available to support countrywide asymmetric Internet services, for which it would be well suited. It is not difficult to predict that third-generation mobile radio systems will have to face up to this and present satisfactory solutions. Since the *Internet In the Air* (IIA) is a quite concrete vision of an attractive application for the future use of mobile radio, it might be that the demand to support Internet services much better than is possible with UMTS will soon result in designs of UMTS++ systems.

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6
Trunked Mobile Radio and Packet Data Radio

In addition to the public radio telephone service and the paging service, there are other radio services that are not accessible by the public. These radio systems, called the *non-public land mobile radio network*, have access to frequencies that cannot be used by the public but only by specific users or groups of users.

Probably the best known non-public mobile radio service is analogue *Private Trunked Mobile Radio* (PTMR), which has been used for many years by large firms such as airlines, taxi and transport companies, the railways, and ports, as well as by government departments and organizations responsible for security. What is characteristic of previous PTMR systems is that they have one radio channel that is used exclusively by all the mobile terminals of a specific user group. An analysis of conventional commercial radio systems reveals a number of weaknesses that affect both the customer and the operator:

- Because of too many PTMR users, the fixed allocation of radio channels in congested areas leads to a frequency overload.
- Radio supply areas are too small.
- There is the possibility of eavesdropping by unauthorized persons.
- There is no link to the public telephone networks.
- There is limited support of voice and data transmission.

Frequency overload was the main reason for considering new radio systems and infrastructures. This led to the introduction of *trunked mobile radio systems* as the successors to analogue PTMR.

Although it is not possible for trunked mobile radio systems to expand the frequency spectrum available, they are able to improve the quality of service both for the end user and for the network operator through the optimization of frequency utilization and increased channel use. Advances in trunked mobile radio technology have resulted not only in providing user groups with one channel as in PTMR but also in making a trunk of channels available jointly to a large number of users. A channel is allocated to the user by the system only when required, and then immediately withdrawn after use. Whereas

in PTMR a user would have to wait until a channel allocated to his user group was free, a trunked mobile radio user can start speaking as soon as any one of the channels in the channel group is free. In trunked mobile radio, traffic volume is divided evenly over all the available radio channels, with the trunking of the channels achieving a trunking gain, i.e., the loss probability p_v becomes less and less as the number of channels in a group increases and each channel is constantly utilized (see Appendix A.2). The traffic capacity [in Erl./(MHz · km²)], increases with the *trunk group size*.

In addition to frequency economy, trunked mobile radio systems offer other advantages:

- Low installation cost compared with separate radio control centres.
- Radio supply areas corresponding to the economic areas of activity.
- Higher range.
- No undesirable eavesdropping by others.
- Increased availability because of allocation of channels according to need.
- Optional access to the public telephone network.
- Expanded services because of selective calling, variable group calling and priority calls.
- Improvement in quality of service in voice and data transmission.
- Orderly call queuing operation.

6.1 The MPT 1327 Trunked Mobile Radio System

The pacesetter in standardized trunked mobile radio systems was Great Britain, where the Ministry of Post and Telecommunications developed the trunked mobile radio standard MPT 1327/1343, which is also used in Germany as the technical standard for the first generation of (analogue) trunked mobile radio networks.

Following are some of the services offered in an MPT 1327 trunked mobile radio network:

- A *normal call* can be either an individual or a group call.
- A *priority call* can be either an individual or a group call.
- The mobile telephones called do not respond when they receive a *recorded announcement*.

- A *conventional central station call* in which a radio unit wishing to make a call is not immediately allocated a channel but is required to wait until the central station sets up the call at a convenient time.
- A *conference call* in which additional users can participate in a setup call.
- An *emergency call*, which can be either a voice or a data call placed by an individual or by a group.
- A *data call* can take place between different signalling systems, and is either an individual call or a group call that is transmitted either as a normal or a priority call.
- *Call forwarding or call diversion* to another user or group is possible.
- *Status messages* can be interchanged between different radio units or between radio units and the system, whereby there are 30 different special-purpose messages available.
- *Radio telegrams* are up to 184 bits long and can be interchanged between the radio units or between the radio units and the system.
- A *short telephone call* permits access to a private branch exchange and to the public telephone network.

In a trunked radio network a distinction is made between two different types of radio channel: the *control channel* and the *traffic channel*. All switching-related organizational functions between the system controller and the mobile radio devices are carried out over the control channel through the exchange of data. The main tasks of the control channel include:

- Notification of call requests
- Establishment and termination of calls
- Allocation of communications channels to mobile stations

Trunked radio systems can be operated as local systems with only one base station or as area-wide (cellular) systems with cell sizes from 3 km up to 25 km, e.g., in metropolitan areas with a 6 km diameter of the cells.

The basic structure of a cellular MPT 1327 trunked radio network consists of several cells, each with a radio base station (*transceiver*, TRX), a *trunked system controller* (TSC) and a central node, the *master system controller* (MSC), which also implements the gateway to the public telephone network or to the branch exchange networks (see Figure 6.1).

The TSC controls a radio cell and manages the traffic channels and their allocation to the mobile stations when a call is made. Since roaming is allowed in a multiple-cell trunked radio network, the TSC also maintains a home and

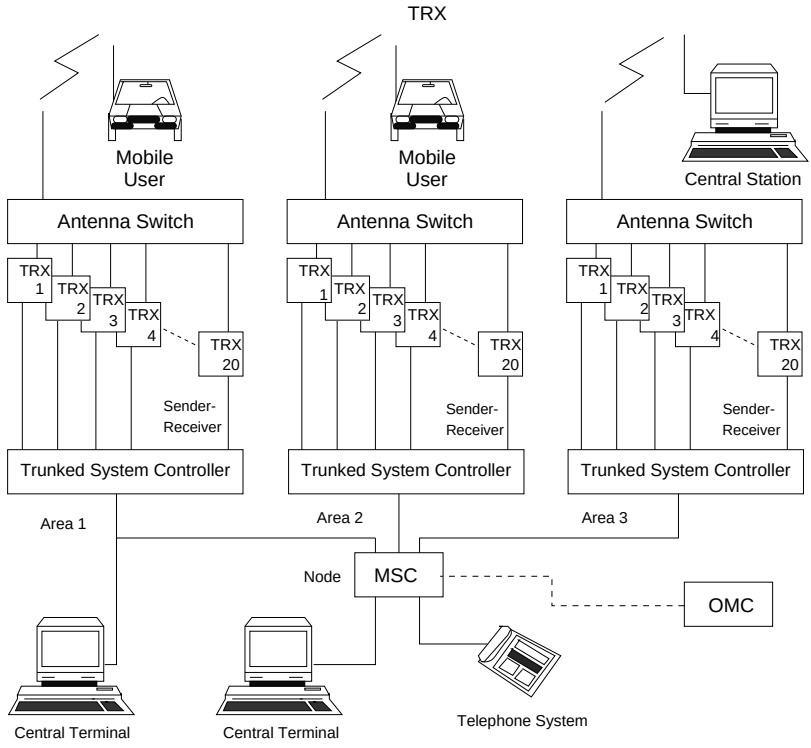


Figure 6.1: Principle structure of a trunked mobile radio network

visitor location register of all the subscribers allocated to the radio cell or who are temporarily operating in the cell. If a call is in progress during a cell change, it is not taken over by the new cell but is broken off; handover is not supported. *Operating and maintenance centres* (OMC), which monitor the system, carry out statistical evaluations and record charges, are coupled to the MSC.

In addition to the MPT 1327 standard described, which defines the signalling protocols between the TSC and the mobile devices, the following standards are also of importance:

- MPT 1343 specifies the operations of the terminal equipment and defines the functions for system control and access to the traffic channel.
- MPT 1347 specifies the functions of the fixed network of the system as well as directives on the allocation of identity numbers.
- MPT 1352 describes the procedures for checking the conformity of the network elements of different manufacturers.

Trunked radio networks can operate in any frequency band suitable for mobile communications. In Europe trunked radio networks operate in the 80–900 MHz range.

One example is the Chekker network operated by Deutsche Telekom AG in Germany in accordance with the MPT 1327 standard in the 410–418 MHz (uplink) and 420–428 MHz (downlink) frequency bands. Up to 20 radio channels, each with a 12.5 kHz bandwidth, are available per cell. One channel can normally service 70–80 users. The maximum transmitter power per base station is 15 W.

Messages are transmitted digitally on the control channel, whereas with the MPT standard user information is transmitted on the traffic channels in analogue. Mobile stations use the control channel in half-duplex mode, whereas the base station transmits on this channel in duplex mode. The necessary signalling data is exchanged on the allocated traffic channel during a user connection.

Phase-shift keying (PSK) modulation has been selected for speech modulation. *Fast-Frequency Shift Keying* modulation (FFSK) is used for data. The transmission rate for signalling data is 1.2 kbit/s; the data transmission rate possible is 2.4 kbit/s.

Systems with a small number of channels can employ a technique allowed by the MPT protocol in which the control channel can be used as a communications channel if the need arises.

Mobile stations in a trunked radio system access the control channel in accordance with a *random access* method, called the S-ALOHA protocol.

In a trunked radio network a call is set up through a series of steps. All checked-in radio units follow the sequence of operations on the control channel in *standby* mode. When a call request is made, indicated by a keystroke on the mobile terminal to the central station, the central station checks the availability of the subscriber terminal being called and informs the respective subscriber, in some cases through a paging signal over the control channel. If the subscriber called answers, a free traffic channel is automatically assigned to the respective parties. The maximum call duration in the Chekker service is 60 s (billing for the service is on a monthly fixed basis). When a call has been completed, the terminals switch back to the control channel. In the event that all the radio channels are occupied, an automatic queuing buffer system ensures that radio channels are allocated on an orderly basis, depending on waiting time or priority.

In Germany the federal postal and telecommunications ministry has provided four trunked radio network licences for commercial communications:

Licence type A Trunked radio networks for regional areas (metropolitan areas), which were stipulated by the licensor before the invitation to tender (e.g., Chekker).

Licence type B Other regional areas proposed by the licensee.

Licence type C Trunked radio networks for local, geographically tightly restricted areas.

Licence type D Countrywide trunked radio network for mobile data radio applications.

Trunked radio networks are divided into two categories based on user type:

- *Public networks*, which are operated by an operating company, whose users include small to medium-sized firms (e.g., towing services, haulage firms, other services).
- *Private networks*, which are operated by large groups, such as port authorities, automobile manufacturers, airline companies and the police.

6.2 MODACOM

Trunked radio networks based on the MPT 1327 standard cannot satisfactorily support data transfer (status messages, radio telegrams and data calls). Mobile radio networks were therefore developed for connecting data terminals with an ITU-T X.25 interface to their data processing systems (*host*). Examples of these proprietary data networks include MOBITEX (Sweden/England), COGNITO (England), ARDIS (USA) and the MODACOM network (e.g., Germany since 1992).

MODACOM is a public mobile radio service that was specifically developed to provide frequent, high-quality and cost-effective data transmission and support access to the public X.25 network. Data transfer in this system is particularly frequency-economic, because it is transmitted digitally and packet-switched, and, compared with other services, is very economical for low-volume transmission. Because of the direct, bi-directional data transfer between data processing systems and mobile data terminals, MODACOM can provide considerable cost savings in operational organization.

The MODACOM network was initially developed for operation outdoors, and, in contrast to GSM, was not planned to be used across borders. It is directed towards customers who would benefit from services being expanded from the wired, packet-switched data network to the mobile area. Applications for the MODACOM service include:

- Database access by mobile terminals over the public X.25 network
- Scheduling applications
- Dispatching services, e.g., for shipping and haulage companies
- Telemetry applications such as emission tests, burglar alarms and parameter requests from vehicles
- Service and maintenance, i.e., remote diagnostics, fault searches or elimination, access to inventory and stock data

6.2.1 Services in the MODACOM Network

After a terminal has been switched on, it searches for a free channel in the designated channel grid and is checked into the system. After it has been checked in, a continuous virtual connection exists over which control signals are exchanged from time to time. These are not transmitted as data packets until actual data is ready to be sent. Network management ensures that a user terminal has continuous send and receive capabilities within the overall MODACOM network as well as access to the following services or performance characteristics:

- Transmission of status messages or file transfer (bi-directional)
- Communication between mobile subscribers themselves
- Intermediate storage of data through the mailbox service
- Connection to other data services/networks
- Closed user group
- Support of group calls
- Automatic acknowledgement of receipt of data sent
- Roaming
- Secure data transfer
- Password query, personal identification and authorization

6.2.2 The MODACOM Network Structure

The MODACOM system has a cellular structure in which each cell is served by a base station (BS). The cell radius in urban areas is 8 km and in rural areas it is 15 km. Therefore the radio coverage area in a radio data network (RDN) consists of at least one BS that is connected to the *area communications controller* (ACC) over a direct data link at 9.6 kbit/s (see Figure 6.2).

The ACC is a switching computer that controls and coordinates the base stations attached to it. One or more radio areas and the ACC responsible for the areas together form a domain. MODACOM terminals are allocated to the ACC with the coverage area in which they are most frequently likely to be active (home domain). Mobile terminals (MT) also operate outside their home domain and are allowed to move across domain boundaries, in which case they are handed over from ACC to ACC (*handover*). Communication between ACCs takes place—unnoticed by the user—over the public X.25 network. The transition from the radio data network (RDN) to the X.25 data network materializes over one or more nodes (ACC|G, G = gateway). This transition is implemented indirectly over dialled-up *switched virtual circuits* (SVCs) or *permanent virtual circuits* (PVCs). A *network administration host* (NAH) is responsible for the configuration and the monitoring of the radio network.

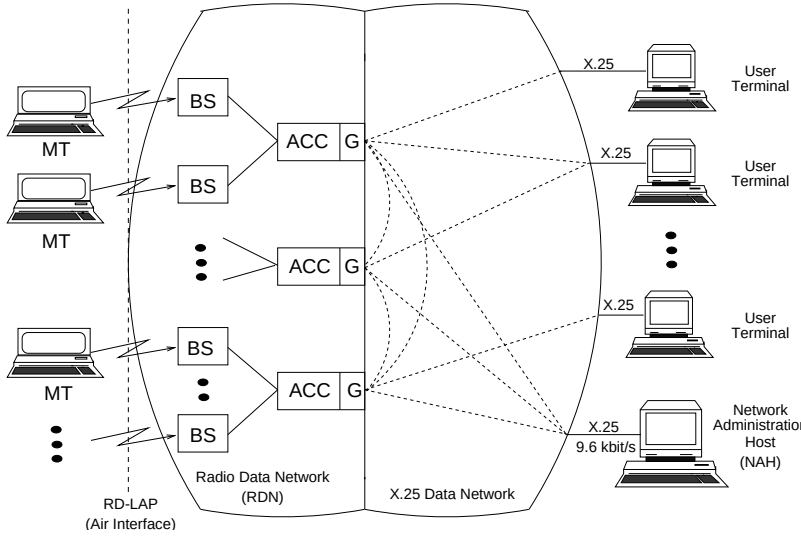


Figure 6.2: The architecture of the MODACOM system

6.2.3 Technical Data

The MODACOM system was allocated in Germany the frequency ranges 417–427 MHz (uplink) and 427–437 MHz (downlink), with a duplex separation of 12.5 kHz. These frequency bands are divided into 12.5 kHz bandwidth channels. The 4-FSK technique is used for modulation. The transmitter power is 6 W for the base station and a maximum of 6 W for the mobile terminal.

Data packets are transmitted over the air interface in accordance with the radio protocol RD-LAP (*Radio Data Link Access Procedure*), which alternatively allows connection-oriented or connectionless communication for synchronous dialled calls in half-duplex mode between MT and host. At the radio interface the RD-LAP protocol is based on the *Slotted Digital Sense Multiple Access* (DSMA) channel access method, and incorporates the following characteristics:

- Maximum packet length is 512 bytes, with shorter packets also allowed.
- If a channel is occupied, the packets to be sent are deferred and the timing of the next transmission attempt is determined at random (non-persistent behaviour).
- If a channel is free, transmission is with the probability p (p-persistent behaviour)
- Connection-oriented transmission.

Table 6.1: Technical parameters of MODACOM packet radio network

Frequency ranges	417–427 MHZ and 427–437 MHz
Duplex separation	10 MHz
Channel grid	12.5 kHz
Modulation	4-FSK
Radiation power	6 W ERP
Air interface	Open standard RD-LAP
Data transfer	Digital, packet-oriented
Bit rate	9.6 kbit/s net
Message length	Max. 2048 bytes
Packet size	Max. 512 bytes
Block size	12 bytes
Response time	Approx. 1.5 s
Channels/carriers	One data channel per carrier
Channel access	Slotted DSMA
Forward error correction	Trellis coding with interleaving
Error detection code	CRC-check sum
Bit-error ratio	Better than 10^{-6}, typically 10^{-8}

- Reservation is not possible.

The RD-LAP protocol includes error detection and correction as well as procedures for message segmentation and reassemling after receipt. Layer 2 is able to process messages of a maximum length of 2048 bytes segmented into four data packets each 512 bytes in length and transmitted at 9.6 kbit/s. The data packets are automatically acknowledged by the network, and in the case of error a transmission attempt is repeated three times. CRC check sum procedures (*cyclic redundancy check*) are used for error detection. Data is transmitted with forward error correction procedures using trellis coded modulation and interleaving with a bit error ratio less than 10^{-6}. The technical parameters of the MODACOM system are summarized in Table 6.1 [15].

6.2.4 Different Connection Possibilities in the MODACOM Radio Data Network

6.2.4.1 Connection of Two Mobile Terminals

This type of connection (*messaging*) allows the exchange of free message texts with manual or automatic acknowledgement between two mobile terminals. Simplified message routing to a third terminal is possible.

A terminal is dialled through the manual input of a terminal address, which for simplification can be associated with an alias table of names or abbreviations. The messages are provided with an appropriate header. The system accepts the message, converts it per the sender's request and transmits it to

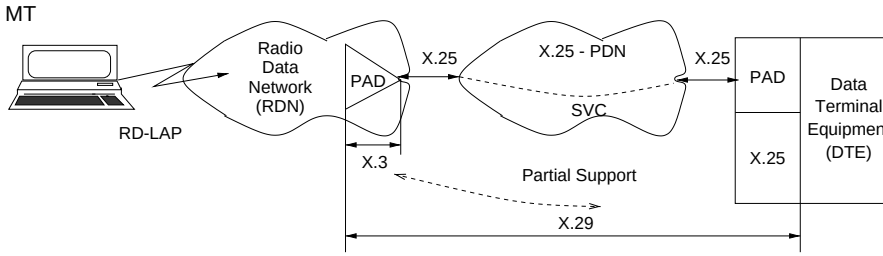


Figure 6.3: Outgoing individual call

the addressee terminal. If the terminal is not available, the message is then temporarily stored in the MODACOM box.

6.2.4.2 Connection to Fixed Network

Connections between MT and the wireline fixed network are exclusively carried out over the public X.25 network. The MODACOM system supports three types of connection betwen host and MT [17]:

Type 1: Outgoing individual call These calls can be used to interrogate public databases, etc. An outgoing individual call is always initiated by an MT and used for interactive communication with the dialled host. The connection between MODACOM system and host is set up as a *switched virtual channel* (SVC) in the X.25 network. Figure 6.3 illustrates a Type 1 connection over a PAD (*packet assembly disassembly*).

In order to set up the connection, the MT sends a special data packet that contains the ITU-T Rec. X.121 address of the target host. The connection between MT and X.25 network is implemented through X.3-PAD functions and the SVC connection in the wireline network. The PAD links asynchronous (start/stop) terminals to an X.25 host.

The MODACOM system emulates only a subset of the X.3 and X.29 interfaces, whereas the X.28 specifications, which describe the configuration of a PAD by the asynchronous terminal, are not required. A special data packet is sent by the MT in order to terminate a connection. A connection can also be terminated by the host using the normal resources of the X.29 specification.

Type 2: Incoming individual call These connections are based on exclusive switched virtual channels (SVC) that can only be set up by the host. Figure 6.4 illustrates an example of an incoming individual call.

Since the public X.25 network is only able to connect data terminal equipment (DTE) with an X.25 interface, the MODACOM network RDN or its gateway node G is connected to the network as an X.25 subscriber. Every incoming call requires a connection and termination

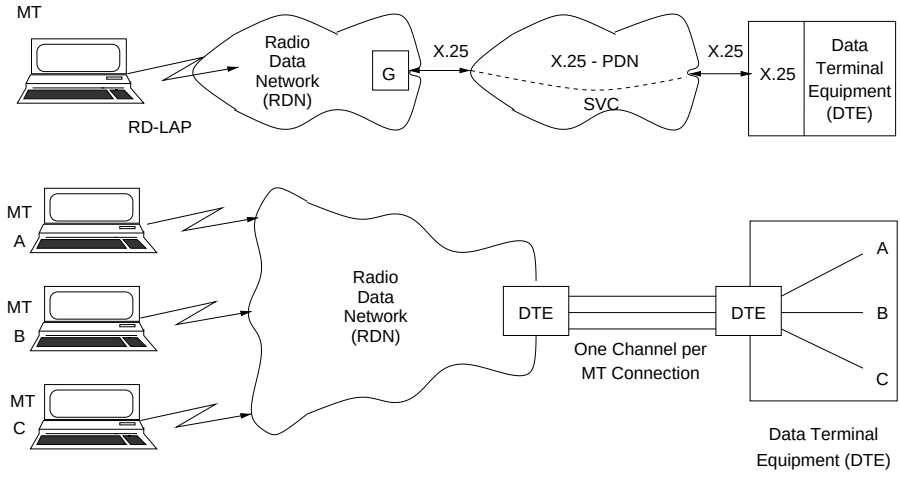


Figure 6.4: Incoming individual call

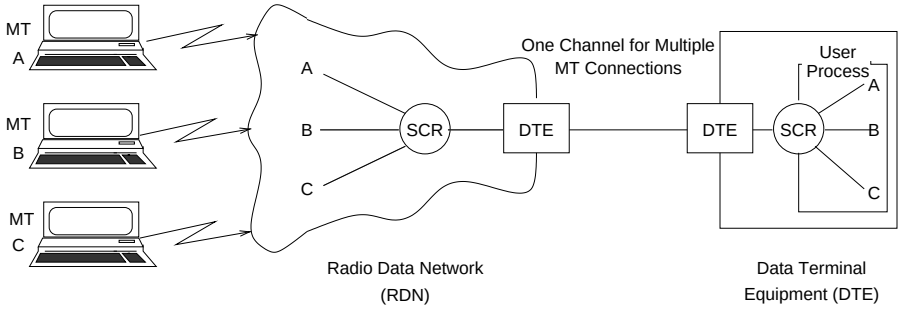


Figure 6.5: Pooling

procedure. When a connection is established, the MT is allocated an SVC connection, which must be managed exclusively by the MT if it makes further connections. The number of MTs that can be served by a host is equal to the number of SVCs that are available between the host and the MODACOM system. The combination of the SVC and logical connection to the MT is stored in the MODACOM system to enable incoming data to be routed to the responsible MT on the basis of the FIFO method.

Type 3: Pooling This is the standard connection between a host and the MODACOM system in which a large number of (e.g., >100) MTs are connected to a host through an SVC or a PVC (see Figure 6.5).

Standard context routing (SCR) is used in pooling calls in order to distinguish between the data packets of the different MTs in a time-division

multiplexed virtual connection. An SCR header that contains the logical destination address of the MT or the host and other applications-related parameters is added to each X.25 data packet. To identify an MT, the host must decode the SCR header that has been received and add it before each packet when it is transmitting messages, to ensure that each message is received by the right MT. In a switched connection (SVC) between the MODACOM system and the host, the connection must first be set up by the host. A PVC (*permanent virtual circuit*) connection that requires no time for setting up or terminating a connection is suitable for users with high traffic volumes.

6.2.4.3 Group Calls

Along with the individual call, group calls are another feature of pooling connections. Each MT can respond to up to seven group addresses; thus up to seven different groups can be formed within a pool of MTs.

A group call is only transmitted once, and there is no acknowledgement of receipt by the terminal or storage in the MODACOM box. Application software is available that enables a serial group call to be initiated in which the same message is sent to MTs in succession. With this procedure each call is acknowledged or temporarily stored in the MODACOM box.

6.2.5 Roaming and Handover

In the MODACOM system MTs are not restricted to a particular area and, through constant accessibility by the host, can move about freely in the radio coverage area. Logical connections are supported by a handover procedure when there is a changeover from one radio cell to another.

If the terminal establishes that the field strength of the selected radio channel is too low or the bit-error ratio is too high, it initiates a roaming process and assigns itself to a new base station. The MT thereby searches for a new radio channel, evaluates the quality and the availability of the radio channel on the basis of status messages that are regularly sent by the base station, and, if the channel is considered satisfactory, transmits a registration packet to register itself with the relevant base station. Two types of roaming are differentiated in the MODACOM system:

Roaming within the home ACC area in which messages coming from the host are merely rerouted to the other base station.

Roaming between two ACC areas occurs when an MT leaves its ACC area and registers in another ACC area. The visited ACC checks the authorization of the MT with the home ACC and, if the MT is accepted, exchanges all the data required for the operation of the MT with the home ACC. The MT is then registered in the visitor register of the new ACC. All data relating to the MT is then forwarded from the home ACC

to the visited ACC. The visited ACC in turn reroutes all messages from the MT to the home ACC.

6.3 Second-Generation Trunked Mobile Radio Systems: The TETRA Standard*

Despite the introduction of GSM throughout Europe, it is expected that the subscriber numbers for trunked mobile radio systems will continue to grow steadily, possibly reaching around five million by the year 2000.

None of the first-generation trunked radio systems currently on the market offers satisfactory voice and data services or is technically capable of dealing with the anticipated number of subscribers. In an effort to harmonize the trunked radio market in Europe, and taking all these factors into account, ETSI decided in 1988 to produce a standard for a digitial, Pan-European trunked radio network. The first working title for this system, which was developed by the Technical Subcommittee RES 06, was MDTRS (*Mobile Digital Trunked Radio System*). However, in late 1991 the new name TETRA (*Terrestrial Trunked Radio*) was introduced for MDTRS.

Two families of standards have been produced for TETRA (see Table 6.2):

- Voice plus Data Standard (V+D)
- Data only (Packet Data Optimized Standard, PDO)

*With the collaboration of Martin Steppler

Table 6.2: The series of the TETRA standard

	Series	Content
V+D	01	General network description
and	02	Definition and description of air interface
PDO	03	Definition of interworking function
	04	Description of air interface protocols
	05	Description of user interface
	06	Description of fixed network stations
	07	Security aspects
	08	Description of management services
	09	Description of performance characteristics
V+D	10	Supplementary services—Level 1
	11	Supplementary services—Level 2
	12	Supplementary services—Level 3

The TETRA V+D standard is envisaged as the successor to existing first-generation trunked radio networks, whereas the PDO standard defines a second-generation packet radio system. Both standards use the same physical transmission technology and largely the same transmit/receive equipment.

European-wide standardization is forcing the issue of interoperability, i.e., manufacturer-independency of terminal equipment in the TETRA network, as well as interworking between different TETRA networks and the fixed networks. Local and regional voice and radio data applications are being replaced by a European trunked radio system that covers all voice and data services and satifies current requirements for bit rates and transmission delay. The main application areas for TETRA are fleet management, telemetry, service companies and for communication within government departments and organizations responsible for security.

Network operators, legislators, manufacturers and users were included in the standardization process to ensure that the ETSI-TETRA standard would have the chance of widespread implementation in the European market. The first TETRA products were available in late 1996. In 1997 the system was able to offer individual and group calls and data and other services described in detail in Section 6.3.2.

6.3.1 Technical Data on the TETRA Trunked Radio System

The trunked radio system TETRA can be used as a local or a multicell network. Since the transmitter power of the terminal equipment is 1 W, 3 W or 10 W, the maximum cell radius in rural areas is limited to 25 km. Several frequencies in the ranges between 380 MHz and 470 MHz and 870 MHz up to 933 MHz have been allocated to the frequency bands for the uplink and the downlink (see Table 6.3). The possibility of using the 1.8 GHz band is being examined.

The TETRA system uses $\pi/4$-DQPSK modulation and offers a gross bit rate of 36 kbit/s in a single 25 kHz channel. With an average quality of service guaranteed by the channel coding, the net bit rate is at 19.2 kbit/s. Without channel coding it is possible to achieve a maximum net bit rate of 28.8 kbit/s (see Table 6.4). With V+D four TDMA voice or data channels are available per carrier; with PDO there is no circuit-switched communication and a statistical multiplexing of packets is employed instead.

Slotted-ALOHA (with reservation in data transmission) is used as the access procedure with V+D. In TETRA PDO a choice can be made between the access methods Slotted-ALOHA with reservation and *Data Sense Multiple Access* (DSMA), depending on traffic load. With V+D the frame structure consists of four 510-bit time slots per frame, 18 frames per multiframe and 60 multiframes per hyperframe, the latter representing the largest time unit and taking approximately one minute (see Section 6.3.4.2).

With the PDO protocol the length of an information block is 124 bits protected by convolutional coding with a code ratio 2/3 and transmitted con-

Table 6.3: Technical data on TETRA in Europe

Frequencies	Uplink: 380–390 MHz, Downlink: 390–400 MHz Uplink: 410–420 MHz, Downlink: 420–430 MHz Uplink: 450–460 MHz, Downlink: 460–470 MHz Uplink: 870–888 MHz, Downlink: 915–933 MHz
Channel grid	25 kHz
Modulation	$\pi/4$-DQPSK
Bit rate	36 kbit/s gross; 19.2 kbit/s net (in 25 kHz channel)
Channels/carriers	V+D: 4 TDMA voice or data channels in 25 kHz PDO: Statistical multiplexing of packets
Access methods	V+D: TDMA with S-ALOHA on the random access channel (reservation offered with packet data) PDO: S-ALOHA with reservation or DSMA, depending on traffic load
Frame structure	V+D: 14.17 ms/slot; 4 slot/frame; 18 frame/multiframe; 60 multiframe/hyperframe; slot length: 510 bit PDO: Uplink and downlink use 124-bit length blocks that are protected by FEC with code rate 2/3; continuous transmission on downlink, discontinuous transmission on uplink
Neighbouring channel protection	−60 dBc
Connection setup	<300 ms circuit-switched; <2 s connection-oriented
Transmission delay of a 100 byte reference packet	V+D: <500 ms in a connection-oriented service, <3–10 s in a connectionless service, depending on transmission priority PDO: <100 ms with a 128-byte message

tinuously on the downlink, discontinuously on the uplink. An exact profile of the channel coding for V+D is given in Section 6.3.4.3.

The time required to establish a connection should not exceed 300 ms for a circuit-switched call or 2 s for a connection-oriented transmission of packet data (*Connection Oriented Network Service*, CONS). The transmission delay of a 100-byte reference packet in connection-oriented transmission with V+D should be a maximum of 500 ms; with connectionless transmission, depending on the respective transaction priority, it should be a maximum of 3 s, 5 s or 10 s. With PDO a transit delay of a maximum of 100 ms has been stipulated as the upper limit for connection-oriented services for a 128-byte reference packet.

6.3.2 Services of the TETRA Trunked Radio System

The TETRA system provides packet data services, which are offered by the PDO and V+D standards, and circuit-switched data and voice services, which are only available with the V+D standard. The packet-oriented services differentiate between the following types of connections:

- Connection-oriented packet data transmission in accordance with ISO 8208 *Connection-Oriented Network Service* (CONS) and services based on ITU-T recommendation X.25.
- Connectionless packet data transmission in accordance with ISO 8473 *Connectionless Network Service* (CLNS) for acknowledged point-to-point services and/or TETRA-specific acknowledged point-to-point and non-acknowledged point-to-multipoint services.

Circuit-switched voice can be transmitted unprotected over bearer services or (preferably) protected over teleservices (see Table 6.4). The teleservices for voice transmission offer five different types of connection:

Individual call Point-to-point connection between calling and called subscribers.

Group call Point-to-multipoint connection between calling subscriber and a group called through a common group number. The call is set up quickly because no confirmation is required. The communication takes place in half-duplex mode through the activation of a *push-to-talk* switch.

Direct call (Direct mode, DM) Point-to-point connection between two mobile devices with no use of the infrastructure. A mobile station establishes a connection with another mobile station without the services of a base station, maintains the connection and takes over all the functions needed for local communication normally handled by the base station. Frequency ranges not normally used in the network are used for this purpose. At least one of the stations must have a connection to a base station on another channel [1]. For example, a connection can be established between two subscribers one of which is not operating in the supply area of the base station.

Acknowledged group call Point-to-multipoint connection between a calling subscriber and a group called through a shared group number in which the presence of the group members is confirmed to the calling subscriber through an acknowledgement. If one of the group members is not present or is on another call, the TETRA infrastructure informs the calling subscriber. If not enough members in the group can be reached, the caller can decide whether to discontinue or to maintain the call. An option is to have the group members who initially were not available to switch into the conversation later.

Broadcast call Point-to-multipoint connection in which the subscriber group dialled through a broadcast number can only hear the calling subscriber.

The bearer services and teleservices provided in the standard for the protocol stack V+D and PDO are listed in Table 6.4.

The TETRA system supports the following data and text services:

Table 6.4: Bearer services and teleservices for V+D and PDO in the TETRA standard

TETRA		V+D	PDO
Bearer services	7.2–28.8 kbit/s circuit-switched, unprotected speech or data	×	–
	4.8–19.2 kbit/s circuit-switched, minimally protected data	×	–
	2.4–9.6 kbit/s circuit-switched, highly protected data	×	–
	Connection-oriented packet transmission (point-to-point)	×	×
	Connectionless packet transmission in standard format (point-to-point)	×	×
	Connectionless packet transmission in special format (point-to-point, multipoint, broadcast)	×	×
Teleservices	4.8 kbit/s speech	×	–
	Coded speech	×	–

- Group calls
- Status messages
- Data messages
- Emergency call messages
- Electronic mail (e-mail)
- Facsimile and videotex

In addition, different supplementary services are offered, e.g.:

- Indirect access to PSTN, ISDN and PBX over a *gateway*.
- *List search calls* (LSC) in which subscribers or groups are called in the order of sequence of the entries in a list.
- *Include calls* in which by dialling the respective number an additional subscriber can be included in an existing telephone call.
- Call forwarding and call diversion.
- Call barring of incoming or outgoing calls (BIC/BOC).
- *Call authorized by dispatcher* (CAD), in which a request is made to place a particular kind of call.
- *Call report* (CR), allowing the telephone number of a calling subscriber to be recorded so that the called subscriber can return the call later.
- Calling number/connected line identification (CLIP, COLP). This function can be prevented using *calling/connected line identification restriction* (CLIR). *Talking party identification* is also possible.

- *Call waiting* (CW), indicating to the subscriber who has called while the talking party was on another call.
- *Call holding, connect-to-waiting*, allowing a subscriber to interrupt his current call to take another call and then to continue with the original call.
- *Short-number addressing* (SNA), allowing a user to make a call using an abbreviated calling number. The TETRA infrastructure converts the abbreviated number to the subscriber number.
- Priority calls.
- Priority calls with interruption.
- Access priority.
- *Advice of charge* (AoC) is a service that indicates the cost of a call to a subscriber either during or after a call.
- Discrete eavesdropping of a conversation by an authorized person.
- *Ambience listening* (AL), allowing the user of a mobile device to be restricted to emergency calls only.
- Dynamic group number allocation.
- *Transfer of control* (TC), permitting the initiator of a multipoint connection to transfer control of a call to another person involved in the call.
- *Area selection* (AS), allowing an authorized user to select the cell for setting up a call or a subscriber currently being served to determine the cell.
- *Late entry* (LE) is an invitation to potential subscribers in a multipoint connection to be switched into an existing call.

Series 10, 11 and 12 of the V+D standard contain a complete compilation of the supplementary services, along with detailed definitions and descriptions [10, 11, 12].

6.3.3 Architecture of the TETRA Standard

6.3.3.1 Functional Structure of the TETRA System

With few differences, the TETRA system is structured like the GSM one (see Figure 6.6). It has the following three subsystems:

- Mobile station
- Line station
- Switching and management infrastructure

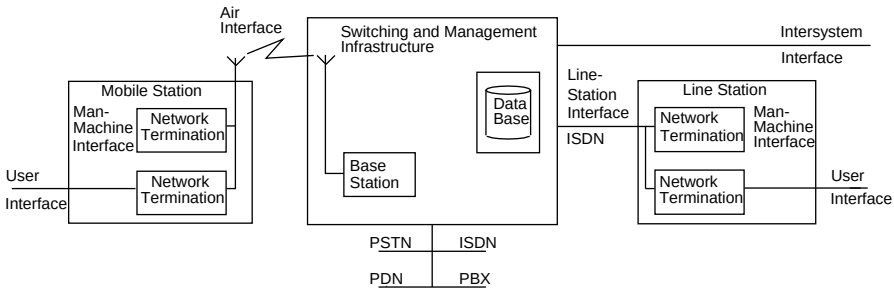


Figure 6.6: The architecture of the TETRA system

Mobile station (MS) The mobile station (MS) comprises all the subscriber's physical equipment: the radio telephone and the interface that the subscriber uses to access the services.

As in GSM, a mobile station consists of two parts: the telephone device, which contains all the necessary hardware and software for the radio interface, and a *Subscriber Identity Module* (SIM), which contains all the subscriber-related information. The SIM can either be in the form of a smart card the size of a cheque card or permanently mounted in the device. The first version has the advantage that it allows a quick change in ownership of the mobile station. The third option is to key in a *login code* to convey the subscriber-specific information. In this case too the mobile unit is not restricted to one particular user.

In addition to the subscriber's identification, each mobile device has a *TETRA equipment identity* (`TEI`) that is specific to the device. This number is input by the operator; only the operator is able to bar the use of the device or release it for use. This means that a stolen device can be disabled immediately and unauthorized access is virtually impossible.

The following numbers and identities are allocated to ensure that a mobile station can be uniquely addressed and managed:

- TETRA Subscriber Identity (`TSI`)
- TETRA Management Identity (`TMI`)
- Network Layer SAP Addresses (`NSAP`)
- Short Subscriber Identity (`SSI`)
- Mobile Network Identity (`MNI`)

The `TSI` consists of three parts: a *Mobile Country Code* (`MCC`) that contains the country identification, a *Mobile Network Code* that identifies the respective TETRA network, and a *Short Subscriber Identity* (`SSI`) that identifies the subscriber. When a connection is to be established within the home network,

only the SSI is used as the address. This reduces the volume of signalling data.

The TMI is used for the management functions of the network layer. The NSAP is employed for the addressing of external, i.e., non-TETRA networks, and is optional. For instance, it can be used to establish a connection to ISDN.

Similarly to GSM, mobile telephones can be installed in vehicles or used as portable/hand-portable devices. All the standard services listed in Section 6.3.2 can be accessed by a mobile station. Supplementary services are offered by the network operator or must be contracted at the same time as the mobile phone contract so that the user can access them.

Line station (LS) In principle a line station is structured in the same way as the mobile station, but with the switching and management infrastructure connected over ISDN. For example, the owner of a company that operates a fleet of lorries would use a line station as the central station for his network. A line station offers the same functions and services as a mobile station.

Switching and management infrastructure (SwMI) The switching and management infrastructure (SwMI) forms the local control unit of the TETRA system. It contains base stations that establish and maintain communication between mobile stations and line stations over ISDN. It carries out the required control tasks, allocates channels and switches calls. It carries out authentication checks and supports the relevant databases such as the *Home Data Base* (HDB), which contains the telephone numbers, the equipment numbers and the subscribed basic and supplementary services for each individual subscriber in the home network, as well as the *Visited Database* (VDB), which contains information on visitors to the network copied from their HDB. It also handles call charging.

6.3.3.2 Interfaces of the TETRA System

Subscriber interface of the mobile station A TETRA mobile station is called a *mobile termination* (MT). Its functions cover radio channel resource and mobility management, speech and data coding/decoding, transmission security as well as data flow control. The following versions are used:

- MT0 (*Mobil Termination Type 0*) contains the named functions with the support of non-standardized terminal interfaces that contain the terminal equipment functions (see Figure 6.7).
- MT2 (*Mobil Termination Type 2*) also supports the named functions, and has an R_T interface for terminal equipment based on the TETRA standard (see Figure 6.7).

The terminal equipment (TE2) is directly accessible by the subscriber, and corresponds to comparable function groups of the GSM and ISDN concept.

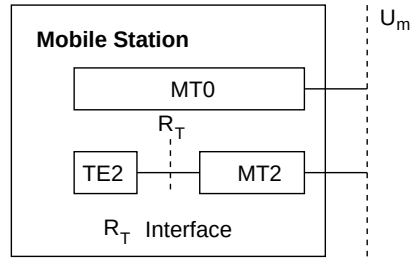


Figure 6.7: Mobile station network interfaces at reference points R_T and U_m

The interface that supports access using traffic and signalling channels is situated at reference point U_m.

Subscriber interface of the line station The line station has a *network termination functional group* (NT) because it is linked to the fixed network over a *transmission line* (TL) interface via an ISDN line (see Figure 6.8). The NT is planned in two versions, and contains the functions defined in ITU-T Rec. I.411 [13]:

- NT1 supports the functions of the NT functional group, special functions of the TL interface and, possibly, the NT2 interface.
- NT2 likewise supports the functions of one or possibly several NTs and an interface to the functional groups TE1 and TETRA TA.

The TE functional group is available in two versions, with TE1 containing an ISDN interface and TE2 a TETRA-specific interface. Both versions support the *man–machine* and eventually also a TETRA TA or an ISDN interface. The *terminal adapting* (TA) functional group is responsible for data-rate adaptation as well as for flow control. In the current version it has interfaces to a TE2 and to an NT2 functional group.

The radio interface The radio interface at reference point U_m is discussed in Section 6.3.4.2.

6.3.4 The Voice+Data Protocol Stack

This section will cover the Voice+Data protocol stack in general. This will be followed by detailed discussion of the radio interface, the physical layer and the link layer, with emphasis on functions, service elements, data structures and states.

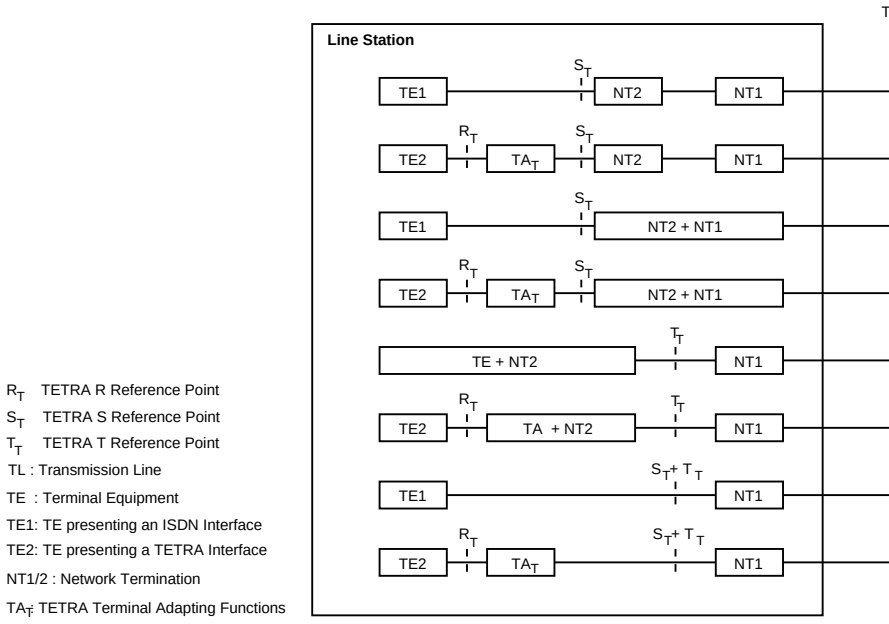


Figure 6.8: Line station network links

6.3.4.1 Structure of the Voice+Data Protocol Stack

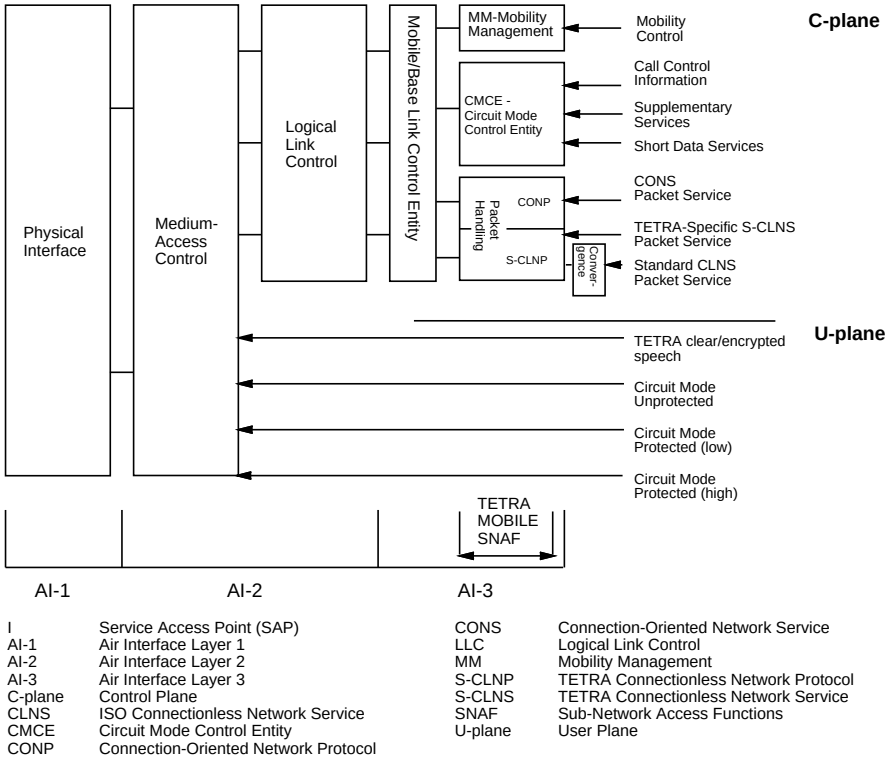
The protocol stack comprises three layers (*air interface*, AI) (see Figure 6.9): the physical layer, which is identical for both V+D and PDO; the data link layer, which is divided into *Medium Access Control* (MAC) and *Logical Link Control* (LLC); and the network layer (N), which is divided into several sublayers and offers management services to base and mobile stations. The MAC layer is based on two protocol stacks: the *user plane*, which is responsible for information transport, and the *control plane* for signalling.

6.3.4.2 The Radio Interface at Reference Point U_m

The radio interface U_m is between the mobile station and the switching and management infrastructure (see Figure 6.6).

The multiplexing structures Similarly to GSM, the TETRA standard uses a combination of *frequency-division multiplexing* (FDM) and *time-division multiplexing* (TDM) with multiple access of the mobile stations (TDMA) (see Figure 6.10). These procedures, along with speech coding and modulation, play an important role in the standard.

The TETRA system uses a cellular concept in which a supply area is divided into cells and has an SwMI installed in the centre. A mobile station is able to measure the receive level of the FDM channel allocated to it. If the level falls



I	Service Access Point (SAP)	CONS	Connection-Oriented Network Service
AI-1	Air Interface Layer 1	LLC	Logical Link Control
AI-2	Air Interface Layer 2	MM	Mobility Management
AI-3	Air Interface Layer 3	S-CLNP	TETRA Connectionless Network Protocol
C-plane	Control Plane	S-CLNS	TETRA Connectionless Network Service
CLNS	ISO Connectionless Network Service	SNAF	Sub-Network Access Functions
CMCE	Circuit Mode Control Entity	U-plane	User Plane
CONP	Connection-Oriented Network Protocol		

Figure 6.9: Architecture of the Voice+Data protocol stack

below a certain limit, a *cell-reselect* procedure is triggered that reroutes the call or data transmission to another cell with a momentary interruption of a minimum of 300 ms. Different versions of the call-reselect procedure are being planned in the standard, with a basic version being obligatory for all mobile stations. The versions based on it are optional and will facilitate a speedier change of cell. It will not be imperative to have a change of FDM channel. The TETRA system does not have any real handover functions, because the users of mobile stations (e.g., taxi fleets) normally do not have a large radius of action.

Frequency-Division Multiplexing Structure Several frequency bands that have been allocated to the TETRA network in Europe do not completely comply with the specifications of the TETRA standard [8] and use additional frequencies above 870 MHz or 915 MHz (see Table 6.3). The frequency bands for both uplink and downlink are the same width. The carrier frequency separation is 25 kHz, and each uplink and downlink band is divided into N carrier frequencies. A G kHz wide band has been added to each boundary of the band in order to prevent interference from outside the band. Thus N and

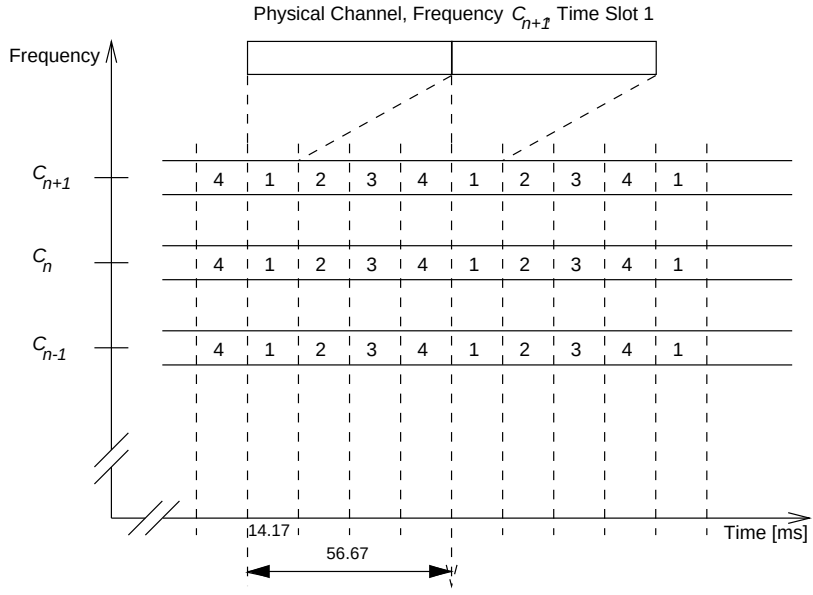


Figure 6.10: Implementation of physical channels using FDM and TDM

G constitute the total bandwidth. The following formulas can then be used to calculate the carrier frequencies. For the uplink

$$F_{up}(c) = F_{up,min} + 0.001G + 0.025(c - 0.5) \text{ MHz}, \quad c = 1, \ldots, N$$

and for the corresponding downlink

$$F_{dw}(c) = F_{up}(c) + D \text{ MHz}, \quad c = 1, \ldots, N$$

D stands for the constant duplex separation between the uplink and the downlink carrier frequencies. $F_{up,min}$ is the cutoff frequency on the lower boundary of the respective frequency band.

Time-Division Multiplexing Structure As can be seen in Figure 6.10, the time axis is split into four time slots, each 14.17 ms in duration corresponding to 510 bits through the use of the TDM technique on each carrier frequency. A periodic time slot produces a physical TDM channel onto which a logical channel is mapped. The logical channel is characterized by its carrier frequency and the time slot recurring at 56.67 ms intervals.

Figure 6.11 shows the TDMA structure for the Voice+Data system. It consists of hyper-, multi- and TDMA frames as well as the time slots and subslots which only occur in uplink traffic. A subslot (half a time slot) consists of 255 bits and a duration of 7.08 ms. Four time slots are combined to form a TDMA frame, and are numbered consecutively here from one to four (*time*

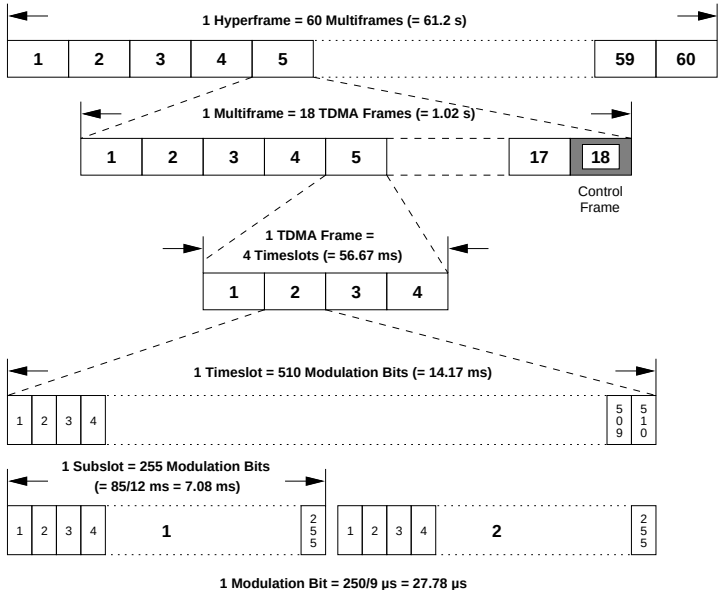


Figure 6.11: TDMA structure of the Voice+Data system

slot number, TN). A TDMA frame is 56.67 ms in length. Eighteen cyclically numbered frames (*frame number*, FN) are combined into a *multiframe*, which is 1.02 s in length.

The 18th frame of a multiframe is reserved for the signalling channels, and is called a *control frame*. If required, other frames can also be reserved for signalling purposes. A *hyperframe* consists of 60 multiframes, and represents the largest structure possible. It has the length of 61.2 s. In contrast to the downlink, the entire frame structure on the uplink is staggered by two time slots to enable a mobile station to transmit and receive at the same time. The frame synchronization of the mobile station is adaptively dependent on the signal propagation delay.

A burst is a set of data bits modulated on a carrier frequency (see Figure 6.12). In V+D it represents the physical content of a time slot, in other words the physical channel information. There are three different types of channel:

- A control channel, *Control Physical Channel* (CP), which exclusively transmits the control channel data.
- A traffic channel, *Traffic Physical Channel* (TP), onto which the logical voice and data channels are mapped.
- An *Unallocated Physical Channel* (UP), which is not allocated to any mobile station and is used for sending broadcast and dummy messages.

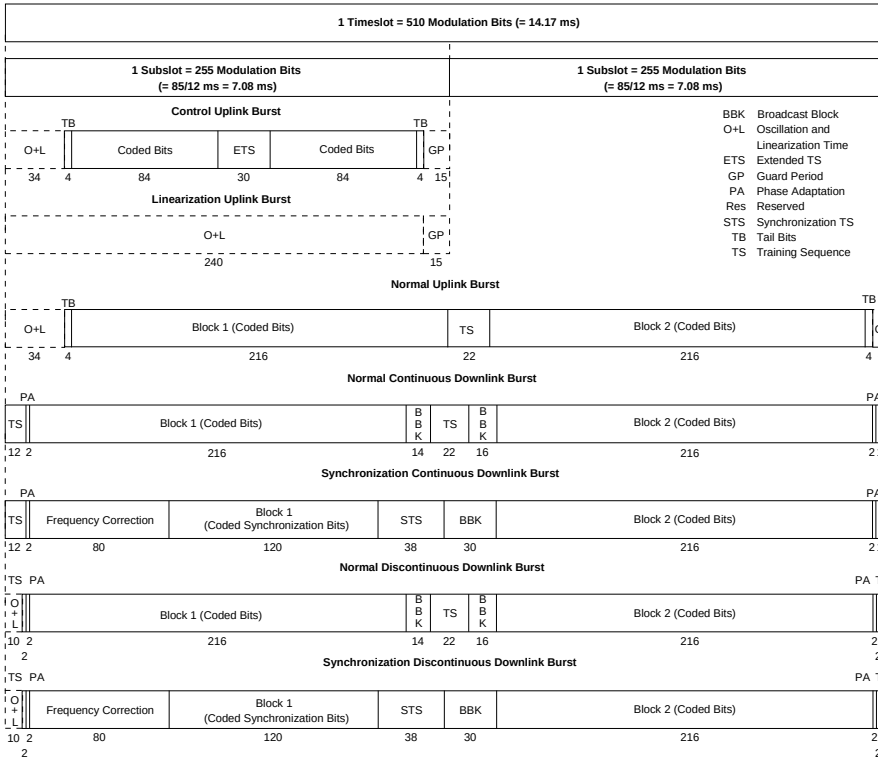


Figure 6.12: V+D downlink and uplink bursts

An existing physical channel always uses the same time slot in successive TDMA frames.

Figure 6.12 (see [6]) presents an overview of the burst structure, which is clarified further below. Three different uplink bursts are defined in the standard: *Control Uplink Burst* (CB), recognized on the basis of its expanded training sequence; *Linearization Uplink Burst* (LB), which allows mobile stations to linearize their transmitters; and *Normal Uplink Burst* (NUB), which is used after the initialization process to transmit control and traffic messages. Whereas the first two bursts only occupy one subslot, the third uses an entire time slot. The multiplex blocks produced by the channel coding are divided into two different blocks (Block Number 1 (BKN1), BKN2) for this.

In addition, there are four downlink bursts: a *Normal* and a *Synchronization Continuous Downlink Burst* (NDB, SB) and a *Normal* and a *Synchronization Discontinuous Downlink Burst*, each of which occupies an entire time slot. This differentiation is made because the base station can chose between *continuous mode* and *discontinuous mode* of transmission.

Logical Channels The logical *channel* is defined as the logical communications path between two or more subscribers, and forms the interface between the communicating protocol entities and the radio subsystem. The time slots and the corresponding physical channels are used to transmit the data of the logical channels. Since four time slots per frame are defined in the TETRA standard, four logical channels can also exist at the same time on one carrier frequency.

Two categories of logical channels are defined:

- Traffic channels
- Control channels

The *traffic channels* (TCH) are used to transmit voice and data in a circuit-switched connection. The *control channels* (CCH) transmit the signalling information and data packets. The following logical channels are defined:

TCH There are four different traffic channels that are suitable for speech and data services: the TCH/S (S for *speech*) for transmitting speech; and the TCH/7.2, TCH/4.8 and TCH/2.4 channels for transmitting data. As their labels imply, they offer net bit rates of 7.2 kbit/s, 4.8 kbit/s and 2.4 kbit/s respectively. The difference in bit rates is due to the differing requirements of the error protection methods used (see Section 6.3.4.4). The speech codec provides a data rate of 4.8 kbit/s, and therefore uses the TCH/4.8.

CCH Five different control channels are available for the transmission of signalling and packet data information.

BCCH The *Broadcast Control Channel* is a unidirectional downlink channel that is used by all mobile stations in general. There are two categories of BCCH: the *Broadcast Network Channel* (BNCH) provides network information, and the *Broadcast Synchronization Channel* (BSCH) supplies information for time and cryptographic synchronization.

LCH The *Linearization Channel* is used by the mobile and base stations for linearizing their transmitters. Here too there are two types: the *Common Linearization Channel* (CLCH) for the uplink of the mobile stations, and the *Base Station Linearization Channel* (BLCH) for the downlink.

SCH The *Signalling Channel* is shared by all the mobile stations, but is able to contain information for only one or for one specific group of mobile stations. The TETRA system functions require that there is at least one SCH per base station. There are three categories of SCH, which depend on the length of a message. A bidirectional *Full-Size Signalling Channel* (SCH/F) always occupies a complete time slot, and the unidirectional *Half-Size Uplink/Downlink Signalling Channels* (SCH/HU, SCH/HD) always occupy half a time slot, in other words a subslot.

AACH The *Access Assignment Channel* is sent on all downlink slots in the broadcast block and contains details about the allocation of the next uplink and downlink slots on the corresponding radio channel. It is transmitted in each *broadcast block* (BBK) of a downlink burst (see Section 6.3.2).

STCH The *Stealing Channel* is a bidirectional channel that is associated with a TCH. It *steals* some of the capacity of the TCH for the purpose of transmitting control information. In half-duplex mode the STCH is a unidirectional channel and is rectified with the respective TCH. The STCH is used with a high priority in signalling, e.g., during the cell change procedure (*cell reselect*).

Mapping logical channels onto physical channels The logical channels are mapped onto physical channels in the lower part of the MAC layer. Table 6.5 shows how the mapping of the logical to the physical channels is defined.

Some of the important aspects of the table are explained below. The BCCH and the CLCH are mapped onto the control frame (18th frame of a multiframe) of a physical control or traffic channel through functions of the time slot and the multiframe number. The following algorithms are used:

$$\begin{array}{ll} \text{Downlink:} & \text{BNCH, if } \texttt{FN} = 18 \text{ and } (\texttt{MN} + \texttt{TN}) \bmod 4 = 1 \\ & \text{BSCH, if } \texttt{FN} = 18 \text{ and } (\texttt{MN} + \texttt{TN}) \bmod 4 = 3 \\ \\ \text{Uplink:} & \text{CLCH, if } \texttt{FN} = 18 \text{ and } (\texttt{MN} + \texttt{TN}) \bmod 4 = 3 \end{array}$$

Furthermore, the base station is able to map the CLCH onto the uplink subslot 1 and the BLCH to the downlink block 2 of a physical control channel (CP). It accomplishes this on a slot-to-slot basis and indicates it on the AACH. Mobile stations can linearize their transmitters to any CP when there is a CLCH if in so doing they are not violating any of the rules on mapping and the linearization process is longer than one multiframe. The BLCH is also mapped onto a downlink block 2 if an SCH/HD or a BSCH is mapped onto the first block. However, there cannot be more than one BLCH per four multiframes on one carrier.

If a CLCH is not mapped onto the first subslot, an SCH/F or two SCH/HUs can be mapped onto the uplink. Otherwise only subslot 2 can be used for an SCH/HD. An SCH/F or two SCH/HUs can be mapped onto the downlink if a block 2 is not being used for a BNCH.

The base station indicates on the AACH which logical channel type will be used on the next uplink time slot. This indication is only one frame in length and valid for one physical channel. The logical channel number on the downlink is determined by the type of *training sequence* (TS) used. When several downlink traffic channels are used by a connection, the uplink and downlink SCHs are mapped onto the control frame (FN 18) and to the lowest time slot number. If several uplink traffic channels are used by a connection

Table 6.5: Mapping logical channels to physical channels

Logical channel	Direction	Burst type	SSN/BKN	Physical channel	FN	TN
BSCH	DL	SB	BKN1	CP, TP	18	$4-(\texttt{MN}+1) \bmod 4$[a]
				UP	1...18	1...4
BNCH	DL	NDB, SB	BKN2	CP, TP	18	$4-(\texttt{MN}+3) \bmod 4$[a]
				CP, UP	1...18	1...4
AACH	DL	NDB, SB	BBK	CP, TP, UP	1...18	1...4
BLCH	DL	NDB, SB	BKN2	CP, UP	1...18	1...4
				TP	18	
CLCH	UL	LB	SSN1	CP, TP	18	$4-(\texttt{MN}+1) \bmod 4$[a]
				CP, UP	1...18	1...4
SCH/F	DL	NDB	BKN1, BKN2	CP	1...18	1...4
	UL	NUB				
SCH/HD	DL	NDB, SB	BKN1, BKN2	CP, UP	1...18	1...4
				TP	18	
SCH/HU	UL	CB	SSN1, SSN2	CP	1...18	1...4
					18	
TCH	DL	NDB	BKN1, BKN2	TP	1...17	1...4
	UL	NUB				
STCH	DL	NDB	BKN1, BKN2	TP	1...17	1...4
	UL	NUB				

[a]Mapping to respective time slot is stipulated

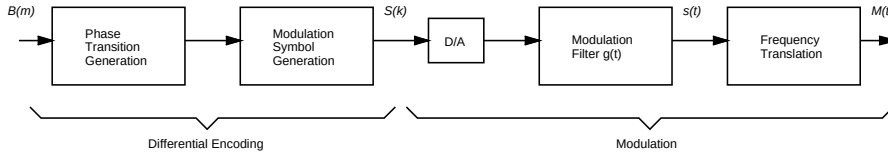


Figure 6.13: Block diagram of the modulation process

then the uplink and the downlink SCHs are also mapped onto the control frame, but to the highest time slot number.

Logical traffic channels (TCH) are mapped onto frames 1–17 of the physical traffic channels (TP), in other words onto block 1 and block 2. An STCH can be mapped onto all frames available for traffic, and always first steals the first block of a time slot. This is indicated by a special training sequence.

6.3.4.3 The Physical Layer in the TETRA Standard

The physical layer forms the physical radio interface of the TETRA system. It generates the bursts, which consist of a range of symbols and are transmitted and received.

It is responsible for the following functions, which are explained in detail below:

- Radio-oriented:
  - Modulation and demodulation
  - Transmitter and receiver management
  - Radio frequency characteristics
  - Fine tuning of radio parameters
- Bit and symbol-oriented: symbol synchronization
- Burst formation:
  - Receiving and transmitting of data from and to MAC layer
  - Slot-flag coding/decoding
  - Scrambling and deciphering

The modulation method is $\pi/4$-*Differential Quaternary Phase Shift Keying* ($\pi/4$-DQPSK) at a modulation (gross) bit rate of 36 kbit/s.

The modulation process is illustrated in Figure 6.13. The sequence $B(m)$ of the modulation bits to be transmitted is represented by a differential coding in a sequence of modulation symbols $S(k)$ based on the following rule:

$$\begin{aligned} S(k) &= S(k-1)e^{jD\Phi(k)} \\ S(0) &= 1 \end{aligned} \tag{6.1}$$

As shown below, the phase translation $D\Phi(k)$ is dependent on $B(m)$:

$B(2k-1)$	$B(2k)$	$D\Phi(k)$
1	1	$-3\pi/4$
0	1	$+3\pi/4$
0	0	$+\pi/4$
1	0	$-\pi/4$

It follows from this definition that $S(k)$ can assume eight different values. The modulated signal $M(t)$ on the carrier frequency f_c is

$$M(t) = \mathrm{Re}\{s(t)e^{(j\cdot 2\pi f_c t+\Phi_0)}\} \tag{6.2}$$

with Φ_0 representing the phase offset and $s(t)$ the complex envelope of the modulated signal:

$$s(t) = \sum_{k=1}^{K} S(k)g(t-t_k) \tag{6.3}$$

Here $g(t)$ stands for the inverse Fourier transform of the *square-root raised-cosine* spectrum and K is the maximum number of symbols.

The management functions of the transmitter and the receiver are responsible for the choice of frequency bands and the transmit power. The output power is controlled so that it is increased in very short ramps until the full strength of transmitter power is reached in order to minimize interference in the system. This requires an adherence to a number of limit values, which is guaranteed by the management function. Optimization in this area will result in a reduction in the bit-error ratio. Detailed explanations about this function can be found in [8, 9].

A fine tuning of the radio parameters results in a frequency correction using a special *frequency correction sequence*, which is localized in a synchronization burst within the BSCH. This ensures that there is only minimal deviation from the carrier frequency. It also requires a high level of accuracy in the oscillator of the mobile station. The *power control* function also ensures that a mobile station is always using the adjusted transmitter power. This is controlled by the MAC layer.

Similarly to frequency correction, symbol synchronization is achieved through a specific training sequence, which is longer for an initial synchronization than it is for existing connections. The information for the synchronization is contained in bursts. This enables the physical layer to recognize clearly the boundaries of a burst. As a way of ensuring that a mobile station remains synchronized for a longer period of time, after completion of the synchronization process timers are started that indicate when the next frame, multiframe or hyperframe will begin or end. This requires a high level of accuracy in the time control of the mobile station.

This process allows a mobile station to detect the beginning and the end of a burst, and to use this knowledge to transmit data in order to map the MAC-PDUs (*protocol data units*) to the bursts as well as to place its own

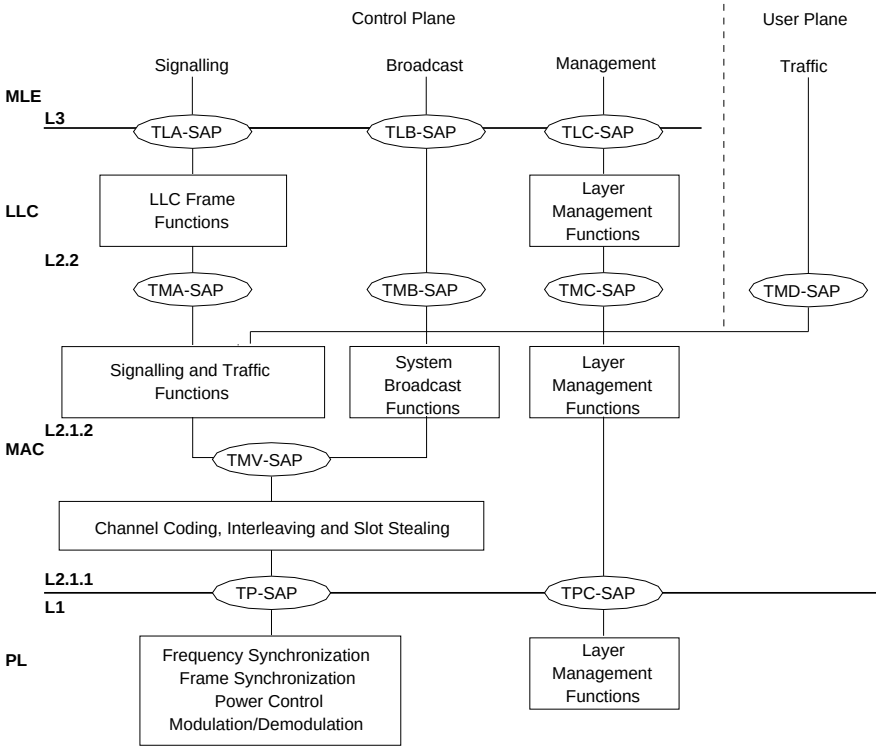


Figure 6.14: Architecture of the Voice+Data data link layer

specific information correctly. During reception this process is reversed. The information specific to the physical layer is removed and the MAC-PDU is retrieved from the burst and forwarded to the MAC layer. The different types of bursts are explained in detail in Section 6.3.4.2 and [8].

The slot-flag appears in two versions and indicates through the bursts contained in the training sequences whether an entire slot (`SF=0`) or half a slot (`SF=1`) is being occupied by the signalling data. Its other tasks are *scrambling* and *descrambling*. Information on the scrambling code is passed on to the receiving station in a non-coded MAC-PDU by a socalled *colour code*. The scrambling is part of the channel coding and is described in Section 6.3.4.4.

6.3.4.4 The Data Link Layer in the TETRA Standard

The architecture of the *data link layer* of the TETRA standard V+D is presented in Figure 6.14. It is organized into the sublayers MAC and LLC, with the MAC layer divided into a *lower* and an *upper* MAC at the TMV-SAP (*service access point*). On the upper boundary of the data link layer three service access points of the *link control entity* (LCE) (see Figure 6.9) offer

their services in the network layer: TLA (*TETRA-LLC-A*), TLB and TLC. The TLA-SAP provides services for the bidirectional transfer of addressed signalling and data messages. The TLB-SAP provides non-addressed data transfer in which broadcast messages with system information are sent unidirectionally from the base station to the mobile station. The TLC-SAP is only available on the mobile station side, and is used for control and management information. The LLC and MAC sublayers correspond over the SAPs TMA (*TETRA-MAC-A*), TMB and TMC, which have the same functions as the corresponding SAPs of the LLC sublayer. The MAC sublayer contains another SAP (TMD) for the user plane, which transports information over a line-switched connection. The lower and upper MAC sublayers communicate over a virtual TMV-SAP, which provides services for actual radio transmission, such as channel coding, interleaving and slot stealing. At the interface to the physical layer the TPC-SAP (analogously to TLC- and TMC-SAP) offers access to the local layer management. Through the TP-SAP, the lower MAC layer corresponds to the physical layer.

Medium access control As shown in Figure 6.14, the functions of the *medium access control* (MAC) sublayer mainly consist of channel coding, channel access control and radio resource management, which, depending on transmission mode, offer their services at the three SAPs—TMA, TMB and TMC—for signalling and packet data mode or at the TMD-SAP for user traffic mode. After the setup of a circuit-switched voice or data transmission connection for which the traffic mode has been defined, signalling messages can be transmitted with the help of a slot-stealing mechanism.

A base station is allowed to transmit continuously or discontinuously. In the latter case the respective base station interrupts its transmission if there is no further information to be transmitted or if it is sharing the same radio channel for signalling purposes with other base stations that also find themselves in *timesharing mode*. The functions, data structures, service elements and states of the MAC layer are discussed below.

Channel coding Channel coding in the Voice+Data standard is carried out as diagrammed in Figure 6.15. Information bits contained in MAC blocks, called Type 1 bits, are coded by a (K_2,K_1) block coder. Except for the AACH, in which Type 2 bits are produced by means of an abbreviated (30,14) Reed-Muller code [7, 14], at this initial stage of channel coding a systematic cyclical block code [16] is produced from K_1 Type 1 bits $b_1(1), b_1(2), \ldots, b_1(K_1)$, and $K_2 = K_1 + 16$ Type 2 bits $b_2(1)$,$b_2(2)$,...,$b_2(K_2)$.

The code words f are sent together with Type 1 information bits b_1, thereby producing the Type 2 bits b_2 as

$$b_2(k) = \begin{cases} f(k-1) & \text{for} \quad k = 1, 2, \ldots, 16 \\ b_1(k-16) & \text{for} \quad k = 17, 18, \ldots, K_2 = K_1 + 16 \end{cases} \tag{6.4}$$

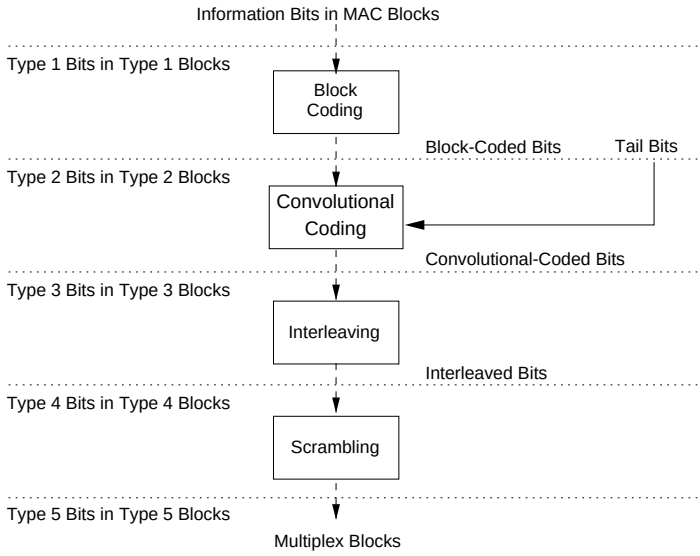


Figure 6.15: Diagram of TETRA channel coding

Since $K_2 \in \{76, 108, 140, 284\} \neq 2^m - 1$ for $m \in \mathbb{N}$, the block code used here is, according to [14, 16], neither a Hamming code nor a *Bose–Chaudhuri–Hoquenghem* code (BCH).

At the next stage of channel coding the K_2 Type 2 bits b_2 are coded by a (4,1,5)-rate-compatible, punctured convolutional coder with the code ratio 1/3 or 2/3 to K_3 Type 3 bits b_3 with $K_3 = \frac{t}{2}K_2, t \in \{3, 6\}$. First the output V is computed by a 16-state convolutional coder with the ratio 1/4 from the Type 2 bits $b_2(k)$.

The puncturing to a 16-state RCPC (*rate-compatible punctured convolutional*) code with the ratio $2/t$ is carried out through the selection of $K_3 = (t/2)K_2$ Type 3 bits from the 4 K_2 coded bits V:

$$\begin{aligned} b_3(j) &= V(k), \quad j = 1, 2, \ldots, \frac{t}{2}K_2, \\ k &= 8\lfloor (j-1)/t \rfloor + P\Big(j - t\lfloor (j-1)/t \rfloor\Big) \end{aligned} \tag{6.5}$$

With a code ratio of 2/3 the three punctured coefficients are $P(1) = 1,\ P(2) = 2,\ P(3) = 5$, with $t = 3$. With a code ratio of 1/3 the six punctured coefficients are $P(1) = 1,\ P(2) = 2,\ P(3) = 3,\ P(4) = 5,\ P(5) = 6,\ P(7) = 7$, with $t = 6$. Two different punctured convolutional codes that originate from the same original code are considered rate-compatible when all non-punctured, i.e., not deleted, bits of the high-rate convolutional code are also contained in the low-rate one. Four tail bits with the value 0 are added to the Type 2 bits b_2 in order to initialize the convolutional coder after coding.

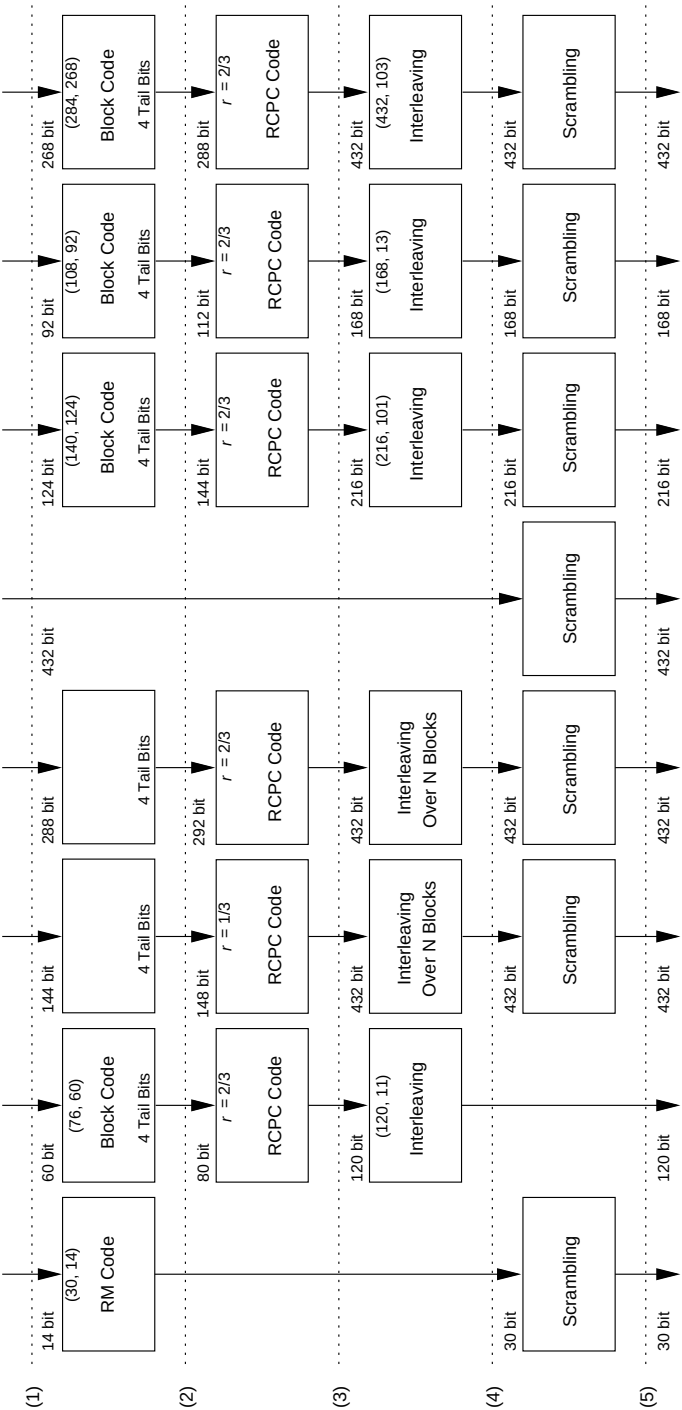


Figure 6.16: Structure of V+D channel coding

In block Type 3 b_3 bits are scrambled into Type 4 b_4 bits by means of a (K, a) block interleaver where $K = K_3$, e.g., $a = 101$ (as shown in Figure 6.16), in order to prevent bursty-type transmission errors:

$$\begin{aligned} b_4(k) &= b_3(i), \quad i = 1, 2, \ldots, K = K_3 = K_4, \\ k &= 1 + \big((a \cdot i) \bmod K\big). \end{aligned} \tag{6.6}$$

For 432-bit long Type 3 blocks an alternative of bit interleaving over N blocks in a sequence of Type 4 blocks is being planned in two stages. In the first stage $b'_3(m, k)$ interleaved bits are generated from Type 3 b_3 bits as k bits of the m block:

$$b'_3(m, k) = b_3\big(m - j,\ j + (i \cdot N)\big), \qquad \begin{aligned} k &= 1, 2, \ldots, 432, \\ m &= 1, 2, \ldots, N, \end{aligned} \tag{6.7}$$

$$\begin{aligned} j &= \big\lfloor (k-1)/(432/N) \big\rfloor, \\ i &= (k-1) \bmod (432/N) \end{aligned}$$

The $B'_3(m)$ blocks thus created are interleaved into Type 4 $B_4(m)$ blocks:

$$b_4(m, i) = b'_3(m, k), \quad \text{with} \quad i = 1 + (103k) \bmod 432. \tag{6.8}$$

At the last stage K_4 Type 4 b_4 bits are coded into K_5 Type 5 b_5 bits through the use of a linear feedback register in which a coding sequence p, which is formed from a generator polynomial and an initialization sequence for P, is added to b_4 modulo 2:

$$b_5(k) = b_4(k) + p(k), \quad k = 1, 2, \ldots, K_5 = K_4. \tag{6.9}$$

Figure 6.16 shows the channel coding based on the logical channel for the Voice+Data protocol stack. What is mainly noticeable is that the synchronization data of the BSCH is not coded, because when mobile stations are switched on they first listen in on the BSCH and at this point it is too early for any agreement on a cryptographic code. Furthermore, it should be noted that the different transmission rates for the TCH are achieved through a channel coding according to the quality of service aimed at such that TCH/2.4 has the code ratio $r = 1/3$, TCH/4.8 has the ratio $r = 2/3$ and TCH/7.2 is not convolutionally coded at all. Block coding does not occur with any of the variants of the traffic channel.

Since, according to [14], a Reed–Muller block coder is particularly efficient for small volumes of data, it is used only on the 14-bit wide Access_AssignPPDUs of the AACH. Convolutional coding and bit interleaving are no longer carried out for these small volumes of data.

Channel access control Channel access control contains the functions for frame synchronization that detects the frame number within a multiframe:

- Fragmentation of the PDUs received by the LLC sublayer into several MAC-SDUs (*service data units*) for transmission and *reassociation* on the receiver side of the received fragments.
- Multiplexing/Demultiplexing of the logical channels and the formation of multiframes, which are therefore explicitly counted.
- Synchronization of multiframes through synchronization blocks of the base station that are transmitted on the downlink and contain, e.g., information about the frame number.

An important function is the *random-access protocol* to access the channel by the mobile stations. During initial access, the MAC layer of a mobile station uses a Slotted-ALOHA access protocol in order to send information to the base station when no request for the transmission of a message exists. When the base station requests information or during a reserved, circuit-switched connection, the mobile station uses reserved access. With appropriate selection of the access parameters (see Section 2.8) it is possible:

- To control collision resolution of access by the individual mobile stations.
- To minimize access delay for a particular traffic load and to maximize throughput.
- To avoid protocol instability.
- To dynamically prevent random access for different access priorities or for selected categories of groups and users.
- But at the same time, and independently of the *grades of service* (GoS), to offer access to different categories of groups and users.

A base station offers different mobile stations the possibility of random access through the use of four *access codes*, which are referred to as A, B, C and D and should be interpreted as priorities. The access code represents a combination of user services and is permanently allocated to mobile stations by the network operator. It is not necessary for every code to be accessible by all mobile stations. A mobile station is only allowed access if the conditions for access to the subslot of a particular code have been met. Base stations inform mobile stations that they have access by the means of two different PDUs. An Access_Define PDU (see Section 6.3.4.4) is sent in the intervals specified by the network operator, and provides information about the access code and the priority allowed for sending as well as the time period and number for a repeated access. An Access_Assign PDU that is contained in each downlink time slot is used to indicate the access rights of mobile stations to the uplink time slots, and a *traffic usage marker* indicating permission to transmit and receive in the current uplink and downlink time slot. The information of both PDUs is stored until receipt of an update.

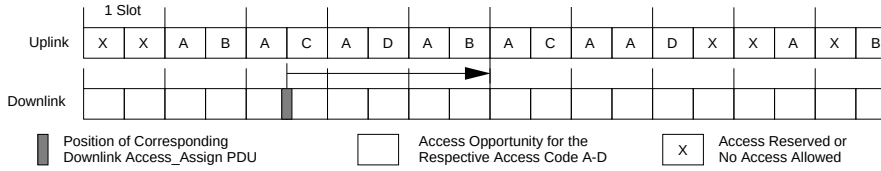


Figure 6.17: Access procedures of the V+D random-access protocol

Example of uplink subslots characterized by different access codes or a reservation and indicated in an Access_Assign-PDU are shown in Figure 6.17. A mobile station wishing to send a message first compares the transmitted conditions for access with those that it is allowed. If access is allowed, it uses the MAC_Access-PDU that contains the message to be transmitted and sets its access timer. If the message is fragmented, it makes a reservation request for several time slots in the MAC_Access-PDU sent to the base station. The base station responds to the MAC_Access-PDU with a MAC_Resource-PDU, in which it acknowledges the successful receipt of the message. The mobile station is also notified whether time slots have been reserved, and if so, which ones. These are represented by crosses in Figure 6.17. If a mobile station still has not received a response to the Mac_Access-PDU when the access timer has expired, it repeats the random access. The random-access protocol is explained in detail in [4].

Radio resource management These functions are only sometimes available in mobile and base stations or in both. They should be available at all times, even without the incorporation of layer 3. The following functions are among those being planned:

- Measurement of *bit-error ratio* (BER) and *block-error ratio* (BLER), independently or under the control of layer 3.
- Path-loss calculation through monitoring of current and neighbouring cells. Calculation of path-loss parameters based on information from the current cell as well as periodic *scanning*. Determination of path-loss parameters of neighbouring cells from the scanned signal level.
- Address management for individual, group and broadcast calls. Two addresses can be used: a copy of the *individual* or *group subscriber identity*, or an *event label* that can be used by the MAC layer to reduce the amount of signalling required during existing connections.
- Management of *power control* in layer 1.
- Setup of a connection through the use of the details on frequency, time slot and *colour code* of the *mobile link entity* (MLE).
- Storage of control data and speech frames until time of transmission.

Service elements of TMA, TMB, TMC and TMD service access points (SAP) A TMA-SAP is used by both LLC connection types—basic link and advanced link (see Section 6.3.4.4)—for the transmission of signalling and packet data information (see Figure 6.14). Five *service primitives* (SP) are available for this purpose.

The LLC sublayer uses the TMA_Unitdata_req-SP to ask the MAC sublayer to send LLC frames, which at this interface are called TM-SDUs (SDU=service data unit).

The TMA_Unitdata_ind-SP is used by the MAC sublayer in order to forward received messages.

A TMA_Cancel_req-SP can be used to clear a TMA_Unitdata_req that was sent by the LLC sublayer.

With a TMA_Release_ind-SP an LLC *advanced link* connection can be interrupted (see Section 6.3.4.4) if the MAC sublayer releases a channel and all *advanced links* on this channel are quietly interrupted.

In the TMA_Report_ind-SP the MAC sublayer presents information to the higher layers on the sending of Request procedures. During the initialization process, this SP indicates whether a connection has been established successfully and the partner entity has acknowledged receipt of the message. This SP is also used to notify the higher layers of errors, such as failed random access, which occur during these processes.

The TMB-SAP carries out the transfer of non-addressed broadcast messages that contain information on network and system organization. In this case no special TMB-SAP-related functions exist in the LLC sublayer, and it is necessary for Requests at the TLB-SAP to be directly mapped into TMB-SAP Requests and TMB-SAP Indications into TLC-SAP Indications. This is why the same service primitives (explained further in Section 6.3.4.4) also exist at both these SAPs.

The information for local layer management is transmitted in the same way over the TMC-SAP. There are no TMC-SAP-relevant functions in the LLC sublayer. Therefore Requests and Responses at the TLC-SAP have to be mapped to Requests and Responses at the TMC-SAP, and vice versa Indications and Confirms at the TLC-SAP to Indications and Confirms at the TMC-SAP.

In a circuit-switched voice or data connection the voice or data frame is forwarded to the MAC sublayer through the TMD-SAP, which forms the interface between the TETRA speech codec and the MAC sublayer. Voice frames are permitted to contain coded or non-coded speech; the information they contain is irrelevant to the MAC sublayer. Individual time slots or subslots can be *stolen* for end-to-end signalling between users of circuit-switched services. Depending on whether a time slot contains voice, data or signalling information, or whether an entire or half a time slot is stolen, appropriate adapted channel coding is carried out. The TMD_Unitdata-SPs Request and Indication are used to transfer the data or voice frame that has been received or is to be transmitted. The TMD_Report_ind-SP informs the MAC service user of the status of the current transmission.

Table 6.6: Mapping of V+D-MAC-PDUs to logical channels

MAC-PDU	Logical channels	Type of information
Access_Assign	AACH	MAC internal information
Access_Define	SCH/HD, SCH/F, STCH	
MAC_Access	SCH/HU	TMA-SAP information
MAC_End_HU	SCH/HU	
MAC_Data	SCH/F, STCH	
MAC_Resource	SCH/HD, SCH/F, STCH	
MAC_Frag	SCH/HD, SCH/F	
MAC_End	SCH/HD, SCH/F, STCH	
MAC_Traffic	TCH	TMD-SAP information
MAC_U_Signal	STCH	
Sync	BSCH	TMB-SAP information
Sysinfo	BNCH	

Data structures Twelve different PDU types are defined in the MAC sublayer (see Table 6.6).

Access_Assign This 14-bit (net) long PDU is contained in the *broadcast block* (BBK) (see Figure 6.12) of each downlink time slot, and carries information on the current time slot and access rights to the uplink time slot, which is delayed by two time slots and is associated with this downlink time slot. The content varies depending on whether the PDU is being sent in one of the frames from 1 to 17 or in frame 18.

Access_Define The random code specifically valid for random access, the ALOHA parameter and the access procedure are stipulated by the BS in this downlink-PDU, which occupies either half a time slot or an entire time slot and uses the logical SCH/HD or SCH/F channel. At the user plane (see Figure 6.14) the STCH channel is also occupied.

MAC_Access This PDU contains LLC data and is sent on the uplink and in half subslots for random access. A TM-SDU is sometimes fragmented and divided among several MAC_Access- and MAC_Frag-PDUs or MAC_End_(HU)-PDUs [3]. Filler bits are added when a message is shorter than the number of user data bits available.

Depending on whether a 24-bit wide layer-2 address (*short subscriber identity* `SSI`) is used for the initial access or a 10-bit wide *event label*, `EL`) allocated by a BS to reduce the addressing overhead for a transaction, either 56 bits or 70 bits of net user data can be transported in this PDU. The beginning and the end of a fragmented TM-SDU are indicated in the MAC_Access-PDU and in the MAC_End_(HU)-PDU. The MAC_Access-PDU is mapped to the SCH/HU channel.

MAC_End_HU This PDU is only sent on the uplink, and in a fragmentation indicates that it contains the last fragment. It always occupies only half a time slot.

MAC_Data This is for the transfer of LLC data in full time slots on the uplink or, with a *stolen* channel, in the first and, if required, in the second time slot. In the second time slot the MAC_Data-PDU can be used to send other data at the signalling level. This PDU can only be used in reserved transmission. With a MAC_Access-PDU or a MAC_Data-PDU, a base station obtains uplink reservation requests from the length data contained in the *length indication or capacity request*. A fragmentation of the message is sometimes also indicated. Depending on the type of addressing—(`SSI` or `EL`)—231 bits or 245 bits of user data net can be sent.

MAC_Resource This PDU is used by a base station to transmit information at the signalling level on the downlink. It is also able to confirm that a random access has been successful and in some cases to allocate capacity on the uplink. Depending on the addressing and whether a half or a full time slot is used, the maximum length of user data is 95 bits or 239 bits. A MAC_Data-PDU or a MAC_Resource-PDU can be used to indicate in the PCI that the second subslot is being stolen and contains a MAC_U_Signal-PDU.

MAC_Frag This PDU is used to send further fragments of fragmented messages of the signalling level from the LLC sublayer on the downlink. The header attached to the PDU is 4 bits long, which means that 264 user information bits can be sent.

MAC_Traffic Voice and data frames of circuit-switched connections are transmitted with this PDU. Except for the 2 bits used for the identification of the PDU type, this PDU exclusively contains user data.

MAC_U_Signal End-to-end signalling at the user level is carried out through this PDU from the TMD-SAP of the sender to the TMD-SAP of the partner entity. It is always mapped onto the STCH. Similarly to the MAC_Traffic-PDU, this PDU also exclusively contains user data except for the first three bits. An indication whether one or two subslots are being stolen is given in the third bit. Even if the second subslot is occupied, it can contain LLC data.

MAC_End This PDU is only sent on the downlink, and occupies either a whole time slot on the SCH/F channel or half of one on the SCH/HD channel. It is sent with the last fragment of fragmented LLC signalling data.

Sync Per broadcast, a base station sends non-addressed Sync- and Sysinfo-PDUs on the logical BSCH and BNCH channels to the mobile stations, which decode all of the information contained in both PDUs.

A Sync-PDU establishes the initialization code (*colour code*) responsible for the channel coding, and contains the current multiframe, frame and time slot numbers (see Section 6.3.4.2). It also notifies mobile stations whether it is transmitting continuously or discontinuously and whether it is in *Timesharing mode* with other base stations.

Sysinfo A mobile station computes the path loss for the current cell using the parameters sent by the base station per broadcast in this PDU. The PDU notifies the mobile station of the carrier frequency and the frequency band of the main control channel and provides details about the cell cluster and power adaptation.

Table 6.6 lists the logical channels over which the MAC-PDUs are forwarded to the lower MAC sublayer for channel coding, and which type of information is contained in these PDUs. The data of the STCH is either transmitted over the SCH/HU or the SCH/HD. The allocation of a whole time slot (SCH/F) to the STCH is also possible.

State Diagram and Transitions of a Mobile Station The MAC protocol of a mobile station comprises seven states (see Figure 6.18 and Table 6.7), with `M0 NULL` corresponding to the switched-off state. When a mobile station is switched on (1), it is in the state `M1 IDLE UNLOCKED`. After it is switched off (2), which can take place in any state, it returns to the `M0 NULL` state. Only after an MS has received the Sync and Sysinfo PDUs from the BS per broadcast and a cell has reached a sufficient level of quality, does it select this cell and inform the MLE accordingly.

It then moves to the state `M2 IDLE LOCKED` (3). In the event that the MAC sublayer determines a deterioration in signal quality and notifies the MLE, it will then be instructed by the MLE to find a new cell with sufficient quality, and will return (4) to the state `M1 IDLE UNLOCKED`.

Depending on the actions of the user, after a specific period of inactivity the MS is operated in power-saving mode and moves to `M3 STAND-BY` state (5).

After the timer, which can be set to a waiting time between 1 and 432 frames (57 ms to 25.9 ms) based on the user's discretion, expires, there is again an exit from power-saving mode and the MS returns (6) to the state `M2 IDLE LOCKED`.

In this state an MS only receives. It monitors the SCH for the paging messages sent to it by the BS and for general system information. The MS also prepares for future transmitting jobs whereby it analyses Access_Define-PDUs received over the SCH and observes the AACH. If there are no paging messages being received or data being sent, the MS (13) scans the signal

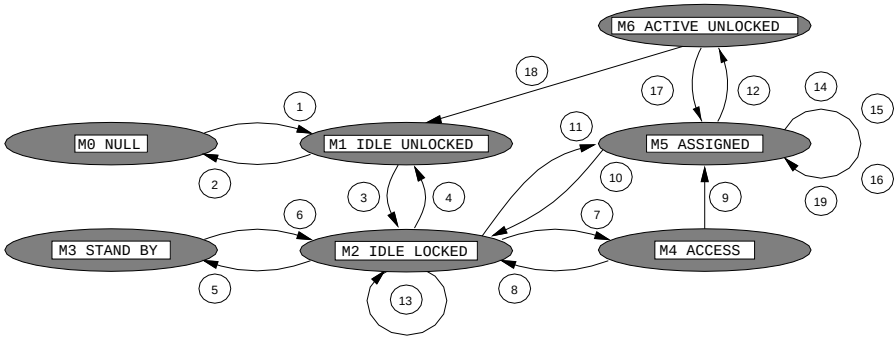


Figure 6.18: State diagram of a V+D mobile station in the MAC sublayer

strength of the neighbouring cells and attempts to decode all the network information of these cells.

If the MS receives a paging message or packet data from the BS or carries out a random access, it finds itself (7) in active `M4 ACCESS` state and, in the case of a circuit-switched connection, waits until it is allocated a traffic channel (TCH). When it is in this state, all the packet data is transmitted if the connection is not a circuit-switched one. If the timer runs out or the transmission of packet data has been completed, the MS returns (8) to the `M3 IDLE LOCKED` state.

Once a BS has exclusively allocated a channel to an MS or to a group of MSs, there is a move (9) to the `M5 ASSIGNED` state. The speech and data frames of a circuit-switched connection are transmitted in this state. The end-to-end signalling between users takes place through the use of slot-stealing mechanisms (15).

Similarly to state M2, an MS regularly measures (14) the signal strength of the neighbouring cells during a call and tries to decode all the network information of these cells. If after the receipt of this information the MS prefers a different cell, it requests a handover (19), which activates the fast-call-reestablishment procedure.

At the end of a transmission (*transmission trunking*) or a call (*message trunking*), the MS moves (10) to the state `M2 IDLE LOCKED`. With a group call or the forwarding of a call with transmission trunking, there is a direct channel allocation and a return (11) to the `M5 ASSIGNED` state. Since the channel is permanently occupied in a message trunking situation, transmitting requests are processed in this state (16).

In V+D transmission trunking is the method in which a traffic channel (TCH) is only occupied if there is actual voice activity, and then released again immediately. With message trunking the traffic channel is permanently occupied for the entire duration of a call, which, compared with transmisison trunking, results in a lower protocol overhead but also lower throughput.

Table 6.7: States of a V+D mobile station in the MAC sublayer

1	The MS is switched on.
2	The MS is switched off. This changeover can be carried out from any state.
3	A cell with sufficient quality is found, the MLE layer informed and the cell selected.
4	The MAC layer has informed the MLE layer of the deterioration of signal quality and is instructed to return to search mode.
5	The MS switches to power-saving mode.
6	The MS leaves power-saving mode.
7	The MS carries out a random access or is paged. In both cases there is a wait in the `ACCESS` state until the allocation of a traffic channel (circuit-switched connection).
8	Time-out or end of packet-data transmission.
9	Receipt of the channel allocation.
10	End of a transmission (transmission trunking) or of a conversation (message trunking).
11	Direct channel allocation (group call or call diversion—transmission trunking).
12	Signal loss during a call. MS begins to restore the connection after a waiting period.
13	Search of neighbouring cells during idle state.
14	Search of neighbouring cells during assigned state.
15	Slot-stealing procedures.
16	Voice or data transmission with message trunking.
17	Return after brief signal loss.
18	Continuation of call reestablishment procedure.
19	Handover (fast call reestablishment) to a neighbouring cell.

If a mobile station loses contact with the base station (12), it begins to reestablish the connection after a short waiting period while in `M6 ACTIVE UNLOCKED` state. After successful restoration of the connection, it returns (17) from this short signal break to the state `M5 ASSIGNED`. If this is not the case, it moves (18) to the state `M1 IDLE UNLOCKED` and continues the establishment of the connection in this position.

State Diagram and Transitions of the Base Station Four states are defined for the V+D base station in the MAC sublayer (see Figure 6.19 and Table 6.8), with the BS in state `B0 NULL` before the switching-on process. The transition to state `B1 ACTIVE` takes place after the BS has been switched on (1). When it is switched off (2), the BS returns to the original state, no matter which state it is currently occupying.

Since the BS produces a new instance of the MAC entity for each connection, it can assume a number of states B1 to B3 at the same time.

Synchronization and system information (Sync- and Sysinfo-PDUs) are sent to inactive MSs in state `B1 ACTIVE`. In addition, the BS transmits access def-

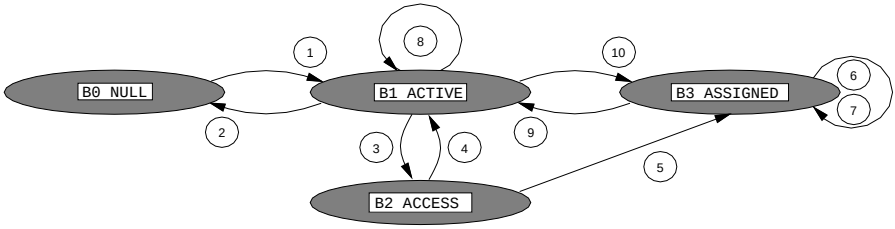


Figure 6.19: State diagram of the V+D base station in the MAC sublayer

Table 6.8: State transitions of the V+D base station in the MAC sublayer

1	The BS is switched on.
2	The BS is switched off.
3	Successful reception of a random access or paging message by an MS.
4	Time-out or end of packet-data transmission.
5	Traffic channel (TCH) is allocated.
6	Slot-stealing in traffic mode.
7	Sending permission to transmit (message trunking).
8	Sending permission to transmit (transmission trunking).
9	End of transmission (transmission trunking) or a call (message trunking).
10	Direct allocation of a traffic channel (TCH) for a group call or an existing connection (transmission trunking).

initions (Access_Define-PDUs) and access assignments (Access_Assign-PDUs) to enable MSs to prepare themselves for random access in the future. However, data transfer traffic does not take place in this state.

After a BS has successfully recognized (3) and received a random-access request from an MS, it moves into state `B2 ACCESS`. In the event that the BS wants to establish a connection with the MS, the latter is informed (paging message) and a transition (3) to B2 takes place. Packet data is transmitted between BS and MS in this state. Upon completion of this transmission or the running out of a timer, the BS returns (4) to the state `B1 ACTIVE`.

If an MS has requested the allocation of a traffic channel during its random access, the BS acknowledges this and moves (5) into state `B3 ASSIGNED`. In this state voice and data frames are exchanged and transmission capacity for end-to-end signalling between users can be stolen (slot-stealing, 6). In the case of message trunking the connection is permanent and transmission requests are acknowledged in this state (7).

Upon completion of a call (message trunking) or a transmission (transmission trunking), the BS returns (9) to state `B1 ACTIVE`, where it acknowledges further requests for transmission (8) in the case of transmission trunking and directly allocates a traffic channel for an already-existing connection (9). A direct channel allocation is also made in the case of group calls.

Logical Link Control

Overview and functions of the LLC sublayer The *Logical Link Control Protocol for TETRA* (LLC) provides two types of connection at the TLA-SAP: a *Basic Link* (BL) and an *Advanced Link* (AL) (see Figure 6.14). Acknowledged or non-acknowledged data transmission is possible with both types of connection. A BL constitutes a bidirectional, connectionless path between one or more MSs and a BS, and is immediately available after the synchronization of MS and BS. In contrast, an AL is defined either as a bidirectional, connection-oriented path between an MS and a BS or as a unidirectional path from a BS to several MSs. An AL offers a better quality of service (QoS) than a BL and always requires that a connection be established. Compared with a BL, which only provides for the use of expanded error protection (*frame check sequence*, FCS) to minimize the number of undetected faulty messages, an AL offers a more favourable quality of service by providing flow control, segmentation of layer-2 service data units, a window mechanism that allows the transmission of more than one SDU without having to wait for an acknowledgement of the previous SDU, and the possibility to select different throughputs. FCS is always used with an AL.

The LLC sublayer of an MS supports simultaneously up to four ALs, each differentiated by its own *connection endpoint identifier* (CEP). Each of these CEP identifiers is associated with the time slots used by the MAC sublayer. For each AL and each circuit-switched service there is a BL that can use the respective number of reserved time slots of the corresponding ALs or line-switched services.

The transmission mode of an LLC connection is not contingent on the other LLC connections of an N-connection (N=network), e.g., in point-to-multipoint mode a sending MS can use an AL to the BS from one MS to several MSs, and conversely the BS can use unacknowledged BLs for the receiving MS.

Service elements of the TLA, TLB and TLC-SAP The TLA-SAP is used for addressed data transmission and for the control of data transmission procedures in layer 2. Each independent entity of a service is represented by a separate connection endpoint at the TLA-SAP. Up to four AL entities can be maintained by an MS at the same time. The information flow is possible from an MS to a BS and in the reverse as well as within an MS or a BS.

The service primitives listed in Table 6.9 are available at the TLA-SAP and are partially used by both connection types BL and AL or by only one of them. An acknowledged connection is set up and maintained through the four primitives of the TL_Connect-SP (SP=service primitive). The procedure is initiated with Request and indicated with Indication by the partner entity, which in turn sends back a Response using a Confirm to provide information about the connection setup to the initiating unit. The TL_Data-SP, which is defined separately for BL and AL, is available for the acknowledged transmission. For a BL there is also a Response-SP, which can be used to respond

Table 6.9: Service elements of the V+D TLA-SAP

Service element	Description	Link type
TL_Cancel_req	Request to cancel transmission before it has even started	BL, AL
TL_Connect_req/conf TL_Connect_ind/resp	Sending/receiving parameters required for setting up or maintaining a connection	AL
TL_Data_req/conf TL_Data_ind/resp	Sending and receiving data with acknowledgement	BL
TL_Data_req/conf TL_Data_ind	Sending and receiving data with acknowledgement	AL
TL_Disconnect_req/conf TL_Disconnect_ind	Termination of a connection between two corresponding AL entities	AL
TL_Release_req TL_Release_ind	Internal termination of a connection without informing the communicating partner	BL, AL
TL_Report_ind	Message to MLE about the status of an operation terminated by the MLE	BL, AL
TL_Unitdata_req/conf TL_Unitdata_ind	Sending and receiving data without acknowledgement	BL, AL

to a message with an acknowledgement or a new message and itself is not acknowledged. Non-acknowledged message transmission is carried out with a TL_Unitdata-SP. However, the Confirm-SP listed in Table 6.9 is optional and only indicates that transmission has been successful. The TL_Disconnect-SP is used to terminate an AL that has been initiated with Request and completed with Confirm, i.e., a confirmation from the communicating partner. A TL_Release-SP can be used to end an abnormally interrupted connection locally in the MS or the BS. A TL_Report_ind-SP is used internally for MLE information on the status of processes it has terminated.

The service elements at the TLB- and TLC-SAPs are summarized in Table 6.10. Using the TL_Sync- and TL_Sysinfo_req-SPs at the TLB-SAP of a BS, the MLE sublayer requests that the BS send the MS the synchronization and system information for the cell selection of the MS. The TL_Sync- and TL_Sysinfo_ind-SPs at the TLB-SAP of the MS are used in the MS to forward the data received from the LLC sublayer to the MLE sublayer.

The service elements of the TLC-SAP are responsible for monitoring a connection. Thus the TL_Configure-SP is used to configure layer 2 on the basis of the selected cell parameters and the state of the respective MS. The

Table 6.10: Service elements of V+D TLB- and TLC-SAPs

Service element	Description	SAP
TL_Sync_req/ind	Broadcast of synchronization parameters for cell selection	TLB
TL_Sysinfo_req/ind	Broadcast of system information for cell selection	
TL_Configure_req/conf	Configuration of layer 2 on basis of cell parameters provided	TLC
TL_Measurement_ind	Notification of measurement results on the quality of a connection	
TL_Monitor_ind	Notification of measurement results on the path loss of these channels	
TL_Monitor_List_req	Request to ascertain path losses of a specific number of channels	
TL_Report_ind	Report to MLE on the status of an action terminated by the MLE	
TL_Scan_req/conf	Scanning of a particular channel	
TL_Scan_Report_ind	Regular reporting on the signal quality of a channel and the current cell parameters after a channel has been scanned	
TL_Select_req/conf TL_Select_ind/resp	Selection of a specific channel for radio transmission	

TL_Measurement_ind-SP reports to the higher layers on the quality of a connection based on measurement results and the parameters of the neighbouring cells determined through *monitoring* or *scanning* (see Section 6.3.4.4). Through the TL_Monitor_List-SP, the monitoring process is applied to all the channels indicated in a list and the TL_Monitor-SP indicates the measurement results for a channel. The TL_Report-SP is used in the same way as at a TLA-SAP. The TL_Scan-SP starts the scanning process of a channel and the TL_Scan_Report-SP reports on the results. With Request and Confirm the TL_Select-SP is used to carry out a request from the MAC sublayer to the MLE sublayer to select a channel or to change channel, and through Indication and Response it is used in the notification of the MLE by the MAC sublayer of the channel change requested by the BS.

All the service primitives of the LLC and MAC sublayers and their interrelationships are listed in Table 6.11. Detailed explanations are available in [2, 5].

Data structures The *protocol data units* (PDU) of the LLC sublayer are called LLC frames and are defined according to the type of connection used: AL or BL.

Table 6.11: Relationship between V+D LLC and MAC service elements

LLC-SAP	LLC service element	MAC service element	MAC-SAP
TLA	TL_Connect_req/resp	TMA_Unitdata_req	TMA
	TL_Data_req/resp		
	TL_Disconnect_req		
	TL_Unitdata_req		
	TL_Connect_ind/conf	TMA_Unitdata_ind	
	TL_Data_ind/conf		
	TL_Disconnect_ind/conf		
	TL_Unitdata_ind		
	TL_Cancel_req	TMA_Cancel_req	
	TL_Release_req/ind	none	
	TL_Report_ind	TMA_Report_ind	
		TMA_Unitdata_ind	
TLB	TL_Sync_req/ind	TMB_Sync_req/ind	TMB
	TL_Sysinfo_req/ind	TMB_Sysinfo_req/ind	
TLC	TL_Configure_req/conf	TMC_Configure_req/conf	TMC
	TL_Measurement_ind	TMC_Measurement_ind	
	TL_Monitor_ind	TMC_Monitor_ind	
	TL_Monitor_List_req	TMC_Monitor_List_req	
	TL_Report_ind	TMC_Report_ind	
	TL_Scan_req/conf	TMC_Scan_req/conf	
	TL_Scan_Report_ind	TMC-Scan-Report_ind	
	TL_Select_req/conf/ind	TMC-Select_req/conf/ind	
		TMD-Report_ind	TMD
		TMD-Unitdata_req/ind	

Basic link With a BL the service data units of the *service primitives* (SP) received by the MLE are segmented not in the LLC sublayer but in the MAC sublayer, supplied with an LLC header and optionally with a check sum (FCS). Whereas an acknowledged service is only planned for point-to-point communication, the non-acknowledged service can also be used in a point-to-multipoint communication. The following PDUs have been specified for the BL:

BL_Adata This frame is used for connectionless transmission to confirm receipt of a frame and to send SDUs that arrive from the MLE and should be acknowledged. Another BL_Adata-PDU can only be sent again after the last PDU to be transmitted has been acknowledged through a BL_Adata-PDU or a BL_Ack-PDU.

BL_Data This frame is used for acknowledged, connectionless transmission, and contains the MLE's SDU which is to be transmitted. As with a

BL_Adata-PDU, a new BL_Data frame cannot be sent until the last one to be sent has been acknowledged.

BL_Udata This frame is used for non-acknowledged, connectionless transmission to send the data of the MLE. In this case several frames can be sent one after another because no acknowledgement is required.

BL_Ack This acknowledges the receipt of individual BL_Data-PDUs. If there is a message from the MLE, the BL_Ack frame can also contain layer-3 information.

Advanced link A layer-2 SDU can be longer with transmission over an AL than over a BL because the LLC sublayer offers AL segmentation as a service parameter. In contrast to the fragmentation in the MAC sublayer used with a BL, in an LLC segmentation each segment is assigned a number to enable the receiver to recognize the segment sequence. This numbering also allows a receiver to request a segment that has been lost, thereby eliminating the need for the complete message to be repeated again. The services of the BL are also available. The following PDUs have been defined for the AL:

AL_Setup The prerequisite for each AL is the establishment of a connection. This frame is used in setting up an AL for non-acknowledged as well as acknowledged transmission, and contains the respective QoS parameters. The called party confirms that a connection has been established and, in the case of acknowledged transmission, its acceptance of the QoS, also by means of an AL_Setup frame. If it cannot fulfil the QoS parameters, it sends its own QoS parameters, which must then be acknowledged by the calling party using a new AL_Setup.

AL_Data This frame is used for the acknowledged transmission of layer-3 information. It contains the respective segment of the TL-SDU, and all the segments except for the last one are transmitted with this PDU. The window process of an AL permits a number of TL-SDUs to be sent before confirmation of receipt of a complete TL-SDU has been received. However, the window is not used in flow control.

AL_Data-AR (*AR = acknowledge request*) This frame has the same significance as the AL_Data frame. In addition, the other side is requested to send back an acknowledgement immediately.

AL_Final This frame indicates to the partner entity that the segment contained in this frame is the last one of the TL-SDU being transmitted.

AL_Final_AR Similarly to AL_Final and AL_Data-AR, this frame contains the last segment being transmitted, and also requests an acknowledgement.

AL_Udata After a connection has been established with unacknowledged transmission, this PDU is used in the unidirectional exchange of information. Each segment of a TL-SDU, except for the last segment of the complete message, is sent with this PDU. The segments are numbered so that the receiver notices if a segment has been lost.

AL_Ufinal This frame transmits the last segment of a non-acknowledged transmission to the receiver. Thus it is only used as the conclusion of one or more AL_Udata.

AL_Ack If an acknowledgement has been accepted or the window has closed, this acknowledges all the Tl-SDU segments that have been received without errors. An AL_Ack frame therefore contains a request for selectively non-acknowledged segments to be resent.

AL_RNR (*RNR = receiver not ready*) The receiver can send this frame at any point in time to control the flow. This frame replaces an AL_Ack frame when a receiver at times is not able to receive new PDUs. However, the segments indicated as faulty can continue to be sent. New frames cannot be sent until an AL_Ack frame has been received or after the timer, which is started with the receipt of the AL_RNR frame, runs out.

AL_Disc An AL_Disc frame is used to terminate an AL in acknowledged or non-acknowledged transmission. In the case of acknowledged transmission the receiver also reponds with an AL_Disc frame.

State diagrams The states in the LLC sublayer are only explained superficially in the standard for a *basic link* and with inconsistencies in the one for an *advanced link*, making it difficult to provide reliable data with any assurance that it is correct.

Basic link For a basic link there are only two states for acknowledged data transmission in the LLC sublayer that can be entered after synchronization has taken place between MS and BS. The LLC sublayer is otherwise in `IDLE` state. The `TX-READY` state is one in which messages can be sent through a TL_Data_req-SP and any operations during transmission can be indicated through TL_Report_ind. Confirmation by the partner side is provided through a TL_Data_resp-SP.

The state `RX-READY` is entered by the receiving station; here too the TL_Report_ind-SP reports on the proceedings during the receiving process. The incoming message is indicated with a TL_Data_ind-SP and acknowledged with a TL_Data_conf-SP.

The standard does not include states for non-acknowledged data transfer.

Advanced link Figure 6.20 provides an overview of the states of the LLC sublayer with an advanced link, with explanations following.

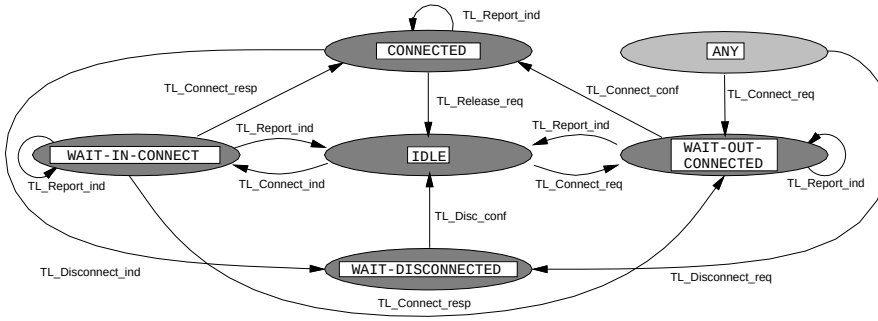


Figure 6.20: State diagram of an advanced link in the V+D LLC sublayer

With the TL_Connect_req-SP it is possible to change from any state into the `WAIT-OUT-CONNECT` state. An AL_Setup-PDU is generated that notifies the partner of the request for a connection. The proceedings of a connection establishment are indicated by a TL_Report_ind-SP. If the connection does not materialize, which is conveyed by different AL_Setup-PDUs and by the AL_Disc-PDU, there is a transition to state `IDLE`. If the connection is established successfully, the LLC sublayer moves to the `CONNECTED` state through a TL_Connect_conf-SP.

If an LLC sublayer in `IDLE` state receives a request to set up a connection from a TL_Connect_ind-SP, it changes to state `WAIT-IN-CONNECT`. If a connection is set up successfully, it moves from here to the state `CONNECTED` with a TL_Connect_resp-SP. If the connection is to be set up with other QoS parameters, a TL_Connect_resp-SP is generated with this information, a changeover is made to the `WAIT-OUT-CONNECTED` state, and the process is continued as described above. If a connection does not materialize, the TL_Disconnected_req-SP is used to move to state `WAIT-DISCONNECTED`. If access to the channel fails, the LLC sublayer reports this with a TL_Report_ind-SP and changes to the `IDLE` state.

If the sending LLC sublayer receives a TL_Disconnect_req-SP, a change is made from every state, except `IDLE`, to the `WAIT-DISCONNECTED` state through the dispatch of an AL_Disc-PDU. When a connection in `CONNECTED` state has been abnormally interrupted (indicated by a TL_Report_ind-SP), i.e., not initiated by the LLC sublayer, it moves into the `IDLE` state by means of an internal TL_Release_req-SP. The state `CONNECTED` in turn consists of a number of states; detailed information on this is given in [2, 5].

- `AL-TX-READY`: An MS is ready for the acknowledged transmission operation.
- `AL-RX-READY`: An MS is ready for the acknowledged receiving operation.
- `AL-UNACK-READY`: An MS is ready to receive or to send non-acknowledged data.

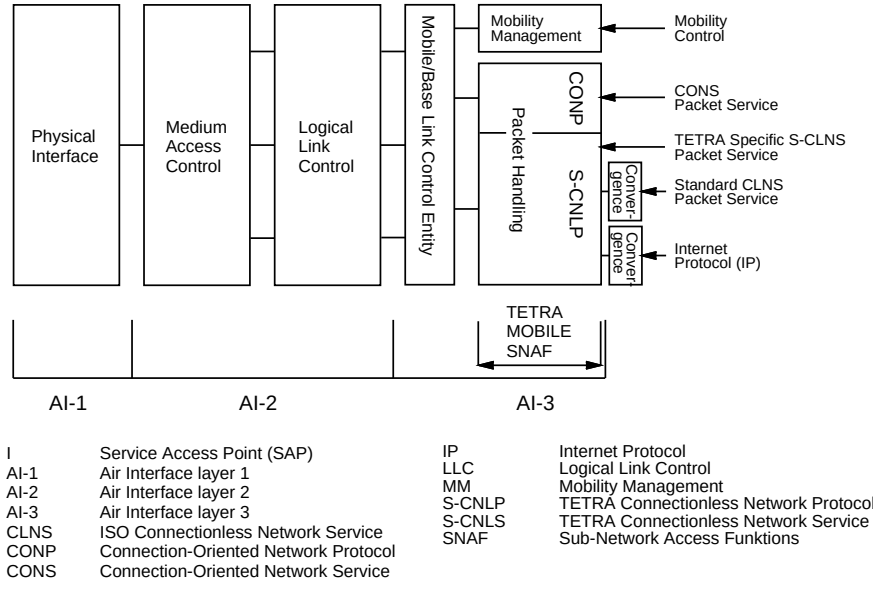


Figure 6.21: Packet Data Optimized (PDO) protocol stack

6.3.5 The Packet Data Optimized Protocol Stack

This is the first stack of protocols in a European standard to incorporate a concept for the statistical multiplexing of packets from a large number of data sources on radio channels under the control of the corresponding base station. This involves a period-oriented planning of the use of the transmission capacity of the individual logical channels in which each period during which the mobile stations are allowed to transmit on the uplink is explicitly identified on the downlink, and each period contains time slots for random access by the mobile stations that want to transmit during the next period. This concept is superior to circuit-oriented transmission when there is only very little data per transmission. The ETSI/HIPERLAN Type 1 standard contains a related concept (see Chapter 13); radio interfaces similar to TETRA-PDO are expected for the wireless-ATM standards ETSI/HIPERLAN Type 2/3 (see Chapter 12).

The PDO protocol stack consists of three layers: the *physical layer* (PL), which is the same for PDO and V+D, the sublayers *logical link control* (LLC) and *medium access control* (MAC) of the *data link layer* (DL), and the network and mobile station management services of the *network layer* (N) (see Figure 6.21).

The services of the network layer can be divided into connectionless and connection-oriented services: The connection-oriented service from TETRA is based on the CONS ISO-8348 standard. When it comes to the connectionless

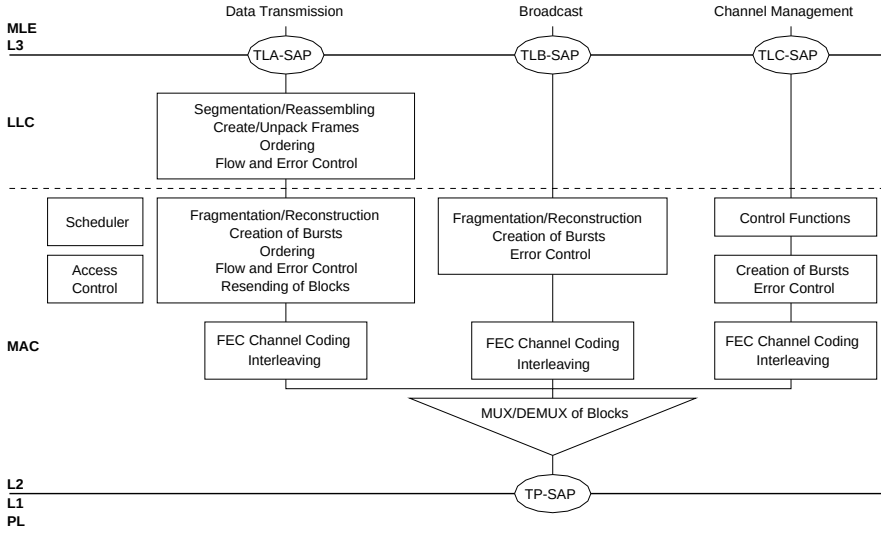


Figure 6.22: Architecture of the packet-data optimized physical layer

services, there is a differentiation between a TETRA *Specific Connectionless Network Service* (S-CLNS), the CLNS ISO-8348 and the connectionless service of the *Internet Protocol* (IP). The latter will be adapted to TETRA. The connectionless service in accordance with ISO or IP will be offered to enable interworking with X.25/X.75-packet-data networks or with the Internet.

Mobility Management (MM) will include registration of mobile stations with the base station and *fast call reestablishment*. The *mobile link entity* (MLE) represents the lowest sublayer of the network layer. The only way all information from the data link layer reaches the higher layers is through the MLE, thereby ensuring that the radio-related aspects of the data link and physical layers appear as transparent as possible. The tasks of the MLE include cell selection and the broadcast of system information.

The LLC and MAC sublayers of the physical layer are explained in detail below.

6.3.5.1 Architecture of the Physical Layer

As shown in Figure 6.22, the physical layer of the MLE offers its services to three different *service access points* (SAP): TETRA-LLC-A, TLB and TLC. The services required for data transmission are provided at the TLA-SAP. The TLB-SAP is only used for sending (base station) and receiving (mobile station) broadcast messages. The TLC-SAP, which is responsible for local control messages, in particular channel management functions, is only used on the mobile side.

Logical Link Control The LLC protocol is a layer-2 protocol for the PDO *air interface* (AI). Each layer-2 entity is used to connect a base station with a mobile station or a group of mobile stations. If a mobile station wants to establish connections with various base stations, the corresponding number of entities are required. According to Figure 6.22, the LLC sublayer does not add any further functionality to the services of the TLB and TLC-SAPs.

The three services—acknowledged and non-acknowledged point-to-point data transfer and non-acknowledged point-to-multipoint data transfer—are offered at the TLA-SAP. However, the latter service is only offered in the BS → MS direction, because a point-to-multipoint service in the other direction already exists in layer 3.

The LLC sublayer functions are described as a *Link Access Protocol for TETRA* (LAP.T). The protocol is only controlled one way (*unbalanced*); an entity of the protocol always offers only one service for a unidirectional transmission. It can be used on the uplink as well as on the downlink. Two entities are required for a bidirectional link.

The non-acknowledged and the acknowledged services offer the functions end-point addressing, segmentation, sequencing and, as an option, expanded error protection using a *frame check sequence* (FCS). Flow control is also being planned for the acknowledged service.

Data structures LAP.T uses protocol data units (PDU), which are specified as LLC frames. The *service data units* (SDU) of the *service primitives* (SP) received from the MLE are divided into various segments and mapped to the bodies of the logical frames (see Figure 6.23). The LLC bodies are reassembled into the right sequence for layer 3 by the LLC frames received. The parameters of the *interface control information* (ICI) of the LLC service element are filled together with their own LLC parameters in the header of each frame. If requested by the MLE, an FCS is added to the LLC body to provide expanded error protection.

The elements of an LLC frame are forwarded as a set of internal parameters to the MAC sublayer, or received by it, and appear later as explicit elements in one or more MAC-PDUs.

LAP.T is a simplified version of the ISDN Link Access Protocol LAPD. A simplified protocol was selected in order to reduce the protocol overhead and to avoid a repetition of functions of the MAC sublayer in the LLC sublayer.

The LLC sublayer recognizes the following frame types:

AI (*Acknowledged information*) AI-frames are only used in acknowledged transmission and only by the sending side. They include layer-3 information and are numbered in sequence. AI-frames are also used to set a connection to a defined initial state.

RR (*Receive ready*) RR-frames that do not contain any layer-3 information are used to indicate the receipt or the readiness to receive AI-frames. In

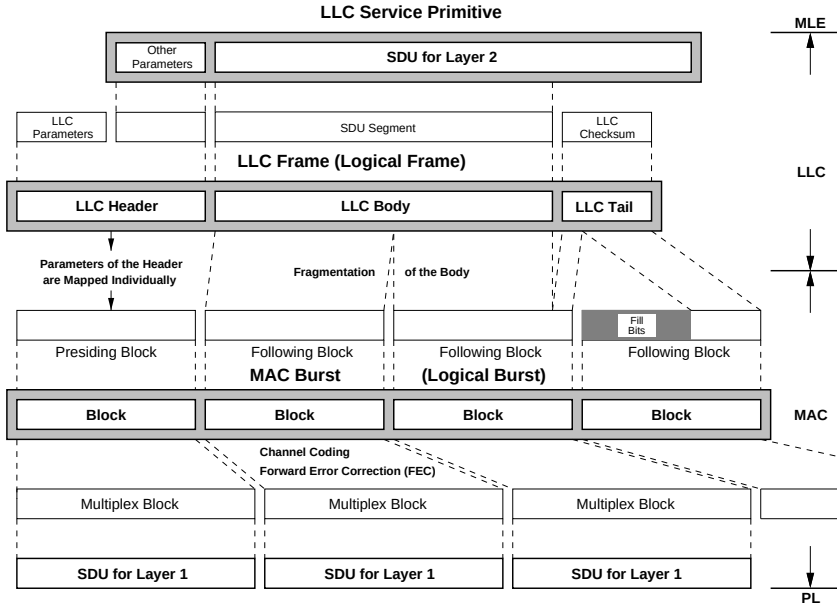


Figure 6.23: Data structures of the packet-data optimized data link layer

addition, the receiving side can use them to confirm or request that a connection to be reset.

UI (*Unacknowledged information*) UI-frames are used solely in non-acknowledged transmission and can only be requested by the MLE. They contain layer-3 information and can be lost if there is no notification from the sender.

Service elements of the TLA-SAP A TLA-SAP (see Figure 6.22) is used for addressed data transfer and the control of the data transfer procedures of layer 2. Each independent entity of a service is represented by a separate *connection end point* (CEP) at the TLA-SAP.

All TLA-SAP service elements correspond to unidirectional transactions (see Table 6.12). The TL_(TETRA LLC)_Connect service elements are used for setting up a unidirectional connection (BS → MS or MS → BS). A connection must be established before it can be used. The TL_Data service elements are used for all unidirectional data transfer over this connection. The service elements mentioned are required for acknowledged, connection-oriented point-to-point transmission. The TL_Unitdata service elements are used for connectionless, non-acknowledged point-to-point or point-to-multipoint transmission. Since no connection exists, they can be sent at any point in time.

Table 6.12: Service elements of the PDO-TLA-SAP

Service element	Description	SAP
TL_Connect_req	Request to layer 2 to establish or reestablish a connection for an acknowledged transmission.	TLA
TL_Connect_conf	Confirmation of the establishment of a connection.	
TL_Connect_ind	Report to layer 3 that a connection has been established.	
TL_Data_req	Request to layer 2 to transmit an acknowledged message.	TLA
TL_Data_conf	Report to layer 3 whether an acknowledged message has been successfully transmitted.	
TL_Data_ind	Forwarding of a received acknowledged message to layer 3.	
TL_Unitdata_req	Request to layer 2 to send a non-acknowledged message.	TLA
TL_Unitdata_ind	Forwarding of a received non-acknowledged message to layer 3.	

State diagram Four states are defined in LAP.T (see Figure 6.24), with `UNCONNECTED` representing the initial state of a LAP.T entity. The states only relate to acknowledged transmission; non-acknowledged transmission is permitted in any state. Acknowledged transmission only takes place in state `CONNECTED`, whereas the other states are part of the connection establishment process.

Connection establishment A connection is established by the sender when a TL_Connect_req service element is received. An AI-frame with a request to set up a connection is sent, followed by a transition to `CONNECT EXPECTED` (1) (see Figure 6.24 and Table 6.13). If a time-out occurs, the sender returns to the initial state (2), and the MLE is notified through a negative TL_Connect_conf-SP. The receiver responds to this AI-frame with an RR-frame, informs the MLE with a TL_Connect_ind-SP and moves into state `CONNECTED` (3).

This RR-frame also results in the sender moving into the state `CONNECTED` (4) and the sending of a positive TL_Connect_conf-SP. In the event that a reset of the connection is initiated through an RR-frame from the receiver or through a TL_Connect_req-SP and non-acknowledged AI-frames are still outstanding, the sender moves into the state `CONNECT PENDING` (5, 6, 7). The transition 6 from L3 to L2 can also be triggered through another TL_Data_req-SP. A new AI-frame is always sent that either confirms (RR-frame) or requests (TL_Connect_req, TL_Data_req) the resetting of a connection.

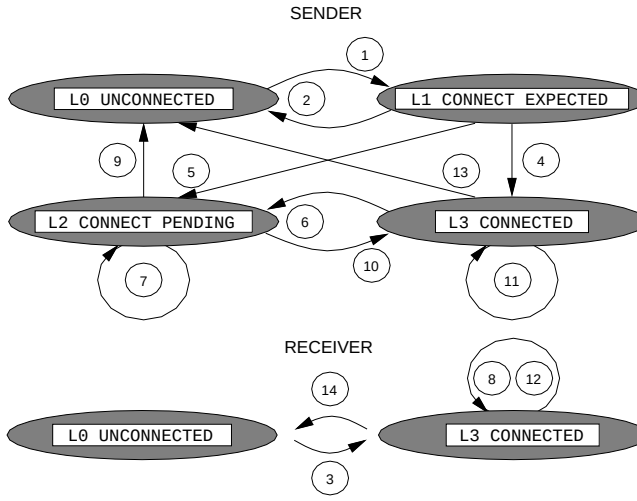


Figure 6.24: LAP.T state transitions

The receiver acknowledges a reset request with an RR-frame (8) and sends a TL_Connect_ind-SP to the MLE. If the sender does not receive a confirmation (time-out), it moves into the initial state `UNCONNECTED` (9) and informs the MLE accordingly through a negative TL_Data_conf- or TL_Connect_conf-SP. Otherwise it returns to the state `CONNECTED` (10) and, if applicable, sends a positive TL_Connect_conf-SP.

Acknowledged data transmission Requested by a TL_Data_req-SP, the sending side sends segmented layer-3 SDUs in several AI-frames (11), which are acknowledged by the receiving side with RR-frames (12). A window of the maximum size 3 is used for flow control. Layer-3 SDUs that are received in their entirety are indicated to the MLE by a TL_Data_ind-SP. On the sending side the success or failure of a transmission is conveyed to the MLE through a TL_Data_conf-SP. If the time out for an AI or RR-frame is expired or if the last frame of a transmission has been received or acknowledged, the respective LAP.T entity returns to the initial state in (13, 14).

Medium Access Control For the broadcast and local channel management functions, the MAC and the MLE exchange information over the TLB-SAP and the TLC-SAP. No formal service boundary has been specified between the LLC and MAC sublayers; neither SPs nor SAPs have been defined for this. The MAC sublayer receives LLC frames consisting of header, body and tail (see Figure 6.23). These three parts are transmitted in logical MAC bursts (MAC-PDUs). A MAC burst consists of a starting block, called a *presiding block*, and between 0 and 40 *following blocks*, with the parameters of the LLC header mapped onto the presiding block, and those of the LLC

Table 6.13: LAP.T state transitions

	Sender
1	TL_Connect_req from MLE → AI-frame to receiver
2	No RR-frame received (time-out) ⇒ negative TL_Connect_conf to MLE
4	RR-frame received ⇒ positive TL_Connect_conf to MLE
5, 6, 7	TL_Connect_req or further TL_Data_req (only 6) from MLE or reset request from receiver (RR-frame) ⇒ AI-frame with reset request /confirmation to receiver
9	No RR-frame received ⇒ negative TL_Data_conf or TL_Connect_conf to MLE
10	RR-frame received ⇒ positive TL_Connect_conf to MLE or start transmission
11	TL_Data_req from MLE ⇒ data transfer with flow control
13	Confirmation of last AI-frame or no RR-frame received (time-out) ⇒ positive or negative TL_Data_conf to MLE
	Receiver
3	AI-frame received ⇒ confirmation (RR-frame) to sender, TL_Connect_ind at MLE
8	Confirmation of reset request or reset request (RR-frame) ⇒ TL_Connect_ind to MLE
12	Confirmation of received data (RR-frame)
14	Final confirmation sent ⇒ TL_Data_ind to MLE or no AI-frame received (time-out)

body and the LLC tail mapped onto the following blocks. This corresponds to continuous radio transmission. Depending on the type of MAC burst, the individual blocks of a MAC burst are allocated to channel coding over two logical channels: a *Master Block Channel* (MBCH) and a *Normal Block CHannel* (NBCH). Depending on the logical channel, the blocks are forward error correction (FEC) coded and are then forwarded as SDUs to the physical layer.

A transaction is a connected sequence of radio transmissions, and corresponds to the transmission of an LLC frame. Consecutive transactions on the uplink can be linked together, thereby cancelling out the need for repeated random access for each additional LLC frame. The transmission of a single MAC burst is considered the smallest possible transaction. However, long transactions occur when several bursts are necessary either because of the segmentation of the layer-2 SDU or because of the retransmission of defectively transmitted blocks. Layer-2 24-bit wide addresses (*Short Subscriber Identity*, `SSI`) are used for each transaction. The BS reduces the address overhead by supplying 10-bit wide *event labels*, which are used in the respective MAC-bursts instead of the `SSI`. An event label is only valid for one transaction.

Table 6.14: Service elements of the PDO TLB and TLC-SAPs

Service element	Description	SAP
TL_Broadcast-1_req	Request to transmit an SIN1-PDU. The transmission is repeated at regular intervals until a new request has been received.	TLB
TL_Broadcast-1_conf	Confirmation of the processing of an SIN1-PDU.	
TL_Broadcast-1_ind	Forwarding of a received SIN1-PDU (MS).	
TL_Broadcast-2_req	Request to send an SIN2-PDU.	
TL_Broadcast-2_conf	Confirmation of the processing of an SIN2-PDU.	
TL_Broadcast-2_ind	Forwarding of a received SIN2-PDU (MS).	
TL_Scan_req TL_Scan_conf	Request on the part of MS to scan a radio channel and to report the measurement results to layer 3.	TLC
TL_Serving_ind	Report of the measurement results of the current cell.	
TL_Monitor_req TL_Monitor_ind	Request on the part of MS to check the signal quality of a list of radio channels and to report the measurement results to layer 3.	
TL_Select_req TL_Select_conf	Request on the part of MS to select a defined radio channel and to confirm the selection.	
TL_Addlist_req	Request on the part of MS to use the addresses supplied for wake-up control.	

A base station can use a number of uplink and downlink carrier frequencies, thereby operating in asymmetric mode, i.e., with an uneven number of uplink and downlink carrier frequencies.

Service elements of the TLB TLC-SAPs On the BS side, system information is forwarded over the TLB-SAP from the MLE to the MAC sublayer using the service elements TL_Broadcast-1 and TL_Broadcast-2, which are sent as broadcast information (*System Information Type 1* (SIN1)- or SIN2-PDU). On the MS side, the information contained in these MAC bursts is sent to the MLE (see Table 6.14).

The MLE and the MAC sublayers of the mobile station exchange channel management information over the TLC-SAP. The TL_Scan-SPs are used to scan of the radio channels and report on the measurement results. The MAC sublayer regularly informs the MLE of the measurement results of the current cell (TL_Serving_ind). If the MLE is interested in the signal quality of other radio channels, it forwards an appropriate list to the MAC sublayer and then receives the signal quality measurements of the respective radio channels (TL_Monitor) in return. If the MLE has selected a particular radio channel, it notifies the MAC sublayer, which reports back with the result of the channel selection (TL_Select).

Table 6.15: Mapping of PDO MAC-PDUs onto the logical channels

PDU	Internal logical channel	Logical channel for channel coding	Direction
SIN1	MCCH	MBCH	Downlink
SIN2 AP	MCCH	NBCH	Downlink
AA WU DD1	ACCH	NBCH	Downlink
DD2 DR1 DR2 DR3	DTCH	NBCH	Downlink
UD1	RACH	NBCH	Uplink
UD2 UR	UTCH	NBCH	Uplink

An MS can be operated in three different modes: *normal mode* (NOR), *low-duty mode* (LOD) or *very low-duty mode* (VLD). In NOR-mode it continuously listens in on the BS. The only difference between the two modes is the length of the time intervals after which the MS again listens in on the BS. These intervals are defined in a Wake-up (WU)-PDU regularly sent by the BS. When an MS wants to transmit, it immediately moves into NOR-mode. Using the TL_Addlist_req-SP, the MLE informs the MAC sublayer to which MAC addresses listed in the WU-PDUs it is to respond.

Internal logical channels The internal logical channels of the MAC sublayer are only apparent within this sublayer and always responsible for a certain type of information. Three logical channels have been defined for the downlink and two for the uplink (see Table 6.15):

MCCH (*Master Control Channel*) This downlink channel transmits system information. Each MS first listens in on this channel when it finds itself in the area of a new base station. The MCCH also guides the MS to the ACCH.

ACCH (*Access Control Channel*) This downlink channel contains all the access control information, which can be subdivided into uplink information for random access and downlink control information for the initiation of downlink data transfer and the sending of wake-up messages.

DTCH (*Downlink Traffic Channel*) Downlink data and responses to uplink data are sent over this downlink channel.

RACH (*Random Access CHannel*) Each non-reserved uplink data transmission is carried out over this uplink channel.

UTCH (*Uplink Traffic Channel*) Uplink data and responses to downlink data are reserved for transmission over this uplink channel.

Data structures The following PDUs are defined for the MAC sublayer:

UR (*Uplink Response*) PDU for responses to downlink data.

UD1 (*Uplink Data Type 1*) First PDU for random access when no *Uplink Transfer Event Label* (UT-label) has yet been assigned.

UD2 (*Uplink Data Type 2*) If a UT-label has been assigned, the MS uses this PDU for all other reserved transmissions of this transaction.

DR1 (*Downlink Response Type 1*) This PDU is used as a response to UD1-PDUs when no UT-label has yet been reserved.

DR2 (*Downlink Response Type 2*) This PDU is sent as a response to UD2-PDUs.

DR3 (*Downlink Response Type 3*) Negative confirmation and request for renewed sending of defective blocks when not all blocks of a UD1-PDU have been received.

AA (*Access Announce*) PDU for the definition of the beginning and end and other parameters of random access that change on a short-term basis.

DD1 (*Downlink Data Type 1*) A BS uses DD1 as the first PDU for data transmission when no *Downlink Transfer Event Label* (DT-label) has yet been assigned.

DD2 (*Downlink Data Type 2*) PDU for all further downlink data transmissions of a transaction.

WU (*Wake-up*) This PDU defines the time intervals that LOD and VLD MSs are permitted to maintain in standby state in order to save battery power.

AP (*Access Parameters*) Random-access parameters that change on a long-term basis are defined in this PDU.

SIN1 (*System INformation Type 1*) Uncoded system information is regularly sent per broadcast from the BS using this PDU.

SIN2 (*System INformation Type 2*) Additional system information is sent coded from the BS using this PDU.

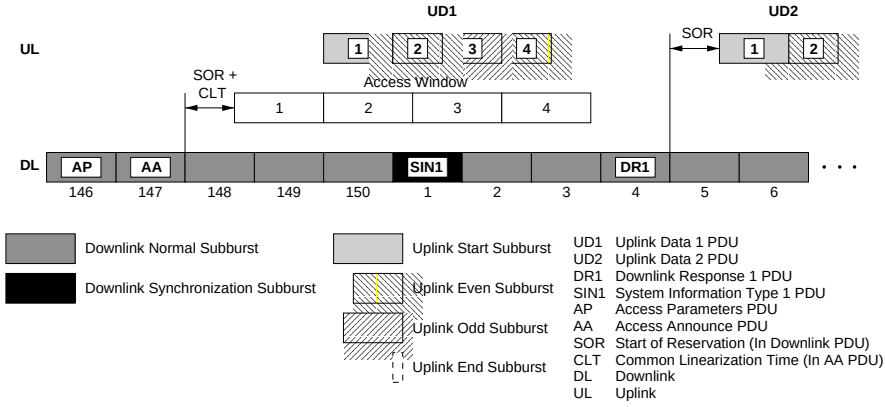


Figure 6.25: PDO random-access protocol

Table 6.15 shows the relationship between PDUs of the MAC sublayer, the internal logical channels and the logical channels over which blocks requiring coding are transmitted. The mapping takes place from left to right; e.g., the SIN1-PDU is mapped onto the internal logical channel MCCH, and this in turn to the logical channel MBCH. The coded blocks are then transmitted over the downlink.

Random-access protocol As can be seen in Figure 6.25, the downlink is divided into 150 subbursts, with an SIN1-PDU always sent in the first subburst. At longer intervals the BS sends an AP-PDU in which only those parameters of the random-access protocol are defined that are changeable in the long-term. Whether or not the *busy flag* is set depends on the training sequence used in the subbursts on the downlink (see Section 6.3.5.2). The AP-PDU specifies whether this busy flag should be heeded, and consequently which procedure is being used: random access DSMA (busy flag used) or *S-Aloha* with packet reservation (busy flag not used).

The period of time in which random access is allowed is called an access window. An access window is divided into several access periods (e.g., four such as in Section 6.3.5.2), with each *access period length* fixed in the AP-PDU. An AA-PDU defines the start of an access window, supplying the parameters for the duration of the linearization of the MSs (*common linearization time*, CLT) and an additional offset (*start of reservation*, SOR), and the end through stipulation of the number of access periods.

An MS that is ready to transmit selects an access period from a uniform distribution in the access window if the access window is still closed, i.e., has not yet begun, and the priority allocated by it for this transaction is not lower than the minimum priority allowed for this access window. If an MS decides to transmit when an access window is open, it selects the next immediate access period in compliance with the priority conditions. If the requests to

send also occur in a uniform distribution during an open access window then the selection of the next immediate access period is justified. With DSMA an access window is always considered closed if a busy flag is indicated. A BS sets a busy flag in order to reduce competiton and throughput as soon as it recognizes that an MS is sending in a defined access period.

Once an MS has successfully selected an access period, it may, depending on the Max_data parameter of the AP-PDU that defines the number of maximum allowable blocks of a UD1-PDU, not only send its reservation request for the uplink but also transmit data (Max data > 1). An MS always starts to send at the beginning of an access period; however, there is no direct relationship between the length of an access period and the length of a UD1-PDU. In fact, a UD1-PDU may even extend into other access periods or into the reserved uplink traffic of other MSs. It is up to the BS or the operator to determine sensible parameters.

The BS responds to a UD1-PDU with a DR1-PDU if all blocks have been received successfully or with a DR3-PDU, which contains a list of the blocks received with errors, if only some of the blocks have been received successfully. In the event that not all data has yet been sent with the UD1-PDU or the UD1-PDU has only been partially received, reserved uplink transfer with the use of UD2-PDUs takes place until the end of the transaction.

On the basis of the `Retry-Delay`-parameters of the AA-PDU, the respective MS establishes how many AA-PDUs it should expect before it is allowed to repeat its last not yet confirmed random access. The `Max-Access-Retries`-parameter of the AP-PDU stipulates how many access retries can be carried out.

State Diagrams and Transitions

Mobile station The MAC protocol of a mobile station is in one out of eight states (see Figure 6.26), of which `M0 START` is the initial state that occurs when the MS is switched on. The state transitions of a mobile station in the MAC sublayer are summarized in Table 6.16.

System information in the form of the SIN1 and SIN2-PDU (1, 20) must first be received before an MS is allowed, after receiving the AP-PDU (2), which contains only those parameters for random access that are changeable in the long term, to move into the state `M1 AWAKE`. If an MS is to be operated in LOD or VLD-mode, it moves into state `M2 STANDBY` (6) after it has received a WU-PDU. The WU-PDU defines when the MS should „wake-up“ again and whether downlink data transmission is being planned for it at a particular time. If the time for receiving a new WU-PDU has occurred (7) or downlink data transmission is starting, the MS returns to the state `M1 AWAKE`.

The three states M3 to M5 are assigned to uplink transmission. If the LLC receives a TL_Connect_req or TL_DAta_req-SP from the MLE at the TLA-SAP and subsequently delivers a frame to the MAC sublayer, this changes depending on whether an open access window exists (3) or not (9) in the

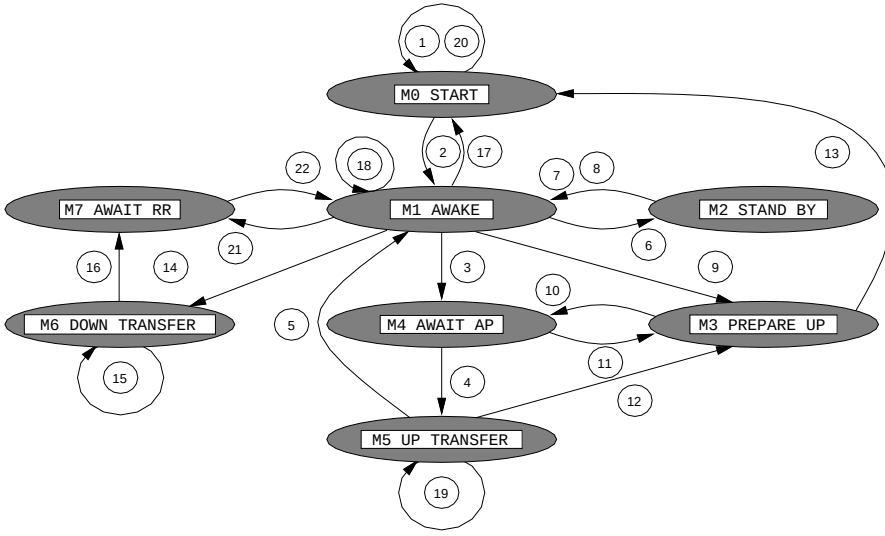


Figure 6.26: State diagram of a PDO mobile station in the MAC sublayer

Table 6.16: State transitions of a PDO mobile station in the MAC sublayer

1	The MS receives an SIN1-PDU.
2	The MS receives an AP-PDU.
3	The MS receives a TL_Connect_req or TL_Data_req service element and an open access window exists.
4	The selected access period begins.
5	The MS receives the final confirmation for this transaction.
6	An LOD or VLD-MS receives a WU-PDU.
7	The time for receiving a new WU-PDU begins.
8	The time defined in a WU-PDU for receiving downlink data begins.
9	The MS receives a TL_Connect_req/TL_Data_req service element and there is no open access window.
10	The MS receives an AA-PDU.
11	The access window is closed.
12	Time-out. No response to uplink data.
13	Time-out. No AA-PDU has been received within last 5 s.
14	The MS receives a DD1-PDU.
15	The MS receives a DD2-PDU.
16	The MS receives the last DD2-PDU for this transaction.
17	Time-out. No AA or data PDU received within last 5 s.
18	The MS receives an AA-PDU.
19	The MS receives a DR1, DR2 or DR3-PDU and has additional reserved data to send.
20	The MS receives an SIN2-PDU.
21	The MS receives a DD1-PDU that is the last one for this transaction.
22	The MAC sublayer receives an RR-frame from the LLC sublayer.

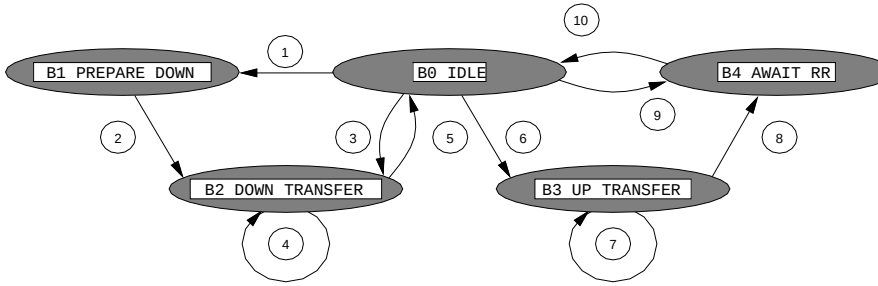


Figure 6.27: State diagram of a PDO base station in the MAC sublayer

state `M4 AWAIT AP` or `M3 PREPARE UP`. If the MS finds itself in the latter state and receives an AA-PDU, it changes to the state `M4 AWAIT AP` (10), returning immediately (11) if the window is already closed or the random-access protocol has not selected an access period for the window. If no AA-PDU is received (time-out) within a period of 5 s, the MS again moves to the initial state `M0 START` (13). If, however, the MS has reached the state `M4 AWAIT AP` and is starting the selected access period, it carries through its random access, sends a UD1-PDU and moves into state `M5 UP TRANSFER`. The MS remains in this state as long as it is still waiting for responses (DR1, DR2 or DR3-PDU) and has further data to send (19). Uplink transmission is completed when the last confirmation has been received (5). However, if there is still a confirmation that has not been received, the MS attempts to carry out a new random access (12). The MAC sublayer informs the LLC sublayer of the success of the entire transaction.

The two last states M6 and M7 are reserved for downlink transmission. Upon receipt of a DD1-PDU and depending on whether the PDU received is the only transaction (14) or other DD2-PDUs are following (21), the MS moves from state `M1 AWAKE` to one of the two states `M6 DOWN TRANSFER` or `M7 AWAIT RR`. As long as the last confirmation is not yet being sent, the MS remains in the M6 state and acknowledges incoming DD1 and DD2-PDUs (15). However, as soon as the last DD2-PDU for the transaction is received, it moves to state `M7 AWAIT RR` (7), from which the LLC requests an RR-frame and upon receipt of the frame sends it in a UR-PDU. Upon completion of the downlink transaction (22), the MS returns to the state `M1 AWAKE`.

In this state it processes the incoming AA-PDUs (18), and then returns to the initial state (17) if an AA-PDU has not been received for 5 s or an uplink or downlink transaction has not begun.

Base station Five states with ten state transitions are defined for the MAC protocol of a BS (see Figure 6.27 and Table 6.17), with `B0 IDLE` representing the initial state after a BS has been switched on. Since the BS produces its

Table 6.17: State transitions of a PDO base station in the MAC sublayer

1	The BS receives a TL_Connect_req or TL_Data_req service element and the addressed MS is in LOD or VLD-mode.
2	Time-out. The time for wake-up defined in the WU-PDU occurs.
3	The BS receives a TL_Connect_req or TL_Data_req service element and the addressed MS is in NOR-mode.
4	The BS receives a UR-PDU that is not the last one for this transaction.
5	The BS receives the last UR-PDU for this transaction.
6	The BS receives a UD1-PDU that is not the last one for this transaction.
7	The BS receives a DD2-PDU that is not the last one for this transaction.
8	The BS receives the last UD2-PDU for this transaction.
9	The BS receives the last UD1-PDU for this transaction.
10	The MAC sublayer receives an RR-frame from the LLC sublayer.

own entity of the MAC protocol entity for each connection, it can repeatedly enter the states B0 to B4 at the same time.

The states B1 and B2 are assigned to downlink data transmision. If the LLC sublayer receives a TL_Connect_req or TL_Data_req-SP from the MLE at the TLA-SAP and subsequently delivers a frame to the MAC sublayer, the BS changes either to `B1 PREPARE DOWN` or to `B2 DOWN TRANSFER`, depending on whether the addressed MS is operating in (1) VLD, LOD or (3) NOR-mode. During the transition to B1 state, the BS sends a WU-PDU in which it defines the starting time for downlink transmission to the addressed MS.

When this time comes, a DD1-PDU is sent to the addressed MS and a transition (2) to state `B2 DOWN TRANSFER` takes place. If the deviation over the wake-up of the MS was not necessary (3), the DD1-PDU is sent immediately. So long as further data is to be sent with DD2-PDUs and the last confirmation (UR-PDU) has still not arrived, the BS remains in the state `B2 DOWN TRANSFER`. It does not return (5) to the initial state until the downlink transaction has been completed, i.e., the last UR-PDU has been received.

The states `B3` and `B4` have been defined for uplink transmission. If the BS receives a UD1-PDU that contains a reservation request for further uplink transmission or the UD1-PDU has only been partially received, it sends a DR1 or DR3-PDU supplied with a reservation and moves (6) into the state `B3 UP TRANSFER`. So long as the last UD2-PDU has still not been received, the BS responds with normal DR2-PDUs and remains (7) in state `B3`.

After it has received the final UD2-PDU, the MAC sublayer requests (8) an RR-frame from the LLC sublayer, waits for it in state `B4 AWAIT RR`, responds to the MS with one of the DR2-PDUs contained in the RR-frame, and subsequently returns (10) to the initial state `B0 IDLE`. In the event that an uplink transaction consists of only one UD1-PDU received, the MAC sublayer immediately requests an RR-frame from the LLC sublayer, and moves (9) from state `B0 IDLE` to state `B4 AWAIT RR`.

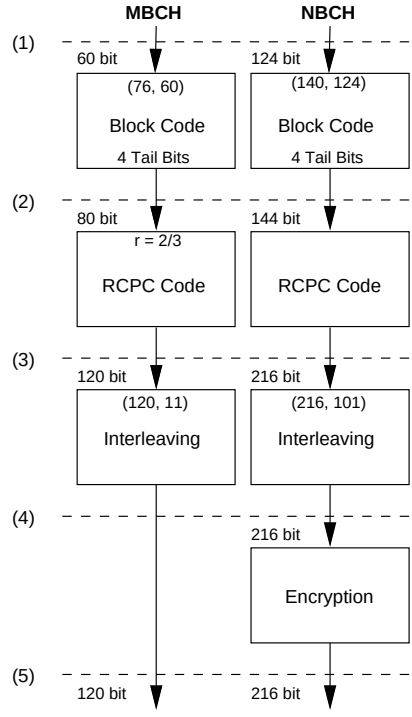


Figure 6.28: Structure of PDO channel coding

Channel coding Figure 6.28 presents the four-level structure of PDO channel coding. All MAC-PDUs, except for the SIN1-PDU, are 124 bits wide and, as shown in Table 6.15, are mapped onto the logical channel NBCH. Of the 124 bits of an NBCH block, there are 4 bits that are reserved and also contain information identifying whether the block is a presiding block or a following block. A following block can therefore contain 8 bytes of user data. The coded and encrypted multiplex blocks, which are 216 bits wide, are forwarded as SDUs to the physical layer. An SIN1-PDU is only 60 bits wide, is mapped onto the MBCH channel, and is not encrypted because new MSs first listen in on the MBCH and at this point in time no agreements on coding can yet exist. These 120-bit wide multiplex blocks are also forwarded to layer 1.

6.3.5.2 Burst Structure

A burst is a data set of a certain length that is modulated onto a carrier frequency and consists of a number of subbursts.

Downlink On the downlink the BS continuously sends 240-bit long subbursts, of which each 150th one is a *downlink synchronization subburst* con-

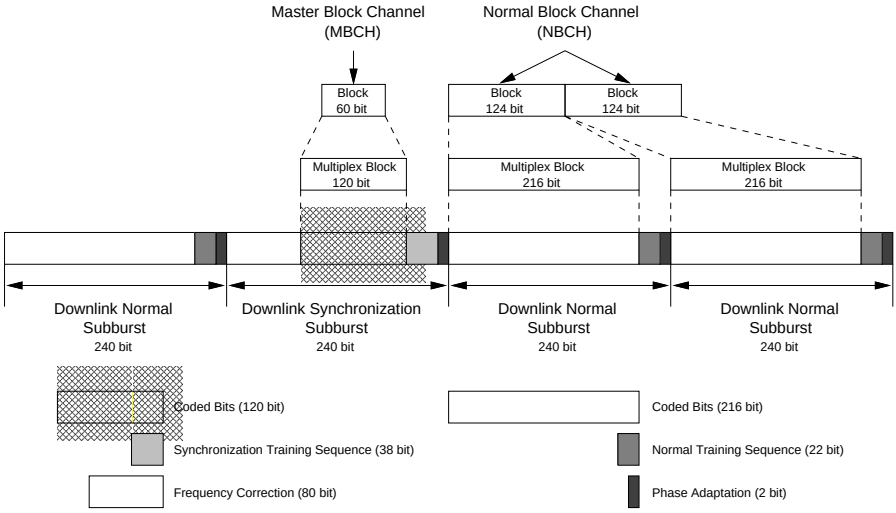


Figure 6.29: PDO downlink bursts

taining the multiplex blocks of the MBCH. Otherwise the multiplex blocks of the NBCH are transmitted in *downlink normal subbursts* (see Figure 6.29). A modulation bit has a duration of 1/36 ms $\sim$ 27.78 µs. From this it can be derived that a downlink subburst takes 6.67 ms and a downlink synchronization subburst can be sent every 1.00 s.

Uplink If a MAC burst consists of N blocks then it is transmitted on the uplink in an *uplink start subburst* (250 bits $\widehat{=}$ 6.94 ms), $\lfloor N/2 \rfloor$ *uplink even subburst* (238 bit $\widehat{=}$ 6.61 ms) and $\lfloor (N-1)/2 \rfloor$ *uplink odd subburst* (216 bits $\widehat{=}$ 6.00 ms) (see Figure 6.30). An uplink burst is completed with an *uplink end subburst* (4 bits $\widehat{=}$ 111.1 µs).

Since the uplink is operated with a time variance, the PDO standard has no synchronous time channels for the uplink. The start of a reservation on the uplink refers to the end of the downlink normal subburst that contains the reservation. The number of modulation bits expected is indicated by the transmitted parameter *Start of Reservation* (see Figure 6.25).

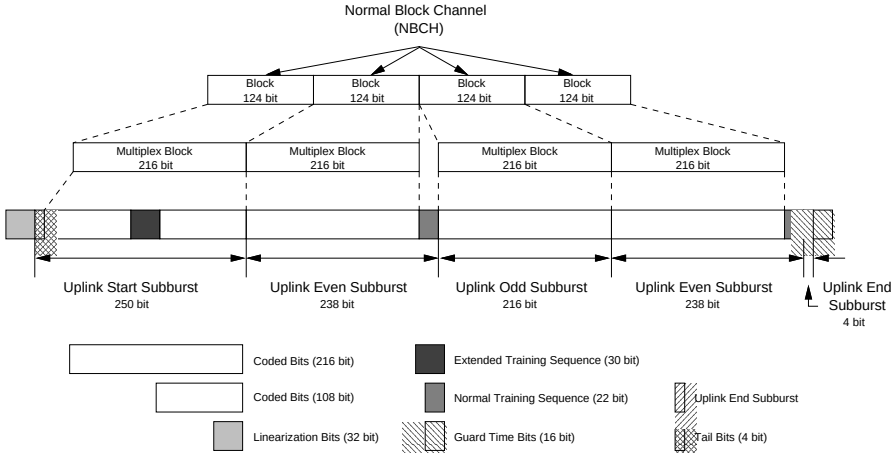


Figure 6.30: PDO uplink bursts

6.3.6 List of Abbreviations for Trunked Radio

AA	Access Announce PDU (PDO)
AACH	Access Assignment CHannel (V+D)
ACB	Access Control Block (V+D)
ACCH	Access Control CHannel (PDO)
AI	Air Interface
AI	Acknowledge Information frame (PDO)
AL	Advanced Link (V+D)
AP	Access Parameter PDU (PDO)
APL	Access Period Length (PDO)
ARQ	Automatic Repeat reQuest
BCCH	Broadcast Control CHannel (V+D)
BCH	Bose–Chaudhuri–Hocquenghem
BER	Bit-Error Ratio
BKN	BlocK Number (V+D)
BL	Basic Link (V+D)
BLCH	Base Station Linearization CHannel (V+D)
BLE	Base Link Entity
BLER	BLock-Error Ratio
BNCH	Broadcast Network CHannel (V+D)
BS	Base Station
BSCH	Broadcast Synchronization CHannel (V+D)
BU	Bad Urban
CB	Control Uplink Burst (V+D)
CCH	Control CHannel (V+D)
CEP	Connection End Point

CLCH	Common Linearization CHannel (V+D)
CLNP	ConnectionLess Network Protocol
CLNS	ConnectionLess Network Service
CLT	Common Linearization Time
CMCE	Circuit Mode Control Entity (V+D)
CNCL	Communication Networks Class Library
CONP	Connection-Oriented Network Protocol
CONS	Connection-Oriented Network Service
CP	Control Physical channel (V+D)
CRC	Cyclic Redundancy Check
DDx	Downlink Data type x PDU (PDO)
DM	Direct Mode
DQPSK	Differential Quaternary Phase-Shift Keying
DRx	Downlink Response type x PDU (PDO)
DSMA	Data Sense Multiple Access
DTCH	Downlink Traffic CHannel (PDO)
EL	Event Label
ETSI	European Telecommunications Standards Institute
FCFS	First Come First Serve
FCS	Frame Check Sequence
FEC	Forward Error Correction
FN	Frame Number (V+D)
FSN	Frame Sequence Number (PDO)
HT	Hilly Terrain
ICI	Interface Control Information
ISDN	Integrated Services Digital Network
IP	Internet Protocol
L1	Layer 1
L2	Layer 2
L3	Layer 3
LAP.T	Link Access Protocol for TETRA
LCH	Linearization CHannel (V+D)
LLC	Logical Link Control
LOD	LOw Duty mode (PDO)
MAC	Medium-Access Control
MBCH	Master Block CHannel (PDO)
MCCH	Master Control CHannel (PDO)
MD	Max Data (PDO)
MLE	Mobile Link Entity
MS	Mobile Station
MM	Mobility Management
MN	Multiframe Number (V+D)
N	Network layer
NAP	Number of Access Periods (PDO)
NBCH	Normal Block CHannel (PDO)

NDB	Normal Downlink Burst (V+D)
NOR	NORmal mode (PDO)
NUB	Normal Uplink Burst (V+D)
OSI	Open Systems Interconnection
PBX	Private Branch eXchange
PCI	Protocol Control Information
PDN	Public Data Network
PDO	Packet Data Optimized
PDU	Protocol Data Unit
PL	Physical Layer
PSTN	Public Switched Telephone Network
PTMR	Private Trunked Mobile Radio
QoS	Quality of Service
RA	Rural Area
RACH	Random Access CHannel (PDO)
RCPC	Rate-Compatible Punctured Convolutional code
RD	Retry Delay (PDO)
RES	Radio Equipment and Systems
RR	Receiver Ready frame (PDO)
S-CLNP	TETRA Specific ConnectionLess Network Protocol
S-CLNS	TETRA Specific ConnectionLess Network Service
SAP	Service Access Point
SB	Synchronization Downlink Burst (V+D)
SCH	Signalling CHannel (V+D)
SDL	Specification and Description Language
SDS	Short Data Service
SDU	Service Data Unit
SINx	System Information type x PDU (PDO)
SJN	Shortest Job Next
SNAF	Sub-Network Access Functions
SOR	Start Of Reservation
SP	Service Primitive
SSI	Short Subscriber Identity
SSN	Segment Sequence Number (PDO)
SSN	SubSlot Number (V+D)
STC	Standardization Technical Committee
STCH	STealing CHannel (V+D)
TCH	Traffic CHannel (V+D)
TDM	Time-Division Multiplexing
TDMA	Time-Division Multiple Access
TETRA	TransEuropean Trunked RAdio
TLx-SAP	TETRA LLC type x SAP
TMx-SAP	TETRA MAC type x SAP
TN	Timeslot Number (V+D)
TP	Traffic Physical channel (V+D)

TU	Typical Urban
UDx	Uplink Data type x PDU (PDO)
UI	Unacknowledged Information (PDO)
UP	Unallocated Physical channel (V+D)
UR	Uplink Response PDU (PDO)
UTCH	Uplink Traffic CHannel (PDO)
VLD	Very Low-Duty mode (PDO)
WU	Wake-Up PDU (PDO)
V+D	Voice plus Data

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7
Paging Systems

It is often important to be able to reach certain people very quickly. The conventional telephone network is not always optimal, because a line may be engaged or the person being called might not be available. Mobile telephone systems offer a high degree of reachability of a mobile subscriber, but MSs are not always switched on and are expensive to use.

Paging systems fill a particular gap (see Table 7.1). They allow unidirectional transmission of information in the form of a tone or a numeric or alphanumeric message to the person being contacted, whose location area is not known (see Table 7.2). A terminal is required that is constantly ready to receive but cannot transmit, and consequently is small, lightweight and inexpensive.

A paging message is automatically initiated when a telephone user dials the paging service and, using a telephone keyboard, an Internet terminal or a PC, conveys the pager number of the called person or a short message for it to the computer that responds (see Figure 7.1).

One characteristic of the paging system is that the person sending the message can never be certain that it has been received. There can also be a considerable gap between the time a message has been sent and when it is received by the addressee, and during peak load times this can amount to 10 minutes.

Table 7.1: Public operation of paging systems in Europe

Since 1974	European Paging Service (Eurosignal)
Since 1989	Cityruf
Since 1990	Euromessage
Since 1996	ERMES

Table 7.2: Call types of different paging systems

Call type	Eurosignal	Cityruf	ERMES
Tone only [No. of paging no.]	Up to 4	Up to 4	Up to 8
Numeric [No. of num. char.]	—	Up to 15	20–16 000
Alphanumeric [No. of char.]	—	Up to 80	400–9000
Transparent data transm. [max. length]	—	—	4000 bits

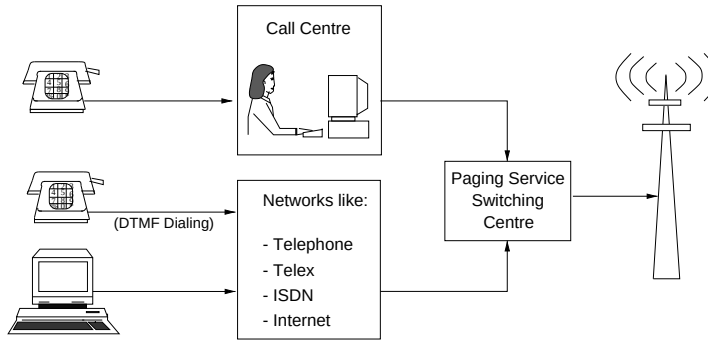


Figure 7.1: Principle of paging

Strengths and Weaknesses of Paging Systems

Strengths

- Inexpensive alerting and information service.
- Small receivers.
- The user of a paging system can be reached anywhere if the receiver is carried on his/her person.
- No additional antenna required (this applies only partially in the case of Eurosignal).
- Discrete communication of messages (messages are signalled through vibration, messages can be read on display).

Weaknesses

- The sender of a paging message receives no confirmation that the message has been received.
- Falsified paging messages and pages by malicious callers can cause the user of a pager to take inappropriate action.

Types of Calls

Individual calls The paging message is transmitted to the defined paging areas (one or more) or in the dialled paging area.

Collective calls A collective call number consists of n individual telephone numbers (receivers with different addresses). The receivers within a *paging area* are called in *consecutive* order.

Group calls Several receivers with the same address are called *simultaneously* in one or several paging areas.

Target calls By dialling additional numbers, the caller specifies the paging area in which the paging message should be transmitted.

It is anticipated that the number of users of pagers will increase in the future, despite the availability of services such as the *short-message service* (SMS) offered by GSM that are regarded as competition. A pager is an ideal extension to the mobile telephone because it can often be reached in places outside the radio coverage range of mobile telephone systems.

The miniaturization of terminals, including waterproof wristwatches that incorporate a radio pager receiver, offer the potential for a mass market for paging services that will extend beyond commercial applications to private and leisure use.

7.1 Paging Service "Cityruf"

Paging systems that operate in different frequency bands (150–170 MHz or 440–470 MHz) in accordance with POCSAG-Code (the Post Office Code Standardization Advisory Group pager system was developed as long ago as 1981) (*CCIR Radio-Paging Code No.1*) are now being used in many parts of Europe. Subscribers can be reached by practically all communications networks over the POCSAG radio calling service (see Table 7.3). A typical example of the POCSAG paging service is Cityruf, which was introduced in Germany in March 1989.

A radio paging system based on POCSAG (see Figure 7.1) consists of:

- A paging service switching centre that forms the link between the various communications services (telephone, data, etc.) and the paging network, in which incoming data is prepared for processing in the paging computer.
- Paging transmitters with transmitter power up to 100 W.
- A paging concentrator, which is used to switch on a paging transmitter at the paging service switching centre and is allocated a paging measurement receiver for the automatic propagation control of the modulation paths.
- A paging receiver.

As the name implies, the Cityruf network is not envisaged to be a wide-area service. It is a regional paging service for cities, and its coverage area is divided into internetworked paging areas. About 50 paging areas with a maximum diameter of 70 km have been implemented in the final configuration. However, subscribers to the Cityruf service are not only booked in locally

or regionally but in several paging areas and, in fact, throughout Germany. The transmitter systems of Cityruf are designed in such a way that they can guarantee reception within buildings without the need for additional antennas. The maximum number of subscribers that the system is capable of addressing is two million. Fifteen paging messages can be sent in one second.

Users of Cityruf have a choice between different calling classes based on different monthly charges:

Calling class 0 (tone only) Input is over the telephone. Tone-only devices issue up to four optically and acoustically different signals.

Calling class 1 (numeric) A maximum of 15 digits or special characters can be entered directly over the telephone using a supplementary device or with a dual-tone multiple-frequency (DTMF) signal generator, and then appear in the display of the terminal.

Calling class 2 (alphanumeric) Text messages up to 80 characters long can be entered, for example over the Internet or using an acoustic coupler device over the PSTN. Several messages can be transmitted in succession.

The following types of calls are available for the three calling classes:

Individual calls A radio call is broadcast in the registered paging areas or to the selected paging area.

Group calls A number of receivers are addressed over a group number at the same time.

Collective calls Up to 20 individual paging numbers are assembled in a list and then dialled up automatically in succession.

Target calls The receiver of a call is assigned a special number. To reach this person, a caller first dials the paging number and then the suffix number of the radio paging area in which the call is to be transmitted.

In addition to terminals and accessories for Cityruf, radio paging network operators offer the supplementary service Inforuf, which allows the user to receive information such as stock market updates, business news and weather reports. To access this information, users require a special Inforuf receiver that receives the Inforuf signals along with the Cityruf signals and is capable of storing 80 000 characters and reading them out on its 80-character display. The current information providers include Reuters, Telerate and pooled information services.

Cityruf is transmitted on the following frequencies:

- 465.970 MHz
- 466.075 MHz
- 466.230 MHz

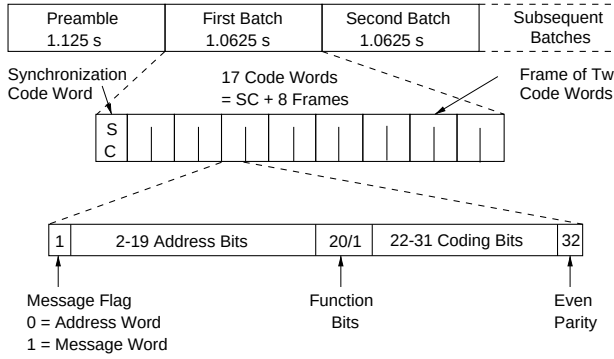


Figure 7.2: Message coding and block format with POCSAG

The transmission rate is 512 bit/s or 1200 bit/s. The digital signals are NRZ (*Non-Return to Zero*) coded and modulated through the use of differential frequency-shift keying (DFSK). The transmitters emit *bursts* of data blocks with code words (see Figure 7.2) [2].

Each burst begins with a 1.125 s long preamble, which is followed by a number of data blocks of 1.0625 s duration. The preamble enables synchronization to take place on the signal pulse on the receiver side, and is a prerequisite for error-free message coding. The data blocks consist of 17 code words, the first of which is used in the synchronization. The remaining 16 code words, each with 32 bits, contain an initial bit that indicates whether the code word is an address or a message, 20 address or message bits, 10 bits for error detection and correction, and a parity bit. The 16 code words form eight frames, each of which contains two code words. Each terminal is only addressed in a specific frame. The message for a receiver can be of any length, and is sent in the form of message words based on the address of the person receiving the message.

Paging areas can be divided into several radio coverage areas, with all transmitters within an area transmitting in a common frequency. The three frequencies are not used simultaneously in a transmitting zone, but are staggered and cyclical. Transmission never takes place simultaneously on the same frequency in the adjacent transmitting zones, so that a three-site cluster occurs. The time (time slot) during which transmission can take place on a frequency can be adapted to traffic volume. The advantages of this procedure are that adjacent zones can be decoupled through radio engineering and a receiver only needs to be operational when its frequency is being sent. If no messages are being sent on the frequency of the receiver or if it recognizes that a radio call is not directed at it, the receiver remains in battery-saving idle state.

Cityruf receivers are small in size, include storage options for characters which are received and use minimal power only. Table 7.3 lists the service codes required for network access.

Table 7.3: Cityruf network access codes

Access over		Code/paging number/paging area
PSTN/ISDN	Tone only	0164/XXXXXX/YY
	Numeric	0168/XXXXXX/YY
	Alphanumeric	01691/XXXXXX/YY
	Bureau service	016951
IDN	Telex network	1691/XXXXXX/YY
	Datex-L	1692/XXXXXX/YY
T-Online		*1691#/XXXXXX/YY

Development of Service

1989 Start of service to the public.

1990 Internetworking of Cityruf with the systems ALPHAPAGE in France, TELEDRIN in Italy, EUROPAGE in Great Britain.

1991 Introduction of the *Inforuf* service for closed user groups, introduction of alphanumeric messages with multifrequency dialling using the access code 0168, internetworking of Cityruf with radio paging system in Switzerland.

1992 Automatic dialling through telephone answering machines, alphanumeric access through access code 01691 at higher transmission rates (up to 24 kbit/s).

1994 Remote control and telemonitoring with Cityruf, input of numeric messages using voice.

7.2 Euromessage

In addition to Cityruf, which is a national service, there is also a European-wide radio paging service based on POCSAG called Euromessage (*European Messaging*). This is an extension of Cityruf that was set up in March 1990 to include an international service through the internetworking of the national radio paging services in Germany (Cityruf), France (Alphapage), Italy (Teledrin) and Great Britain (Europage).

Subscribers to this service wishing to be reached whilst abroad must notify the service operator with details about the period of time they will be away and in which paging area they can be located. All messages that arrive for a subscriber during this period are then rerouted to the appropriate paging area.

Euromessage is considered to be an intermediate solution until the availability of the Pan-European standardized radio paging service ERMES.

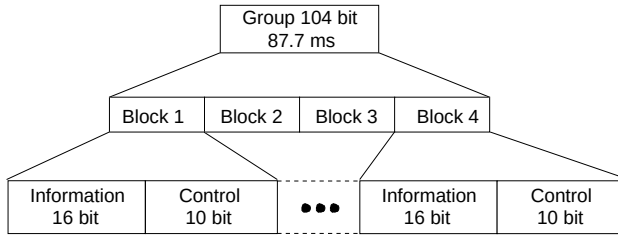


Figure 7.3: The RDS block format

7.3 RDS Paging System

RDS (*Radio Data System*), which was specified by the European Broadcast Union (EBU) and passed in 1984, is used for transmitting supplementary information over VHF radio broadcast transmitters such as:

- Transmitter recognition
- Alternative transmitter frequencies
- Paging
- Programme information
- Traffic information

The supplementary information consists of digital data, which is combined into groups of 104 bits each (see Figure 7.3). Each of these groups has a code to indicate which type of supplementary information it contains. Groups that contain tuning and switching information are transmitted more frequently than other groups.

The countries that use RDS for transmitting radio paging services are Sweden since 1978, France since 1987 and Ireland. Other countries, including Spain and Norway, are planning to introduce the service. Germany introduced RDS in 1988.

The RDS paging service offers the advantages that investment costs are low and subscribers can be reached anywhere because of countrywide VHF radio broadcast coverage with the existing transmitter network and shared use of VHF frequencies. Alphanumeric messages cannot be transmitted in the RDS system.

7.4 ERMES

ERMES (*European Radio Messaging System*) is a European-wide paging service that was developed as the result of a resolution adopted by the CEPT countries in 1986. The standardization work begun under the responsibility of CEPT was continued by ETSI in 1989 and completed in 1992.

By late 1990, 27 European PTTs and radio paging network operators had signed a memorandum of understanding (MoU) that guaranteed the implementation of the ERMES radio paging service. The reason why European countries are so keen on ERMES is that it offers significant advantages over existing paging services. Compared with POCSAG-based paging systems, ERMES has a higher channel capacity because of a high transmission rate; furthermore, because of the availability of transparent data transmission, it is possible to transmit any data with up to 64 kbit/s. Added to this is the possibility of international roaming and, in the case of several operators, national roaming.

ERMES represents the first standardized paging system in Europe that operates in the same frequency band throughout Europe and guarantees the accessibility of its subscribers throughout Europe.

Because of its 6250 bit/s bit rate, the capacity of an ERMES channel is four times higher than a 1200 bit/s POCSAG channel. Between 300 000 and 400 000 subscribers can be served per channel, which equates to a system capacity of around six million subscribers. ERMES pagers are being designed to require extremely little battery power and are smaller than equivalent POCSAG models.

7.4.1 The Services of the ERMES Paging System

ERMES offers its subscribers a number of basic services as well as supplementary services. The basic services that each operator is required to provide include [3]:

- Tone-only calling in which ERMES supports up to eight different tone signals per *Radio Identity Code* (`RIC`). This means that a tone receiver with a `RIC` can produce eight different alarm signals.
- Numeric radio paging in which the receiver has a display with at least 20 digits and also supports the tone-only function.
- Alphanumeric paging with a minimum of 400 characters and a receiver that is also suitable for tone-only calls and numeric pages.
- Transparent data transmission at 64 kbit/s, which can also be used for process control, telemetry and alarm activation.
- Roaming: the network recognizes the paging area in which a subscriber is located.

ERMES is planning the following supplementary services that can be offered by an operator as an option:

- Standard text allows an alphanumeric page to be sent through the input of DTMF codes over the telephone. Each code is linked to a standard text, such as “Meet in one hour”.

- Group call (see Section 7.1).
- Collective call (see Section 7.1).
- Call forwarding to another receiver.
- Storage of incoming messages, which the subscriber can then request to be sent at a later time.
- Numbering of messages and automatic retransmission of the number of the last message. The receiver will recognize from the sequence number whether certain messages have been received.
- Repetition of call when requested.
- Temporary call barring.
- Transmission of calls with different categories of priority.
- Target call in which the caller determines the paging area in which a call should be broadcast.
- Closed user group.
- Display of the category of urgency of a message and acceptance of a message based on its urgency.
- The person calling can indicate the time when a message should be sent.
- Encryption of messages.

7.4.2 ERMES Network Architecture

The system structure for ERMES is presented in Figure 7.4 [4]. The *Paging Network Controller* (PNC) processes the input received over the telephone, data network and from other networks, with consideration of the services agreed with a subscriber and stored in the network controller. With ERMES, messages for a tone or numeric pager can be conveyed over DTMF (*Dual-Tone Multiple Frequency*) signals. Paging text messages can be input over any conventional data network (ISDN, PSPDN, CSPDN, etc.) using standardized UPC protocols. Furthermore, the broadcast of a radio page can be initiated through arrangement with the operator. The network controller is therefore responsible for the following tasks:

- Provide a user with access to ERMES over the fixed network.
- Control and administer the database containing subscriber data.
- Provide the possibility of roaming through its link to other ERMES networks over an interface in accordance with CCITT Rec. X.200.

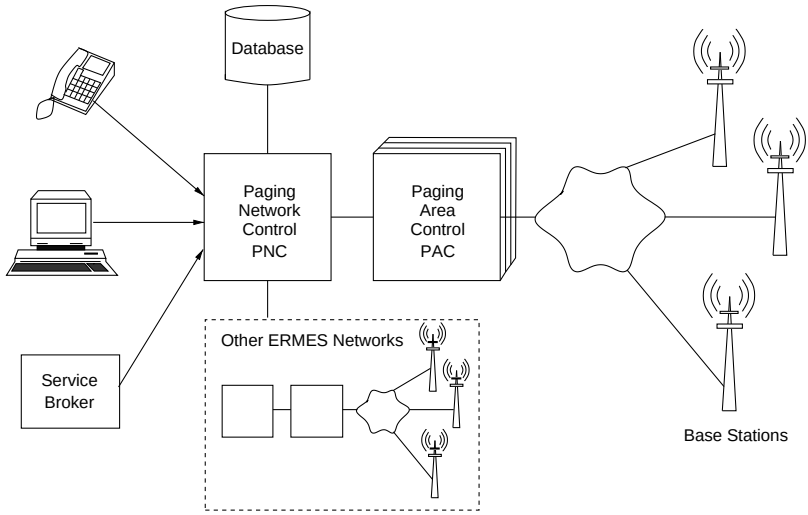


Figure 7.4: ERMES system structure

- Control radio transmission in the service areas.

Up to 64 *Paging Area Controllers* (PAC) can be connected to a network controller (see Figure 7.4). An area controller organizes the broadcast of a call and, in accordance with the agreement with the subscriber, activates one or more base stations in order to ensure that coverage is provided to the paging area concerned. Depending on the priority of incoming pages, the PAC forwards them to the base stations. The base stations (BS) transmit with up to 100 W and can reach receivers within a radius of up to 15 km.

7.4.3 Technical Parameters of the ERMES Paging System

Sixteen channels with a bandwidth of 25 kHz in the 169.4125–169.8125 MHz frequency band have been allocated to ERMES in Europe. With the use of the 4-PAM/FM modulation procedure the data transmission rate is 6.25 kbit/s or 3.125 kBaud/s.

An abbreviated cyclical code (30,18) derived from BCH (31,21) is used as an error-correction code. This code is supported by interleaving with a depth of nine.

Network operators and users have been assigned identities to ensure that messages in the ERMES system are routed correctly [1]. An *operator identity* `OPID` is 13 bits long (see Figure 7.5).

The zone and country codes are derived from CCITT Rec. E.212. The 3-bit long operator code supports eight different operators per country. Should further operator codes be necessary, additional country codes will be granted to differentiate between them.

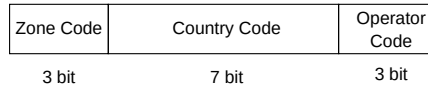


Figure 7.5: The structure of an `OPID`

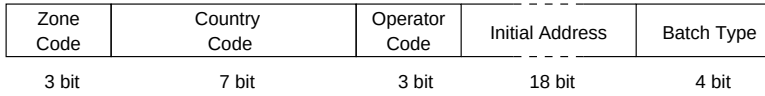


Figure 7.6: The structure of an `RIC`

The structure of the user identity (*Radio Identity Code*, `RIC`) consists of 35 bits and is shown in Figure 7.6. The `RIC` contains the 13-bit long `OPID` and a further 22 bits for the initial address and the batch type, which together form a so-called local address. A receiver is able to have several user identities stored.

Depending on its description, an ERMES network can be set up as a:

- *Time Division Network* (TDN): Transmission in neighbouring or overlapping paging areas is at different time intervals.
- *Frequency Division Network* (FDN): Transmission in neighbouring or overlapping paging areas is on different frequencies.

A combination of the two types of operation is also possible. The channel and block structures of the transmission protocol are presented in Figure 7.7.

The periodic sequence forms the primary structure of 60-minute duration. It consists of 60 *cycles*, each of which lasts one minute and is used in the coordination between different networks. Because the receivers are designed to save on battery power, they can only listen in on one or a few cycles. Each cycle contains five subsequences, each 12 s in duration. In TDN the transmission for a paging area depends on the traffic volume and takes place in at least one of these subsequences. Each subsequence is divided into 16 *batches*, with each of the first 15 batches containing 154 words (30 bit/word) and the 16th having a length of 190 words. Through the last 4 bits in the `RIC`, the receiver is allocated one of the 16 possible types of batches and can be addressed in the batch type specified for it. A batch is divided into four parts:

Synchronization Partition The preamble word (PW) and the synchronization word (SW) enable the receiver to carry out the symbol or code word synchronization for the batch.

System Information Partion This part consists of:

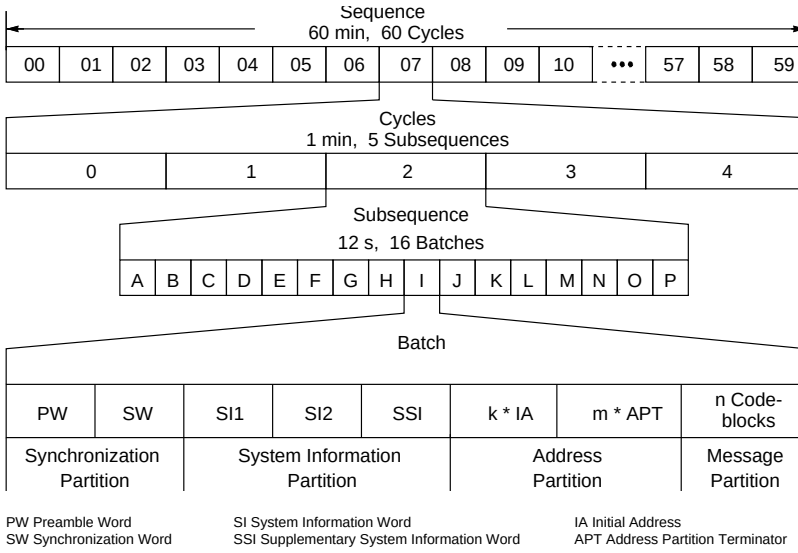


Figure 7.7: Message coding and block format in ERMES

- Two system information words (SI1, SI2), of which SI1 notifies the receiver of the operator identity (ID) and the paging area (PA) number. SI2 gives the position of the batch in the sequence and the channel of the transmitter.
- An additional system information word that contains either the area code, the time and the date or only the date.

Address Partition This partition contains a number of initial addresses (IA) and at least one *Address Partition Terminator* (APT). The initial addresses are transmitted in descending order. If the initial address received by a receiver is smaller than its own then it cannot receive any further radio pages in this batch.

Message Partition This part contains the local address of a receiver, the message number, the type of message and the message itself.

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8
Cordless Telephone Systems

Cordless telephone systems are mobile radio systems that, although their transmission range is not suitable for wide-area coverage, are attractive to many users because of the services offered and their low cost.

Cordless telephony (CT) can be used within the area of a subscriber line, and provides a radio link within a radius of approximately 50 m inside a building or 300 m outdoors between the fixed terminal and a mobile part. This requires the fixed terminal, as the base station, and the mobile part each to be equipped with transmitting and receiving capabilities.

Along with their suitability for home use (a base station, a mobile part), CT systems lend themselves to other uses:

- Microcellular cordless private branch exchanges, e.g., for use in offices and industrial plants.
- Public cellular systems with local coverage and as regional or nationwide networks for public cordless coin-operated telephones from which calls can be made from appropriate portable terminals without use of previously conventional telephone cells (Telepoint service).
- Wireless access technology for stationary users of telecommunications networks.

The first cordless telephones in Europe came from the USA and the Far East. This CT0-designated equipment, which was not authorized in the European countries, used analogue transmission technology at 1.6 or 4.7 MHz with eight channels (each with 25 kHz), and because of the lack of security provisions was particularly susceptible to eavesdropping (see Table 8.1). Furthermore, the equipment could be used (fraudulantly) as a telephone handset to access other base stations and their lines.

The disadvantages of CT0 equipment led to the development of the CT1 standard for cordless telephony in 1983 under the aegis of CEPT. Two bands with a 2 MHz bandwidth and 45 MHz duplex spacing at 900 MHz were stipulated as the frequency bands for these systems, which incorporated analogue transmission technology (see Table 8.1). The frequencies from 914 to 916 MHz were designated for the link between the mobile and the fixed parts, and the frequencies from 959 to 961 MHz for the transmission from the fixed part to the cordless telephone. In accordance with the CT1 standard, FDMA technology is used to access the total 40 channels available, each with a 25 kHz

Table 8.1: The main parameters of cordless analogue telephone systems

System	CT0	CT1	CT1+
Signal transmission	Analogue	Analogue	Analogue
Frequency band [MHz]	1.6/4.7	914–915 959–961	885–887 930–932
No. of channels	8.0	40	80
Bandwidth [MHz]	0.4	2	4
Channel spacing [kHz]		25	12.5
Access procedure	FDMA	FDMA	FDMA
Duplex procedure	FDD	FDD	FDD
Channel allocation	Fixed	Dynamic	Dynamic
Cellular networks	No	Limited	Limited
Max. transmitter power [mW]		10	10
Range [m]	< 1000	< 300	< 300
Handover	No	No	No
Capacity [Erl./km^2]	1.0	200	200

bandwidth. These channels are allocated dynamically; there is a fixed allocation of frequencies to individual devices or local areas.

It soon emerged that the number of channels available with the CT1 standard was inadequate for metropolitan areas, and this led to the decision by different countries, including Germany, to introduce a system with 80 channels in accordance with the CT1+ standard in 1989. With CT1+ the mobile part transmits to the fixed part in the 885–887 MHz frequency range, whereas the frequencies between 930 and 932 MHz are used for the link between the fixed part and the mobile device (see Table 8.1). An organizational channel is being planned for CT1+.

Unauthorized access to the base station has been almost completely eliminated with the CT1 and CT1+ standards because of a number of codes that provide a clear identification between the mobile and fixed parts; however, these systems are not fully protected from eavesdropping.

8.1 CT2/CAI and Telepoint

In contrast to other European countries, in the mid-1980s Great Britain started using a version of cordless telephony incorporating the T-standard from the USA. Because these systems did not provide adequate capacity (eight channels) and there was concern about a foreseeable frequency collision with GSM, Great Britain was not keen to continue with the CT1-standard, and a digital standard for cordless telephony was developed at the initiative of the network operator *British Telecom*.

With the aim of making cordless telephony attractive to a broad range of customers, it was planned that CT2 terminals should also be used for Tele-

point applications. The Telepoint concept allows users with the appropriate equipment to set up a connection to the public telephone network over a Telepoint base station within a radius of up to 300 m of a public and highly frequented area (pedestrian area, railway station, airport, shopping centre, etc.), but with no facilities for receiving calls [1].

The CT2 standard is frequently supplemented with the abbreviation CAI (*Common Air Interface*), which denotes a radio interface between the fixed part and the mobile part developed by the Department of Trade and Industry (DTI) in Great Britain with the participation of British industry. The interface was required to enable existing non-compatible terminal equipment to access the Telepoint service.

The CT2/CAI standard was designed with enough flexibility so that the same handset can be used to make calls at an office, at home and to the public network. CAI basically allows incoming as well as outgoing calls. The restriction that exists with Telepoint, i.e., that only outgoing calls are possible, is only due to the terms of the licence and not to any technological limitations.

CT2/CAI has now been adopted as an ETSI Interim-Standard.

8.2 Technical Parameters of CT2/CAI

Technically, CT2 is based on the CT1 standard; however, digital transmission has been introduced (see Table 8.2). The system has more than 40 frequency channels, each with a 100 kHz bandwidth, in the frequency range between 864

Table 8.2: The main parameters of cordless digital telephone systems

System	CT2/CAI	DECT
Signal transmission	Digital	Digital
Frequency band [MHz]	864–868	1880–1900
No. of channels	40	120
Bandwidth [MHz]	4	20
Channel separation [kHz]	100	1.728
Access procedure	FDMA	FDMA/TDMA
Duplex procedure	TDD	TDD
Channel allocation	Dynamic	Dynamic
Voice channels/carriers	1	12
Coding	ADPCM 32 kbit/s	ADPCM 32 kbit/s
Modulation data rate	72 kbit/s	1.152 Mbit/s
Modulation	Two-level GFSK	GFSK
Cellular networks	Limited	Yes
Max. transmitter power [mW]	10	250
Range	< 300	< 300
Handover	No	Yes
Capacity [Erl./km^2]	250	10 000

and 868 MHz, which, unlike CT1, does not collide with other standardized mobile radio services in Europe.

Two-level GFSK (*Gaussian Frequency Shift Keying*) is used for modulation. The nominal bit rate per channel is 72 kbit/s, which means that data transmission is also possible, albeit at a relatively low net rate.

CT2/CAI is the first ever mobile radio system in which the base station as well as the mobile terminal transmit on the same radio channel. Yet the two transmission directions are not separated by different frequencies; instead the pingpong technique is employed, in which the transmission direction is changed every millisecond on the same frequency. With this procedure, called *Time-Division Duplex* (TDD), the expensive filters required with *Frequency-Division Duplexing* (FDD) for switching transmission direction are replaced by simple switches.

The functions of the protocols of layers 1 to 3 specified for the air interface in the CAI standard correspond to those of the ISO/OSI reference model:

Layer 1 The specifications apply to the physical transmission system, including modulation procedure, frame structure, synchronization, time behaviour and bit rate, along with channel selection and link control.

Layer 2 The tasks of this layer include error detection, error correction, message acknowledgement, connection control and identification.

Layer 3 The functions and message elements of this layer are used in the signalling for connection control, similarly to the ISDN-D-channel protocol. It is responsible for recognizing the type of messages, for setting up and terminating calls and for maintaining connections.

Three logical subchannels are defined by the standard:

- The B-channel, in which speech and, with the use of a modem, data can be transmitted.
- The D-channel, for signalling.
- The SYN-channel, which transmits information on bit and burst synchronization.

Depending on the application, these subchannels are allocated different channel capacities in four different socalled multiplex frames (see Figure 8.1) [2].

Multiplex 1.2 This defines a 66-bit long burst of duration 1 ms that contains 64 B-channel bits and 2 D-channel bits. Since this frame is used with an existing connection, there is no need for an SYN channel. If the synchronism is lost when this operating mode is used, the connection must be reinitialized. Because this burst is sent every 2 ms, the data rate in the B-channel is 32 kbit/s and that in the D-channel 1 kbit/s.

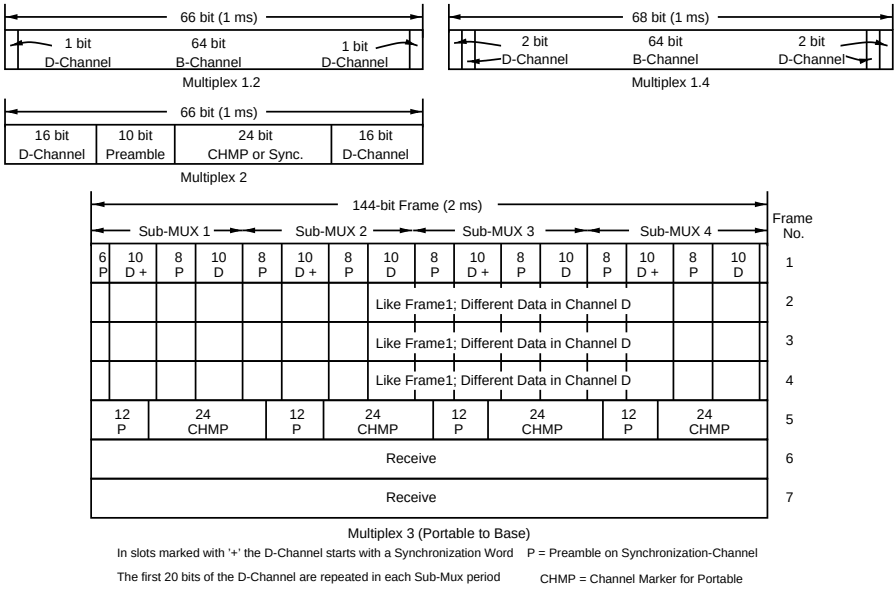


Figure 8.1: The structures of the multiplex frames with CAI

Multiplex 1.4 This is set up similarly to the Multiplex 1.2 burst, although a total of 68 bits are transmitted in 1 ms, of which four are D-channel bits. Therefore the data rate on the D-channel is 2 kbit/s. This multiplex frame is used with an existing connection if the base station and the mobile device have indicated during call setup that they are able to support the 68-bit long burst.

Multiplex 2 This is a 66-bit long burst consisting of 32 D-channel bits and 34 SYN-channel bits, and is used during normal call setup and to restore an interrupted connection. A total of 16 kbit/s are available to the D-channel and 17 kbit/s for the SYN-channel. The SYN-field consists of a 10-bit long preamble, followed by one of three different 24-bit long synchronization patterns—a so-called *channel marker for portable*.

Multiplex 3 This defines a 10 ms long burst that is used to set up and restore a connection that has been initiated by a handset. Here 10 ms are divided into five 2 ms long frames, each of which is subdivided into four identical *subframes*. The first four frames contain 20 D-channel bits and 16 preamble bits in each subframe, whereas the fifth frame only contains SYN-channel bits.

Since the B-channel is used for transmitting speech, the data rate is 32 kbit/s. Speech is coded through *Adaptive Differential Pulse Code Modulation* (ADPCM).

The transmitter power of cordless CT2 equipment is a maximum of 10 mW, which provides a range outdoors of up to 200 m between the mobile part and the base station. Since distances are rarely this great in practice, there are plans in the CAI standard to reduce the transmitter power of the mobile parts dynamically in order to reduce interference. This is carried out by the base station, which measures the received field strength and, if necessary, signals the handset to reduce its transmitter power.

The CAI standard contains a number of security measures. It supports the exchange of identifiers between the fixed and mobile parts, with the cordless CT2 terminal of a subscriber not being available for use until a *personal identification number* (PIN) has been entered. Authentication between the terminal and the fixed part is supported through the appropriate allocation of serial numbers. Because of the use of digital transmission technology, voice encryption is easy but efficient, thereby making CT2/CAI systems safe from eavesdropping. The CAI specifications allow a subscriber to have charges transferred to other telephone numbers (e.g., of his company) and billed there. Compared with CT1+ with the same bandwidth, these systems have a higher capacity of around 250 Erl./km² because of digital transmission.

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9
DECT*

Besides cellular mobile radio networks that are primarily envisaged for use outdoors, systems that have been specifically designed for use in buildings are also important. In recent years cordless telephones with a range of a few hundred metres have become increasingly popular in private households. There is (along with CT2/CAI) a digital alternative to these analogue devices that offers better voice quality and a greater security against eavesdropping, as well as other advantages: the DECT system.

The abbreviation DECT originally stood for *Digital European Cordless Telecommunications*, but to underline its claim of being a worldwide standard for cordless telephony DECT today stands for *Digital Enhanced Cordless Telecommunications*. This standard was specified by the *European Telecommunications Standards Institute* (ETSI) in 1992. A DECT network is a microcellular digital mobile radio network for high user density and primarily for use inside buildings. However, an outdoor application is also possible.

DECT systems allow complete cordless private branch exchanges to be set up in office buildings. Calls can be made over the normal subscriber line as well as between mobile stations through the DECT base station subsystem. When a staff member leaves his office, he usually can no longer be reached for incoming calls over a wireline telephone connection, although he may only be located in another part of the building. But if he uses a DECT terminal, he can continue to be reached under his normal telephone number wherever he may be anywhere on the premises. To enable users to continue to receive calls after they have left the DECT coverage area, ETSI specified an interface between GSM and DECT (see Section 9.16).

The DECT standard permits the transmission of voice and data signals. Consequently cordless data networks can also be set up on a DECT basis. The use of ISDN services (*Integrated Services Digital Network*) is also possible. Users are able to move freely within different cells without risking an interruption to their calls. The handover process switches calls from one radio cell to the next without interrupting the call.

In outdoor areas the maximum distance between base and mobile station is approximately 300 m; in buildings, depending on the location, it is up to 50 m. Larger distances to the base station can be bridged through the installation of appropriate base stations using the relay concept (see also Section 9.12).

*With the collaboration of Christian Plenge and Markus Scheibenbogen

The first time DECT systems were presented to the public was at the CeBIT exhibition in Hanover in March 1993.

Since then, the costs for a DECT mobile have been decreased to be comparable to those of analogue cordless telephones.

9.1 Possible Applications of DECT Systems

The size of a DECT system determines how it is installed. The labels *large* and *small* are relative and relate to the number of mobile stations to be operated within the DECT coverage area. A DECT system is capable of automatically localizing up to 1000 subscribers in one *location area* (LA). With a larger number of users additional location areas are required that fall under the administration of the DECT system (see Sections 9.1.2 and 9.16.1.3).

9.1.1 DECT Fixed Networks

DECT fixed networks are normally designed as dedicated private branch exchanges.

If the DECT systems are installed by a network operator, each customer-specific DECT system can be allocated its own location area. All DECT location areas are then interconnected over a backbone ring and administered centrally by a DECT system, as illustrated in Figure 9.8. Each customer has his own *DECT fixed station* (DFS) with his own identification of the location area.

Despite the efficient location administration of DECT systems, it was planned that each customer would be allocated his own location area to ensure that the channel capacity of other customers would not be affected by the calling activity of his own mobile stations.

Private home base stations These offer a possibility of using the DECT system in small private households (see Figure 9.1). The home base station supplies the entire area of a house and can support one or several mobile stations that are supplied by the base station. The home station consists of a *DECT fixed system* (DFS), which controls the system, a simple *database* (DB) for user administration and a *fixed part* (FP), which provides the radio supply to the mobile stations. An *interworking unit* (IWU) is provided for connection to an external network [20].

Wireless private branch exchanges In terms of installation, central systems with a DFS to which several FPs are connected are suitable for large private households or small companies (see Figure 9.2). Fixed terminals can also be connected. The system is administered by the *DECT fixed system*. The correct location area of the mobile stations is stored in a database (DB), and an IWU enables a connection to the external networks.

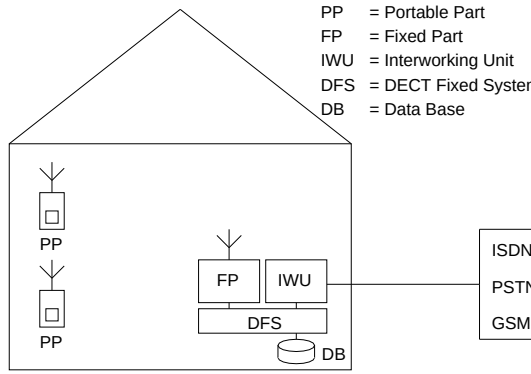


Figure 9.1: Private home base station

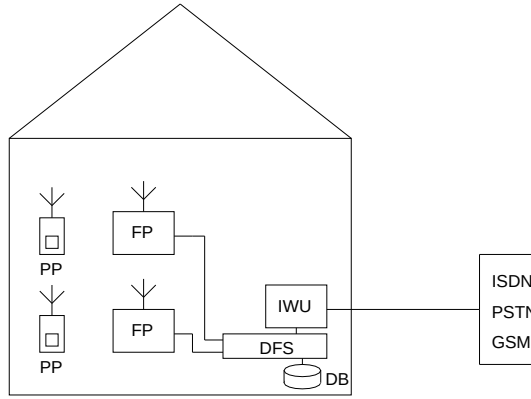


Figure 9.2: Wireless private branch exchanges

Public Telepoint systems These systems provide DECT mobile stations access to the public telephone network over "other" FPs at public sites (see Figure 9.3). Possible application areas are public facilities with high user density such as airports, train stations and city centres. These private and public DECT systems consist of a number of FPs that are administered by a *DECT fixed system* (DFS). Access to the public telephone network can be provided by a network interface (With DECT, PPs in principle can also be reached in the area of a Telepoint. The restriction to outgoing calls is dictated by the respective licensing agreement).

Wireless local rings This is a ring to which a *terminal adaptor* (TA) and all the terminals are linked (see Figure 9.4). The terminal adaptor provides a connection over the radio interface to one of the public FPs in the area. The terminals (e.g., telephone, facsimile) are linked to the terminal adaptor over conventional lines. If necessary, the terminal adaptor

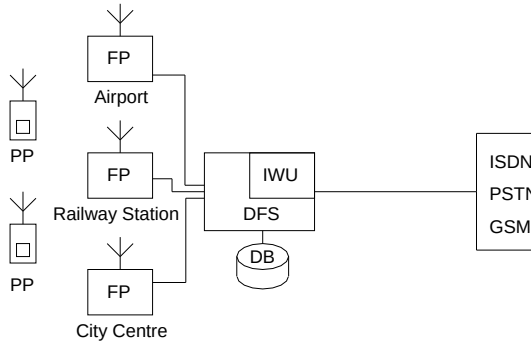


Figure 9.3: Public Telepoint system

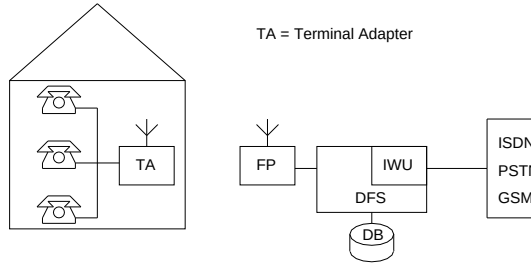


Figure 9.4: Wireless local rings

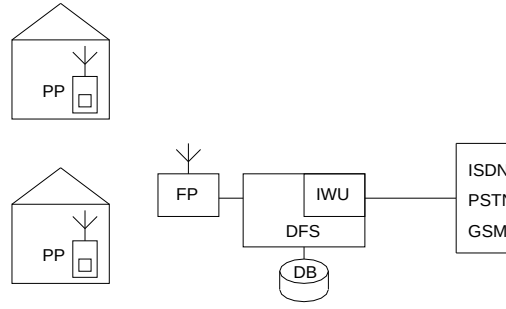


Figure 9.5: Neighbourhood Telepoint

establishes a radio connection to the next FP. The FP is administered in the same way as with the public Telepoint service.

Neighbourhood Telepoint This is a combination of the public Telepoint service and a private home base station (see Figure 9.5). Private households do not have their own FP; instead one FP supplies a number of different private households on the basis of the Telepoint principle. The FPs

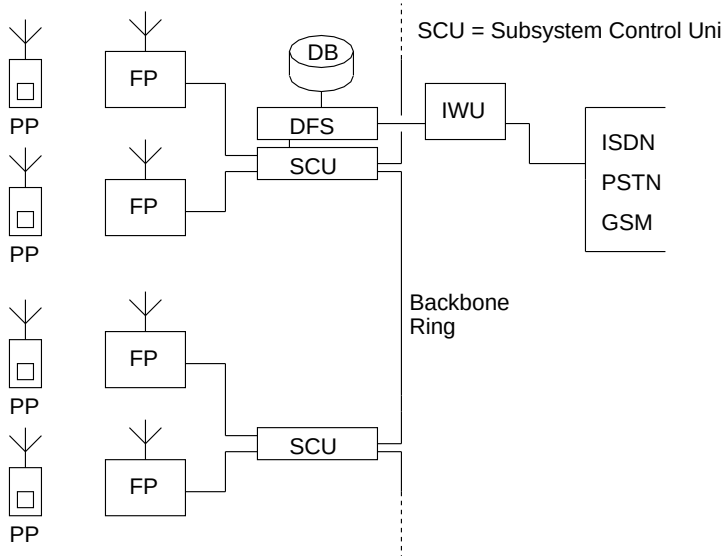


Figure 9.6: Private branch exchanges with ring and central DFS

are installed so that several households will always be supplied by one FP. A DECT fixed system connected over a network interface with an interworking unit (IWU) to the public telephone network manages the FPs.

Because of the way they are designed in terms of size and subscriber capacity, the following systems are also suitable for use by companies with scattered operating locations. One or more nodes can be provided for DECT at each company location. Each implementation of the system must be flexible and tailored to the needs of the customer in each individual case.

Private branch exchanges with ring and central DFS These private branch exchanges consist of a backbone ring with a number of switching nodes (see Figure 9.6). Each switching node consists of a *Subsystem Control Unit* (SCU) and several *base stations*, i.e., FPs. One of the nodes contains the DFS which is responsible for all the nodes connected to the backbone ring and controls them from a central function. The DFS also contains the *home database* (HDB) and the *interworking unit* (IWU), which constitutes the external connection. All signalling data transmitted in this system must be transmitted over several nodes (SCUs) as well as the backbone ring. Transmission channels of the network that are no longer available for transporting user data are used for this purpose. A disadvantage is that the entire network suffers if the central system breaks down. Some relief is provided by the decentral systems, which ensure that there is a reduction in signalling traffic and that only

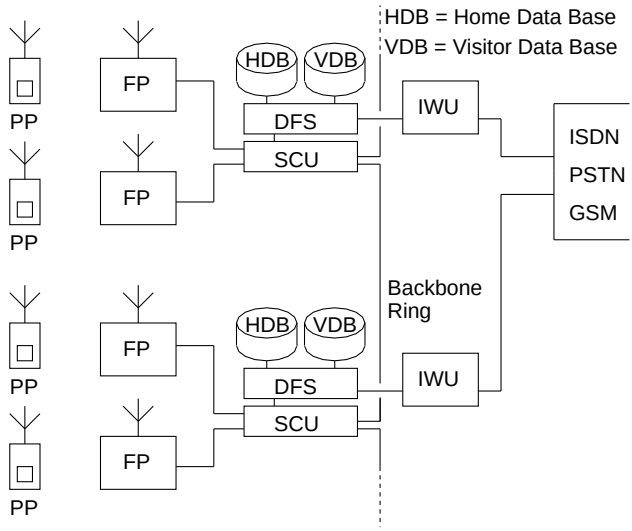


Figure 9.7: Private branch exchanges with ring and distributed DFS

the respective mobile stations are affected when the subsystem breaks down [20].

Private branch exchanges with ring and decentral DFS These are set up similarly to the systems with a central DFS (see Figure 9.7) and consist of a backbone ring that interconnects a number of nodes. A control unit is located at each of these nodes and several fixed stations are connected to a node.

This system operates as a decentral structure because each control unit is controlled by its own DECT fixed system. Furthermore, each DECT fixed system has an interworking unit (IWU) that connects it to the external networks. The data is also stored at decentral locations. Subscriber administration is carried out in a *home database* (HDB), from which important data for operations can be retrieved over the network.

For the purposes of reducing the signalling required when a mobile subscriber is roaming the network, certain information about each subscriber who moves out of the coverage area of his home database is written to the *visitor database* (VDB) of the new coverage area. The network provides access to the data in the visitor database, and the backbone ring is unloaded of responsibility for the signalling. If a DFS suffers a breakdown, only the subscribers moving in the respective coverage area will be affected [20].

Private branch exchanges with direct connection of FP These branch exchanges have the same decentral structure as the decentral private

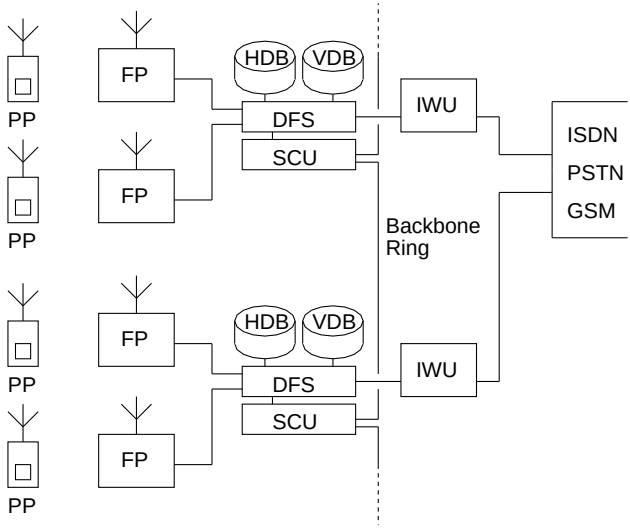


Figure 9.8: Private branch exchanges with direct connection of FP

branch exchanges (see Figure 9.8). The difference between them is that the fixed stations of these private branch exchanges are directly connected to the respective DFS, which has the advantage that there is a reduction in the signalling traffic within a node and consequently a higher capacity for user data [20].

9.1.2 Data Storage

Databases can be set up in different ways. Three different types are presented below:

SS 7-MAP The SS 7-MAP (*Common Channel Signalling System Number 7, Mobile Application Part*) is a method used by public cellular networks such as GSM900 and GSM1800. User data is stored in two registers: a *home location register* (HLR) and a *visitor location register* (VLR). Incoming calls are routed over the HLR to the VLR and from there to the mobile terminal. Outgoing calls only use the VLR. Access of the HLR is not required for these calls.

X.500 Another method is the storage of data based on the standardized register service ITU-T X.500. The database is distributed physically over different locations but logically centralized. The data therefore gives the impression of being stored in a *Directory Information Base* (DIB). All data is organized as objects in a hierarchical *Directory Information Tree* (DIT), in which each branch can be filed in a physically separate *Directory System Agent*. A *Directory User Agent* accesses the individual

objects. The search for the data takes place in a chain of interlinked databases. A critical factor of this method is the time involved in accessing the data, but this is less of a problem for private branch exchanges with a limit on subscriber numbers.

Telecommunication Management Network (TMN) A further method for data storage is TMN based on ITU-T M.30. A management system is controlled by a "manager" that realizes the management system using objects containing important data. These objects are stored in a hierarchical MIB database (*management information base*). This MIB can be distributed physically over different locations. The management system is able to reproduce the HLR and VLR databases and store all location area data in one object.

In comparison with public networks, only a limited number of users are managed in private networks such as DECT branch exchanges. The data for these users can usually be administered in one register. Moreover, it is easier to change objects in the frequently changing organizational structure of office environments than in conventional databases. This makes the HLR/VLR concept less suitable for private branch exchanges than the other alternatives. A comparison between X.500 and TMN shows that X.500 is more flexible because it is not required to copy the HLR/VLR principle.

9.2 The DECT Reference System

The DECT Reference System describes the defined logical and physical components of the DECT system, the interfaces between the different units and the connection points to other networks. The global logical structuring of a local DECT network is explained below. This is followed by examples of different physical implementations.

9.2.1 Logical Grouping of DECT systems

The logical groups of the DECT network are organized according to their functionalities with the intervening interfaces D1, D2, D3 and D4, which, however, do not describe how they are physically implemented (see Figure 9.9).

9.2.1.1 Global Network

The *global network* supports the national telecommunication services. It carries out address conversion, routing and relaying between the individual connected socalled local networks. The global network is usually a national (sometimes an international) network. Examples of local networks include:

- *Public Switched Telephone Network* (PSTN)

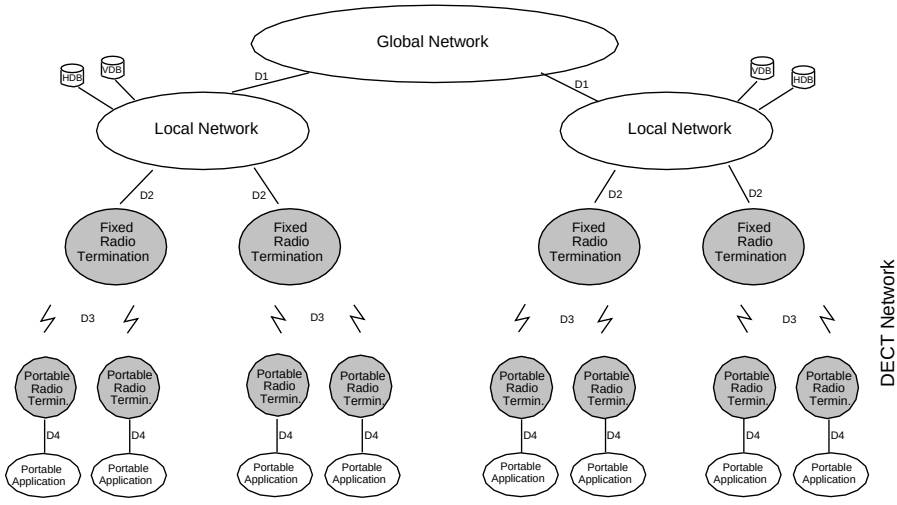


Figure 9.9: DECT Reference System: Logical grouping

- *Integrated Services Digital Network* (ISDN)
- *Packet Switched Public Data Network* (PSPDN)
- *Public Land Mobile Network* (PLMN)

9.2.1.2 Local Networks

Each local network provides a local telecommunications service. Depending on the actual installation, this can vary from a simple multiplexer to a highly-developed complex network. If the subordinate *DECT fixed radio termination* (FT) does not have a switching function then the local network must take over this function. Yet it should be noted that there can be a difference between logical definition and physical implementation, e.g., it is possible for several networks with their functions to be combined into one unit.

A local network converts the global identification numbers (e.g., ISDN numbers) to the DECT-specific `IPUI` (*International Portable User Identity*) and `TPUI` (*Temporary Portable User Identity*). The following networks are often found under the local network:

- analogue or digital *Private Automatic Branch Exchange* (PABX)
- *Integrated Services Private Branch Exchange* (ISPBX)
- IEEE 802 LAN: local area network based on IEEE 802

All the typical network functions must be embedded outside the DECT system and occur either in a local or in a global network. Similarly to GSM,

the HDB and VDB are required for controlling inter-DECT mobility, i.e., allowing users to use their mobile stations to move within different independent DECT areas (see Section 3.2.1.3). Incoming calls are automatically routed to the subsystem in which the user is currently located. When a user moves from one network to another, a new entry for the current VDB is made in the HDB.

9.2.1.3 DECT Network

The DECT network consists of base and mobile stations, and connects users to the local fixed network. No application processes are defined for the system, and it only functions as a multiplexing facility. A DECT system always has only one network address per user, i.e., mobile station, and (from a logical standpoint) consists of one or more *fixed radio terminations* (FT) and many *portable radio terminations* (PT) that have been allocated to it.

Fixed Radio Termination The FT is a logical grouping of all the functions and procedures on the fixed network side of the DECT air interface. It is responsible for:

- layer-3 protocol processes in the C-(*control*) layer (except for mobility)
- layer-2 protocol processes in the U-(*user*) layer
- layer-2 switching (routing and relaying) in the respective DECT network

Except for handover and multicell management, the FT contains no switching functions. Although it can manage a large number of call entities, it cannot establish a direct connection between two users. This can only be carried out outside the logically delineated area of the FT in the local network.

Portable Radio Termination and Portable Application These two parts constitute the logical groups on the mobile side of the DECT network. Whereas the *portable radio termination* with all its protocol elements for OSI layers 1, 2 and 3 is defined in the standard, it is up to the manufacturer of the equipment to define the acceptable application. Therefore it is not standardized.

9.2.2 Physical Grouping of DECT Systems

Whereas the logical structure of the DECT network is clearly defined, the physical grouping can assume different forms. It is adapted to each customer's requirements, and therefore can be conceptualized as a single base station with up to 12 simultaneously communicating mobile stations if it is equipped with one transceiver or as an independent switching centre for office buildings. The logical interfaces D1, . . . ,D4 are thereby partially integrated into a common physical unit and consequently can no longer be clearly apportioned (see Figure 9.10).

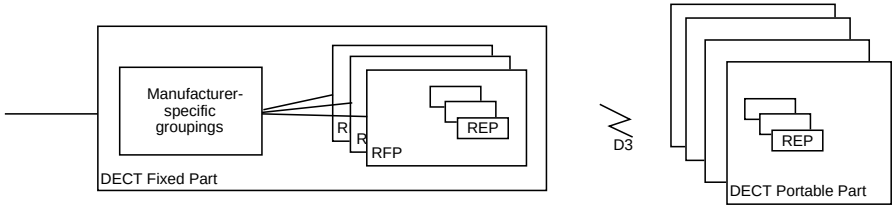


Figure 9.10: DECT Reference System: physical grouping

9.2.2.1 DECT Base Station (Fixed Part)

Physically a DECT system can be split into two parts: a DECT *fixed part* (FP) on the fixed side and a DECT *portable part* (PP) on the mobile side. The fixed part on the wireline side can contain one or more logical groups of the *fixed radio termination* with common control.

An FP can be divided into two physical subgroups:

- *Radio Fixed Part* (RFP): responsible for only one cell in the network
- *Radio End Point* (REP): corresponds to a transceiver unit in the RFP

9.2.2.2 DECT Mobile Device (Portable Part)

The two logical groups *portable radio termination* and *portable application* are physically combined into one *portable part* (PP), typically a handset. A PP normally only has one *radio endpoint*.

9.2.3 DECT Authentication Module (DAM)

A mobile station can be used by different users. Each user must identify himself before being granted access to the DECT network. This function is handled by a *DECT Authentication Module* (DAM), which contains information for the identification (*International Portable User Identity*, `IPUI`) and authentication (*Authentication Key*, `K`) of the user and can be inserted into the PP. The DAM contains all the required encryption procedures.

9.2.4 Specific DECT Configurations

The DECT system description [5] lists several typical DECT configurations. Different physical implementations are required, depending on the higher-ranking network:

• PSTN • ISDN • X.25 • IEEE 802 LAN • GSM

Examples of possible installations are given below. Several suppliers are already marketing private household systems at reasonable prices. Complex

private office installations are currently conquering the market. In addition, DECT systems have been implemented and tested (since 1997) as *Radio Local Loop* systems.

9.2.4.1 PSTN Reference Configuration

Domestic Telephone The private domestic telephone constitutes the simplest DECT configuration. Here the network is connected to a PSTN over a *subscriber interface*, just like a plain old telephone (POT) (see Figure 9.11). The functional characteristics resemble those of the earlier CT 1 and CT 2 generations of cordless domestic telephones (see Chapter 8). There is no provision for a local network.

PBX Similarly to the domestic telephone, in a simple implementation of a DECT-PBX, individual FPs are connected to the switching unit of the PSTN. Technically it would then be very complicated to allow a changeover from one FP to another FP during a call (handover).

In Figure 9.12 the *fixed part* contains several *radio fixed parts*, each of which always serves one cell, thereby giving the system its cellular character. A mobile station establishes a connection to the strongest RFP. If the user is roaming in the area of a neighbouring cell, the FP is then in a position of executing an internal layer-2 handover. The protocols for cell control are controlled by the common control (CC) function, which physically can be integrated into the FP or into the PBX.

Radio Local Loop A DECT system can also be incorporated as a local access network into the PSTN (see Figure 9.13). In this case the radio connection is transparent to the user. The user's wireline telephone is linked to a *Cordless Terminal Adapter* (CTA), which carries out the radio transmission to the RFP. In the radio in the local loop (RLL) area, suppliers of fixed networks are currently trying to avoid the high cost of local network cabling by using DECT-RLL systems for the so-called *last mile*. A comparable configuration using ISDN as the local fixed network and DECT as the RLL system is shown in Figure 9.14.

9.2.4.2 GSM Reference Configuration

Aside from the X.25 reference configuration, which is not covered here, the connection between the two mobile systems GSM and DECT should be mentioned. This development offers users the possibility of coupling a locally based DECT system with the national GSM mobile radio system (see Figure 9.15). From the standpoint of GSM, the portable application, PT, FT and possibly also a local network form a mobile user unit. The D1 reference point is the R reference point in the GSM standard (see Figure 3.3). Detailed coverage of the integration of DECT and GSM systems is provided in Section 9.16.

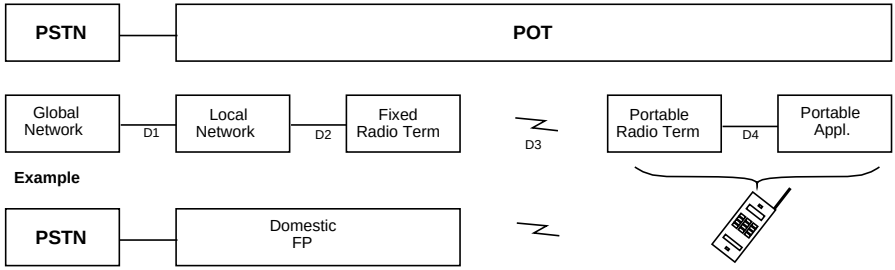


Figure 9.11: Domestic telephone configuration

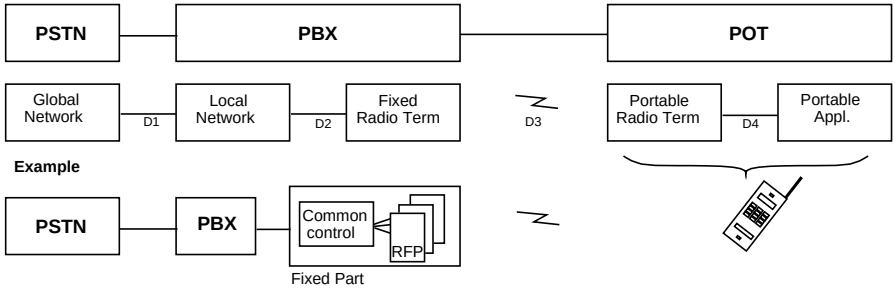


Figure 9.12: DECT-PBX configuration

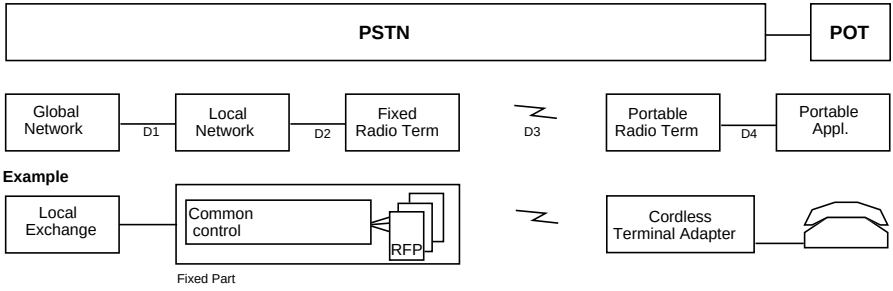


Figure 9.13: Radio local loop configuration for PSTN

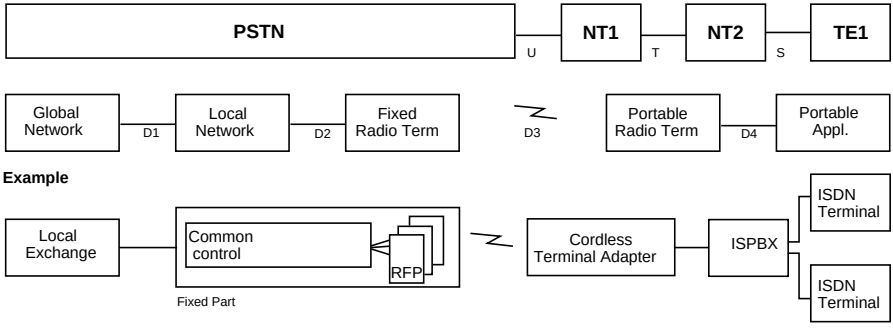


Figure 9.14: Radio local loop configuration for ISDN

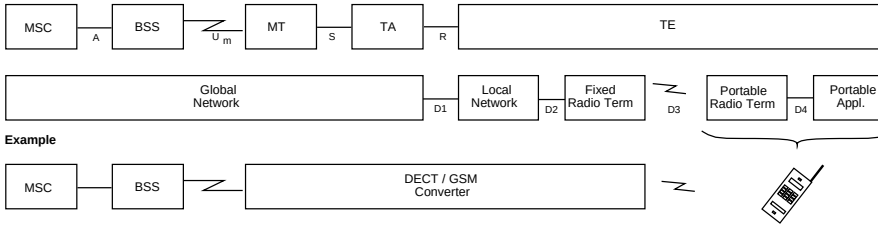


Figure 9.15: GSM-DECT configuration

9.3 The DECT Reference Model

The DECT reference model was designed in accordance with the ISO/OSI model (see Section 2.5). Because the DECT system incorporates the interface between communicating partners, only certain aspects of the applications-oriented layers are included in the standard, e.g., encryption.

The key functions of the DECT system correspond to the three lower layers of the ISO/OSI model: *Physical*, *Data Link* and *Network* layers. Because with DECT the quality of the transmission medium (radio) is continuously changing and channel access is a complicated function that must be carried out frequently, the data link layer has been divided into the two sublayers *Data Link Control* (DLC) and *Medium Access Control* (MAC). Figure 9.16 relates the DECT reference model to the corresponding ISO/OSI layers. Above the MAC layer the functions of the layers are grouped into two parts: the *Control plane* for signalling and the *User plane* for the transmission of user data. The control functions of the C-plane are processed in the network layer, whereas the data of the U-plane is passed on unprocessed.

9.3.1 An Overview of Services and Protocols

The DECT layers with their characteristics are briefly introduced below. A detailed description of the DECT layers is provided in Section 9.4.

9.3.2 Physical Layer (PHL)

The *physical layer* (PHL) is responsible for implementing the transmission channels over the radio medium. At the same time it has to share the medium with many other mobile stations that are also transmitting. Interference and collisions between communicating base and mobile stations are largely avoided thanks to the decentrally organized use of the available dimensions location, time and frequency (see Figure 9.17). Each dimension contains several possibilities for seizing a channel for interference-free transmission.

The TDMA (*Time-Division Multiple-Access*) method is the one used for the time dimension. Each station sets up its channel in any available time slot

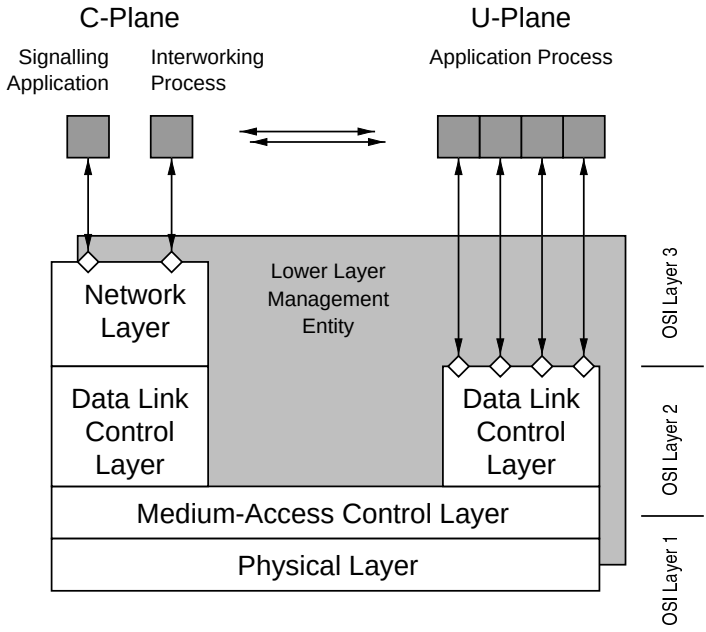


Figure 9.16: DECT reference model

and is then able to transmit in that time slot at a constant bit rate. Because of the TDD (*Time-Division Duplexing*) method the uplink and downlink of this channel are in slot pairs on the same frequency. Consequently a duplex transmission always occupies two time slots, which are separated from another by a fixed interval.

The FDMA (*Frequency-Division Multiple-Access*) method with 10 different frequencies is used for the frequency dimension. This means that each station is able to select and occupy a slot pair on any frequency for its transmission.

Owing to the attenuation of the propagation of the radio waves, frequencies and time slots can be reused at appropriate distances. These resources are occupied based on a dynamic procedure for channel selection that takes into account the local traffic load on the system.

Key technical data for the DECT system is listed in Table 9.1 (see also Table 8.2). With only 12 duplex voice channels in 1.73 MHz, i.e., 144 kHz/channel pair, DECT is very generous in its use of spectrum (GSM only requires 50 kHz/channel pair). However, because of its dynamic channel selection, the resulting small random reuse distances and the microcellular coverage, DECT provides a much higher capacity (Erl./km^2), producing almost unbelievably high values (10 kErl./km^2), especially in high-rise buildings because of the reuse distances at every two levels over the same base area.

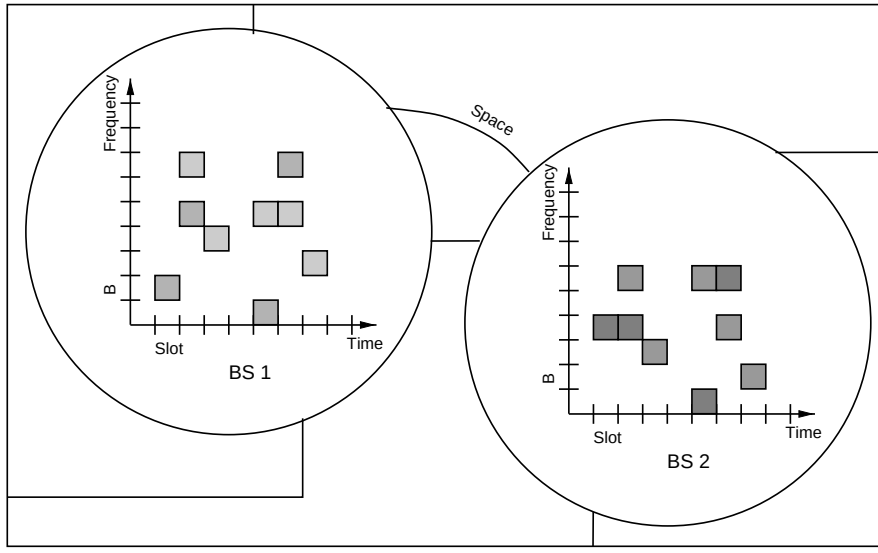


Figure 9.17: Three-dimensional use of spectrum

Table 9.1: Physical data on the DECT system

Frequency band	1880–1900 MHz
No. of carrier frequencies	10
Carrier spacing	1.728 MHz
Max. transmitter power	250 mW
Carrier multiplexing	TDMA
Basic duplex method	TDD
Frame length	10 ms
No. of slots (frame)	24
Modulation	GFSK with $BT = 0.5$/GMSK
Modulated overall bit rate	1152 kbit/s
Net data rate	32 kbit/s data (B-field) unprotected
for standard connections	25.6 kbit/s data (B-field) protected
	6.4 kbit/s signalling (A-field)

9.3.3 Medium-Access Control (MAC) Layer

The MAC layer (*Medium-Access Control (MAC) Layer*) (see Figure 9.16) is responsible for setting up, operating and releasing channels (*Bearer*) for the higher layers. The different data fields of the MAC protocol are protected by cyclical codes, which are used for error detection at the receiver. The MAC layer ensures that service-specific control data is added to each time slot.

The MAC layer comprises three groups of services:

BMC The *Broadcast Message Control Service* is offered on at least one physical channel in each cell, even if there is no user transmitting. This channel is called a *beacon channel*, and produces a continuous connectionless point-to-multipoint connection on the downlink in which a base station transmits its system-related data. This identifies the base station to the mobile unit. By evaluating the received signal, the terminal can determine the current channel quality at the same time.

CMC The *Connectionless Message Control Service* supports a connectionless point-to-point or point-to-multipoint service that can be operated bidirectionally between a base station and a mobile user.

MBC The *Multi-Bearer Control Service* offers a connection-oriented point-to-point service. An entity transmitting in one or both directions is able to support several bearers, thereby achieving a higher net data rate.

Each of these three services incorporates its own independent *service access point* (SAP), which links it to the next-highest layer and can combine several logical channels together.

9.3.4 Data Link Layer

The protocol stack directly above the MAC layer divides into two parallel parts. Similarly to the MAC layer, extensive error protection in the C-plane of the data link layer improves the reliability of the data transmission. Along with a point-to-point service, the C-plane offers a broadcast service to the network layer above it. The U-plane is responsible for processing user data on the radio section, with the spectrum of services ranging from the transmission of unprotected data with minimal delays, e.g., speech, to the provision of protected services with variable delays for data transmission. The required data rate of an existing connection can be changed at any time.

9.3.5 Network Layer

The network layer sets up, manages and terminates connections between users and the network. The U-plane in DECT has no role in the network layer and forwards all data unprocessed in a vertical direction. The C-plane carries out the signalling and is responsible for controlling data exchange. Five protocols based on the *link control entity* are provided for this purpose. Along with the call and connection entities, there is a *mobility management* service that carries out all the tasks necessary to support the mobility of mobile stations. In addition to the data for location area management, messages for authentication and encryption data are also transmitted.

9.3.6 Management of the Lower Layers

The management of layers 1–3 (*Lower Layer Management Entity* (LLME) (see Figure 9.16) contains procedures that affect several protocol layers. Processes such as generating, maintaining and releasing physical channels (*Bearers*) are initiated and controlled from this entity. Furthermore, the LLME selects from the physical channels which are available and evaluates the quality of the received signal.

9.4 Detailed Description of Services and Protocols

9.4.1 Physical Layer (PHL)

A physical channel is the physical transmission path between two radio units. Radio transmission places a great demand on transceivers to ensure that there is good reception of the usage signal. The receiver sensitivity for the required bit error ratio (BER) of 0.001 is −83 dBm (60 dBμV/m) [8]. This was increased to −86 dBm for applications run by public services [13]. In DECT a normal telephone connection requires two independent channels between the device end points on the transmission medium radio.

9.4.1.1 FDMA and Modulation Techniques

Because of FDMA access, the DECT system has the possibility of selecting from a number of frequencies in its channel selection.

It operates in the 1880–1900 MHz frequency band. Within this band 10 carrier frequencies are defined, the mid-frequency f_c of which can be calculated as follows:

$$f_c = f_0 - c \cdot 1728 \text{ kHz}, \quad \text{with } c = 0, 1, \ldots, 9 \text{ and } f_0 = 1897.344 \text{ MHz} \quad (9.1)$$

In an active state the deviation from the mid-frequency should be a maximum of ±50 kHz.

The modulation techniques used are either *Gaussian Frequency Shift Keying* (GFSK) with a bandwidth–time product $BT = 0.5$ or *Gaussian Minimum Shift Keying* (GMSK).

If a transmit signal is formed from two orthogonal bandpass signals with different mid-frequencies, this is referred to as *Frequency Shift Keying* (FSK). A Gaussian filter acting as a low-pass filter (GFSK) removes the high-frequency parts of the signal in order to keep the band of the signal's spectrum as small as possible. If the modulation index is 0.5 and the possibility of a coherent demodulation of the radio signal exists, this shift procedure is referred to as minimum-shift keying (MSK). With GMSK the Gaussian filter also comes into play. For cost reasons, receivers in DECT systems are usually not built with coherent demodulation/modulation [27].

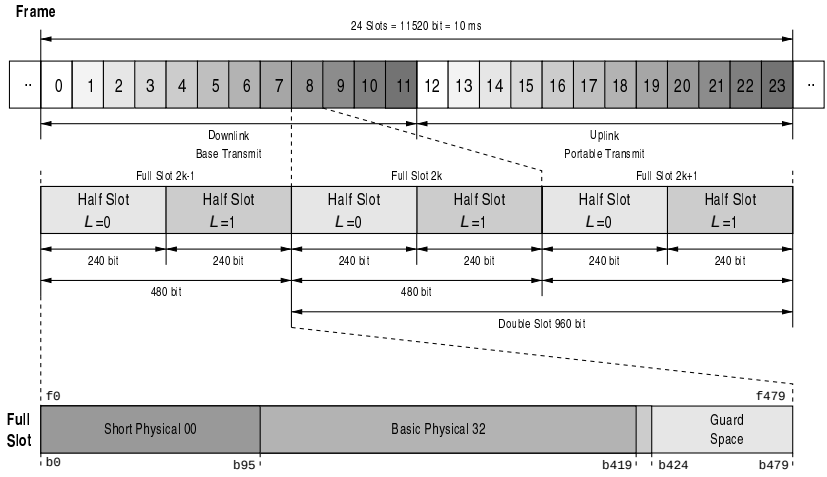


Figure 9.18: Time-multiplex elements of the physical layer

The transmission of a binary `1` in a DECT system produces a frequency increase of $\triangle f = 288$ kHz to $f_c + 288$ kHz. For the transmission of a `0` the frequency is decreased by $\triangle f$ to $f_c - 288$ kHz.

The standard does not provide for any equalizers. With a bit duration of 0.9 µs, waves that reach the receiver with a delay due to multipath propagation (see Figure 2.8) cause signal dispersion (see Section 2.1.7), which with a 300 m alternative path length already corresponds to the symbol duration and makes reception impossible even when sufficient signalling power exists. The literature recommends using 16 of the 32 synchronization bits in the S-field in accordance with Figure 9.19 to estimate the impulse response of a channel in the receiver, which equates to a good (but not one conforming to the standard) implementation of an equalizer [21].

9.4.1.2 TDMA Technique

Each station receives a protected, periodically recurring portion of the overall transmission rate of a frequency. According to Figure 9.18, the frame and time slot structure of the DECT system is explained in the physical layer.

The transmission capacity of each frequency is divided into 10 ms long periodically recurring frames, each of which has a length corresponding to the duration of 11.520 bits. This produces a gross frame transmission rate of 1152 kbit/s. A frame comprises 24 time slots, which are used as either full slots, double slots or half slots (see Figure 9.18).

In a normal *basic connection* the first 12 time slots are used for transmitting data from the base station to a mobile station (*downlink*), whereas the second part of the 24 slots is reserved for the direction from the mobile station to the base station (*uplink*). Since a duplex connection requires both an uplink and a

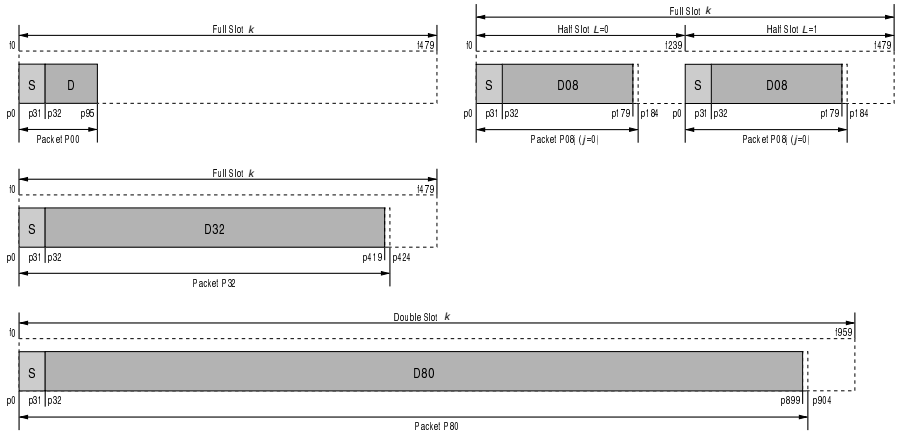


Figure 9.19: The different physical packets in the DECT standard

downlink connection, the DECT system uses a technique called *Time-Division Duplexing* (TDD). If the base station occupies slot k in order to transmit to the mobile unit, slot $k + 12$ is reserved for the mobile station to enable it to send data to the base station. This rigid system of allocation is abandoned when it comes to more complex *advanced connections*, where a generous use of time slots is allowed in each transmission direction.

Each of the 24 time slots has a length of 480 bits (416 µs) and can be used in accordance with the slot type (full, half, double). Different *physical packets* are based on this structure. Each physical packet contains a synchronization field S and a data field D. A physical packet is shorter than a time slot by one *guard period* to prevent an overlapping of packets from neighbouring time slots.

A P00 (*short physical packet*) is used for short connectionless transmissions in the beacon channel or for short messages. It comprises only 96 of the 480 bits and therefore provides a particularly large guard time in a slot. This is especially important at the start of a connection phase when a mobile station is not yet completely synchronized with the network and could cause interference to neighbouring time slots when it transmits a full slot.

A P08j message only requires a half slot for transmission, and therefore because of the decreased transmission rate ends up with double the number of available channels per frame; see Figure 9.19.

Because full slots are usually the ones preferred when a slot is requested, the packet P32 will be described here. The first 32 bits form the synchronization field S, which is used for clock and packet synchronization in the radio network. It consists of a 16-bit preamble, followed by a 16-bit long packet synchronization word. In the S-field the response from the mobile station contains the inverted sequence of the bit of the synchronization field sent by the base station.

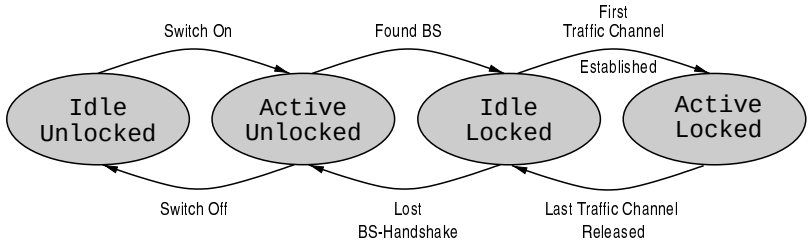


Figure 9.20: Operating states of a mobile station

The S-field is followed by the user data field D, which is 388 bits long. Some of the content of this field is evaluated by the MAC layer, and therefore will be covered in detail later (see Section 9.4.2.5).

The possibility of transmitting a so-called Z-field is offered by the D-field. This 4-bit long word contains a copy of the last 4 bits of the data field, which is also referred to as an X-field. A comparison of these two areas allows the receiver to establish whether a transmission is being disrupted because of errors in the synchronization of neighbouring DECT systems. These disruptions are referred to as *sliding collisions* within the system. A measurement of these disruptions results in the early detection of interference, and can be used as the criterion for an optimal handover decision. Packet P80 is the one with the highest data rate, and requires a double slot. User data rates up to 80 kbit/s can be achieved with this packet.

9.4.2 Medium-Access Control (MAC) Layer

9.4.2.1 Operating States of a Mobile Station

With reference to the MAC layer [9], a mobile station can find itself in one of the four states shown in Figure 9.20.

- `Active Locked` The synchronized mobile station has at least one connection to one or more base stations.
- `Idle Locked` The mobile station is synchronized with at least one base station. It currently does not have a connection, but is in a position to receive requests for connections.
- `Active Unlocked` The mobile station is not synchronized with any base station, and therefore cannot receive any requests for connections. It attempts to find a suitable base station for synchronization to enable it to move into the `Idle Locked` state.
- `Idle Unlocked` The mobile station is not synchronized with any base station, and is not in a position to detect base stations that are suitable.

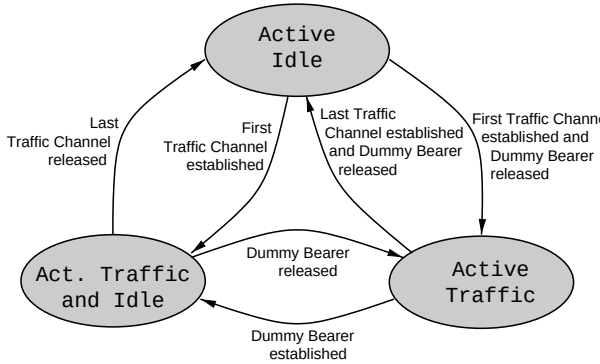


Figure 9.21: States of a base station

If a terminal is switched off, it is in the `Idle Unlocked` state. When it is switched on, a mobile station tries to transit to the `Idle Locked` state. It begins to search for a suitable base station with which to synchronize. If it is successful, a transition to the `Idle Locked` state is made. In this state a mobile station is able to receive or transmit requests for connection. As soon as the first traffic channel is set up, it changes over to `Active Locked` state. If the last channel is cleared after the termination of a connection, the mobile station returns to the `Idle Locked` state. If it loses its synchronization with the base station, it must change to the `Active Unlocked` state and seek a new suitable base station.

9.4.2.2 Operating States of a Base Station

A base station can find itself in one of four states. The `Inactive` state in which a base station is switched off is not shown in Figure 9.21.

- `Inactive` The base station is switched off and can neither receive nor transmit messages.
- `Active Idle` The base station is not operating a traffic channel, and therefore radiates a dummy bearer that the mobile receivers are able to detect when observing the physical channels.
- `Active Traffic` The base station operates at least one traffic channel. The dummy bearer is no longer supported.
- `Active Traffic and Idle` The base station maintains a dummy bearer in addition to at least one traffic channel (*traffic bearer*).

In the basic state `Active Locked`, the base station transmits a dummy bearer in order to enable mobile terminals to synchronize with its frame and slot clock. If a traffic bearer is established, the base station moves into `Active`

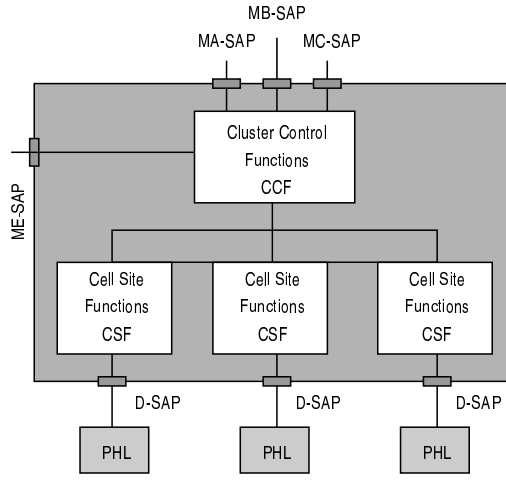


Figure 9.22: Classification of MAC services

`Traffic` state. The dummy bearer can then be eliminated. The transition on the opposite side takes place after the last traffic channel has been released. If a dummy channel is required when the traffic bearer is transmitted, the base station can move into `Active Traffic and Idle` state. Again the dummy bearer can be retained when the first traffic channel is set up. A transition is then made from `Active Idle` to `Active Traffic and Idle`.

9.4.2.3 Cell and Cluster Functions

The MAC layer functions have already been covered in the overview in Section 9.3.3. This layer is used to set up and manage the traffic channels (bearers) requested from the LLME and, when requested, to release them. The control information, which is introduced through different service access points in the MAC layer, is multiplexed and added to the actual user data in each time slot.

The different services of the MAC layer are divided into two groups (see Figure 9.22). The *cluster control functions* in the upper area are linked to the data link layer through the three service access points MA, MB and MC. The cell site functions in the lower part coordinate the transition to the physical layer. The two groups offer the following individual functions:

Cluster Control Functions (CCF) These functions control a cluster of cells. Each logical cluster of cells contains only one CCF, which controls all the cell site functions (CSF). Three different independent services are available within this cluster:

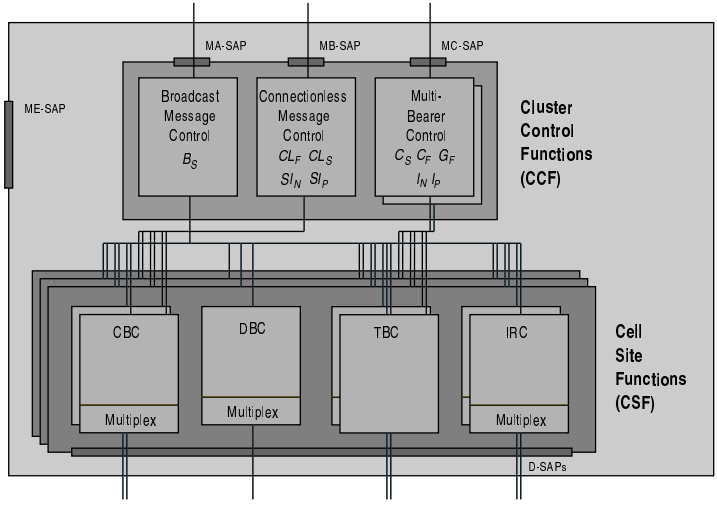


Figure 9.23: Overview of services and channels in the MAC layer

Broadcast Message Control (BMC) This function only exists once in each CCF, and distributes the cluster broadcast information to the respective cell functions. The BMS supports a number of connectionless point-to-multipoint services, which are directed from a base station to the mobile station. The BMC works with any type of traffic channel. An important service is the paging of mobile stations.

Connectionless Message Control (CMC) All information that concerns the connectionless service is usually controlled by one CMC in each CCF. In addition to transmitting information from the control level of the DLC layer, the CMC also processes user data from the so-called U-plane. The services can be operated in both directions.

Multi-Bearer Control (MBC) This service comprises the management of all data that is exchanged directly between two corresponding MAC layers. An MBC capable of organizing several traffic channels exists for each connection-oriented point-to-point connection.

Cell Site Functions (CSF) These functions appear below the CCF services in the MAC layer, and stand in for the respective cell. Therefore several CSFs are controlled by each CCF. The following cell-oriented services are differentiated (see Figure 9.23):

Connectionless Bearer Control (CBC) Each connectionless bearer within the CSF is controlled by its own CBC.

Dummy Bearer Control (DBC) Each CSF has a maximum of two dummy bearers to provide a beacon function for the synchronization of the mobile stations in the event that no user connection exists in the cell.

Traffic Bearer Control (TBC) An MBC must request a TBC for a duplex connection.

Idle Receiver Control (IRC) This service controls the receiver in a cell when no connection is being maintained to a user; a cell can have more than one receiver, and consequently a related number of IRC services.

9.4.2.4 Service Access Points

The cluster and cell-oriented functions include several service access points (SAP), which can be used by the MAC layer entities to communicate with the next-highest OSI layer (*data link control layer*) and the next-lowest OSI layer (*physical layer*) (see Figure 9.22). The following SAPs are available between the CCF and the DLC layers:

- MA-SAP
- MB-SAP
- MC-SAP

Each cell-specific service has its own D-SAP for access to the physical layer. The MAC layer has a separate SAP, the ME-SAP, to support the lower layer management functions. This SAP is not formally specified, and therefore has no logical channels.

MA-SAP Information from the *Broadcast Message Control Service* is transmitted to the DLC layer over this access point. Data from the B_S channel as well as control data that controls the data flow on the B_S channel is transmitted. This logical *Higher-Layer Broadcast Channel* supports a connectionless simplex broadcast service from the base station to the mobile station.

MB-SAP This access point links *Connectionless Message Control* with the DLC layer, and contains four logical channels:

- CL_F
- CL_S
- SI_N
- SI_P

The control channels CL_F and CL_S support a connectionless duplex service between the base station and the mobile terminal. A continuous service exists from the base station to the mobile station but not in the opposite direction. The quantity of data permitted on the CL_S channel is 40 bits, which corresponds to the segment length of this channel. The fast CL_F channel has a permitted segment length of 64 bits, and the quantity of data can amount to four times the segment length.

The information channels SI_N/SI_P offer an unprotected/protected simplex service from the base station to the mobile station.

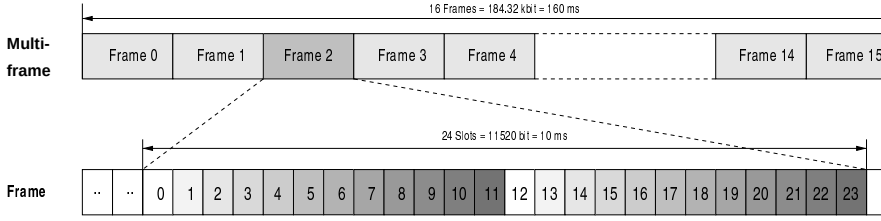


Figure 9.24: DECT multiframe organization

MC-SAP The multi-bearer control unit is linked to the DLC layer over the MC-SAP. Five logical channels are available for the transmission of information on data flow control and establishing, maintaining and releasing MAC connections:

- C_S
- C_F
- G_F
- I_N
- I_P

The control channels C_S and C_F offer two independent connection-oriented duplex services. The maximum throughput for the connection of a slow C_S with a segment length of 40 bits is 2 kbit/s. With a data segment length of 64 bits, the fast C_F control channel achieves a throughput of 25.6 kbit/s with full slots. The information of both control channels is supplied with a CRC checksum (*Cyclic Redundancy Check*) that provides error detection and correction through retransmission with an ARQ technique.

One of the two information channels I_P or I_N is used with each transmission to ensure that the higher layers are offered an independent connection-oriented duplex service. The I_N channel is for voice transmission, with the MAC layer supplying 4 bits (X-field) of error detection protection for the information. I_P channels are planned for the transmission of data with error-detection or error-correction coding.

The G_F channel is a connection-oriented simplex service with a data segment length of 56 bits and is used by the U-plane of the DLC layer. The MAC layer provides error detection for this channel.

9.4.2.5 MAC Multiplex Functions

Multiframe structure In the MAC layer the TDM frame structure implemented from the physical layer (see Section 9.4.1) is superimposed logically with a multiframe structure. A multiframe is composed of 16 individual frames (see Figure 9.24), and normally begins with the first half of the frame 0, used by the base station. The end of a multiframe is used by the mobile terminal in the last half of frame 15. The number of the current multiframe is transmitted in at least each eighth multiframe, and is used for encryption purposes. The frame number is recognized by frame 8 because this is where the logical channel Q_T is transmitted from the base station to the mobile station (see Figure 9.26).

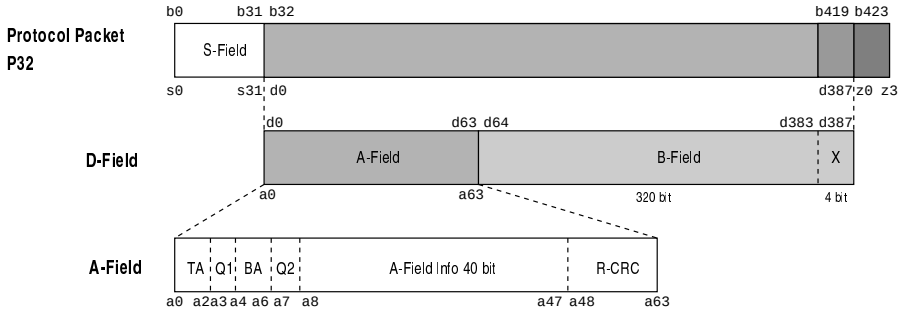


Figure 9.25: DECT time-multiplexing: D32-field

D-field Bit mapping, the compilation of the D-field for subsequent transmission to the physical layer, is based on fixed rules. As already described in Section 9.4.1.2, the length of a D-field depends on the type of transmission requested. If a P32 transmission is requested, the length of the D-field is 388 bits. A different D-structure is used for P00, P08j and P80 operations.

The D-frame of a P32 packet comprises two parts, as shown in Figure 9.25. The A-field is 64 bits long and is used for the continuous transmission of control information. The B-field is available for actual user data, and in full slot operation is 324 bits in size.

The individual parts of the A and B-fields are explained below:

A-field The control field contains three sections (see Figure 9.25). The A-field information of 40 bits is joined to the header of the A-field, which has a length of 8 bits. An R-CRC field with 16 bits for protection of the control data forms the connection.

TA (`a0...a2`) The 3-bit long TA field at the beginning of the header indicates the type of A-field information (`a8...a47`). There are five different logical channels, one of which always transmits data in the A-field. Here a distinction is made between the internal MAC channels N_T, Q_T, M_T and P_T and the C_S channel (for the control data of the higher layers).

Q1 (`a3`) In connection-oriented transmission the Q1 bit in conjunction with duplex bearers carries out the quality control of a channel and serves as the handover criterion. In services which contain error correction it can be used for flow control.

BA (`a4...a6`) This part describes the qualities of the B-field. In addition to normal protected and unprotected information transmission (*U-type*, I_P or I_N), expanded signalling (*E-type*, C_F or C_L) is also possible in certain cases.

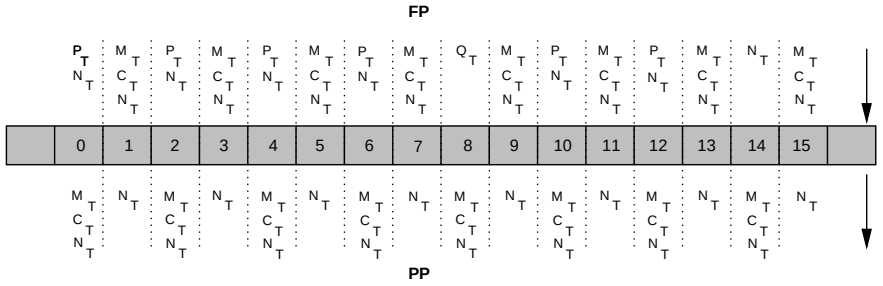


Figure 9.26: Priority of control information in a multiframe cycle

Q2 (a7) Like the Q1 bit, the Q2 bit is designated for the quality control of a connection. A combination of Q1 and Q2 constitutes a handover criterion.

A-field info (a8...a47) Internal MAC messages can be transmitted within the 40-bit long tail field. This involves sending out different control information in consecutive time slots using an *E-type* multiplexer.

R-CRC (a48...a63) The MAC layer protects the logical channels of the A-field with a *cyclic redundancy check* (CRC). This involves calculating and transmitting 16 redundancy bits, which with up to

- 5 independent errors
- bursty errors up to a length of 16
- error patterns with an odd number of errors

can be detected in the A-field [26]. An error-correction mechanism is not provided in the MAC layer.

The internal control data of the MAC layer is contained in the A-field information. One of the following options can be selected during multiplexing (see Figure 9.26):

N_T **Identities Information** Using its *Primary Access Rights Identifier* (PARI) and its *Radio Fixed Part Number* (RPN), the MAC layer of the base station generates its own *Radio Fixed Part Identity* (RFPI), which is transmitted to the terminal.

Q_T **System Information and Multiframe Marker** This channel, used once in each multiframe, is only transmitted by the base station, and therefore is used for indirect synchronization on the multiframe cycle. Information on the technical structure of the base station as well as on the current connection is provided.

P_T **Paging Information** This channel is the only one that a mobile station can even receive when it is in the Idle Locked state (see

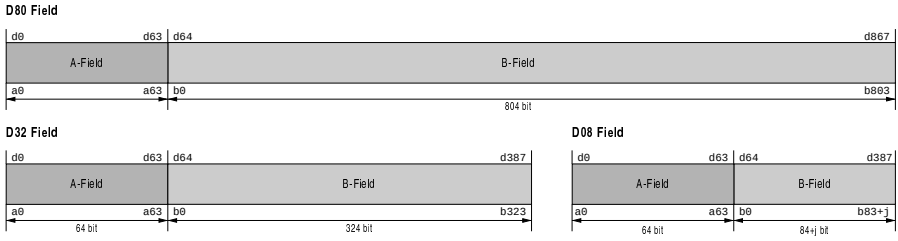


Figure 9.27: D-fields for the different physical packets

Figure 9.20), and contains the broadcast service (B_S) of a base station. In addition to the identification number of the base station, the terminal receives important MAC layer information. For example, blind slot data that evaluates the quality of the current bearer is transmitted and suggestions for other qualitatively good channels are passed on. In response to a Handover_Request, the handover areas approved by the base station are transmitted back.

M_T **MAC Control Information** Administrative tasks such as connection setup, maintenance and termination and handover requests are carried out on this channel. A Wait command is sent if there is a delay in setup. A reciprocal exchange of the channel list can accelerate the selection of the best channel for transmission.

C_T **Control Information Higher Layers** Either C_L or CS_L information from the higher layers is transmitted on this channel. In other words, this is not an internal MAC channel.

Multiplexing of control channels during transmission is carried out according to a specific plan. Different control information for the A-field, which is transmitted on the basis of a priority list, is provided in each multiframe (see Figure 9.26). The base station always sends its system information (Q_T) in the eighth frame, whereas for the mobile station each uneven frame is reserved for the N_T channel. If a control channel with the highest priority attached to it remains unused in a frame, control information with a lower priority can be sent.

B-field With physical packet P32 the 324-bit long B-field, for which a protected and an unprotected format exist, follows the control field (A-field) in the D-field (see also Figure 9.27). In an unprotected transmission of data, as in the transmission of voice, 320 effective usage bits from the B-field are required for a data rate of 32 kbit/s. The remaining 4 bits (X-field) are used for error protection by the means of CRC. In a protected B-field format the 4 bits are kept for error protection, whereas the 320 effective data bits are divided into four blocks. The usage bits within each block are reduced to 64 so that an R-CRC check sequence

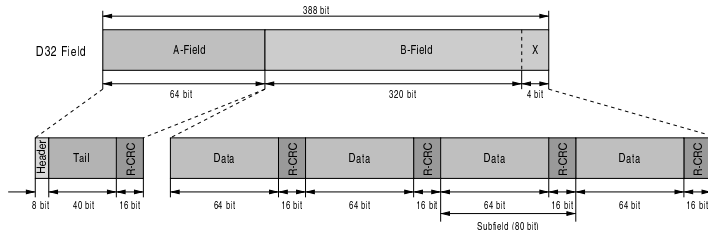


Figure 9.28: Elements of the A and B-fields

can be formed with the remaining 16 bits. The net data rate reduces to 25.6 kbit/s (see Figure 9.28).

9.4.2.6 MAC Services

The different MAC services produced from the combination of different physical packets and the option for protected or unprotected transmission are listed below. These can be carried out as symmetrical or as asymmetrical services. With asymmetrical services the rigid separation between uplink and downlink is eliminated. Thus the service from a base station to a mobile station can use a slot in the first part of a frame as well as the corresponding slot in the second part of the frame for the downlink. It is different with services that offer a guaranteed throughput or guaranteed bit-error ratio, which then results in a variable throughput. In this case the MAC layer uses an ARQ protocol. The services of the MAC layers are listed individually in Tables 9.2 and 9.3.

9.4.2.7 Types of Bearers

MAC bearers are elements that are created by the *cell site functions* (CSF) and always correspond to a service entity in the physical layer. The following is a list of the different bearer types:

Simplex Bearer This type of bearer is used to produce a physical channel in one direction. A differentiation is made between long and short simplex bearers. Whereas the short bearer only contains an A-field, the long one also transmits a B-field next to the A-field. For example, *dummy bearer control* (DBC) (see Section 9.4.2.3) reserves a simplex bearer sent by the base station for the transmission of broadcast information.

Duplex Bearer A pair of simplex bearers transmitting in the opposite direction on the physical channels is called a duplex bearer. The two bearers use the same frequency and are time-deferred by half a frame (slot pair). A *traffic bearer controller* (TBC) uses a duplex bearer, e.g., for a traffic channel between the base station and the mobile station.

Table 9.2: Partial list of symmetrical MAC services[a]

S_T	I-channel cap. [kbit/s]	B-field multipl. scheme	Error detection	Error correction	Delay [ms]
2d	$k \cdot 80$	(U 80a, E 80)	—	—	15
2f	$k \cdot 32$	(U 32a, E 32)	—	—	15
2d	$8 + j/10$	(U 08a, E 08)	—	—	15
3d	$k \cdot 64.0$	(U 80b, E 80)	×	—	15
3f	$k \cdot 25.6$	(U 32b, E 32)	×	—	15
3d	6.4	(U 08b, E 08)	×	—	15
4d	$\leq k \cdot 64$	(U 80b, E 80)	×	×	Var.
4f	$\leq k \cdot 25.6$	(U 32b, E 32)	×	×	Var.
4d	$\leq$ 6.4	(U 08b, E 08)	×	×	Var.

[a] S_T = Service type: xd = type x double slot, xf = type x full slot, xh = type x half slot

Table 9.3: Partial list of asymmetrical MAC services

S_T	I-channel cap. [kbit/s]	B-field multipl. scheme	Error detection	Error correction
5d	$k \cdot 80$	(U 80a, E 80)	—	—
5f	$k \cdot 32$	(U 32a, E 32)	—	—
6d	$k \cdot 64.0$	(U 80b, E 80)	×	—
	$m \cdot 64.0$	(U 80b, E 80)	×	—
6f	$k \cdot 25.6$	(U 32b, E 32)	×	—
	$m \cdot 25.6$	(U 32b, E 32)	×	—
7d	$\leq k \cdot 64.0$	(U 80b, E 80)	×	×
	$\leq m \cdot 64.0$	(U 80b, E 80)	×	×
7f	$\leq k \cdot 25.6$	(U 32b, E 32)	×	×
	$\leq m \cdot 25.6$	(U 32b, E 32)	×	×

Double Simplex Bearer A pair of long simplex bearers transmitting in the same direction on two physical channels is called a double simplex bearer. Similarly to a duplex bearer, the two bearers should operate in one slot pair on the same frequency. This type of bearer only occurs in multibearer connections, and is also used for asymmetrical transmission.

Double Duplex Bearer A double duplex bearer consists of a pair of duplex bearers that is part of a shared MAC connection. Each of these duplex bearers is produced by a TBC. The two TBCs are controlled by an MBC.

9.4.2.8 Types of Connections

Connection-oriented entities exist in addition to the connectionless services that include broadcast services. Each *multibearer control* unit in the MAC layer is responsible for the maintenance of a connection. It controls one or more *traffic bearer control* entities, which are used in the administration of the bearers. A differentiation is made between *advanced connections* and *basic connections*:

Basic Connections A basic connection does not have a common connection number that is known to the base station and to the mobile station. Consequently only one basic connection can exist between a base station and a mobile terminal. It consists of a single duplex bearer. It is possible for two basic connections that serve the same DLC connection to exist together for a short time when a switch is being made in physical channel (handover).

Advanced Connections Because, unlike basic connections, advanced connections use a common number, several connections can exist between two stations. The individual bearers are assigned a logical number to allow a differentiation within a connection in the MAC layer.

Physical Connections Physical connections are not supported by MAC services (see Tables 9.2 and 9.3). They are used for non-standardized data transmission (see Section 9.4.3.8).

When a connection is being made, all the bearers requested from the DLC must be set up within 3 s (`T200`: *Connection Setup Timer*) or else the connection is considered aborted. Additional bearers can be set up or released if necessary during an ongoing connection in order to increase or reduce the overall transmission capacity.

Advanced connections can be either symmetrical or asymmetrical. With symmetrical connections the same services and number of bearers are used in both directions.

9.4.2.9 Connection Setup

Connection-oriented procedures within the MAC layer use two point-to-point connections: *connection* and *bearer*. Only the pure connection is apparent to the data link layer. The bearers that each connection uses for a transmission are managed within the MAC layer and are not apparent to the higher layers. Connection setup is usually initiated by a mobile terminal (*portable-initiated*). If there is a call from the fixed network, the mobile station must first be paged so that it is aware of it and can initiate the setup procedure.

Connection Setup The sequence of connection setup can be explained with a description of the MAC layer primitives (see Figure 9.29). The higher layers

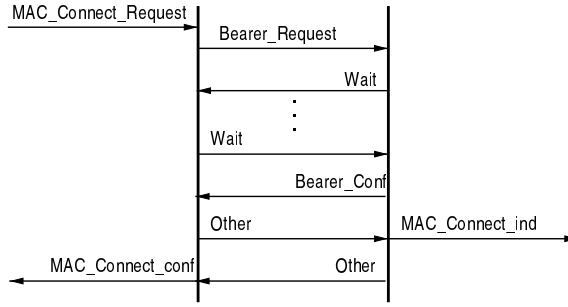


Figure 9.29: Connection setup of a basic connection

(DLC layer) initiate connection setup (MAC-Connect_Request) in the MAC layer. In addition to a MAC connection endpoint identifier (`MCEI`) that applies as the reference address for all subsequent primitives, this primitive contains a parameter that specifies the requested service. If the MAC layer is not able to provide the requested services, it sends a Disconnect_request to the DLC layer.

The instance of an MBC entity (*multibearer control*) produced by the request receives permission from the *lower-layer management entity* (LLME) to set up a connection between the base station and the mobile station. Depending on the service requested, either a basic or an advanced connection is set up.

Bearer Setup If an MBC was established through connection setup in the MAC layer of the terminal, an attempt is made to set up the bearers required for the service. First the mobile station must be in the state `Idle Locked`, in other words, it must be synchronized with at least one base station in the cluster that will allow it to set up its connections. The MBC produces new instances of *traffic bearer control* (TBC) entities, which take over the required bearer setups. A bearer setup is based on the following process (see Figure 9.29):

1. The mobile station sends a Bearer_Request for a selected channel to a known base station. It uses the *first transmission* code within the 40 bits of the A-field.

2. The base station receives the request without any errors, and establishes a new TBC in its MAC layer.

3. The TBC in the base station requests the address of a supporting instance of an MBC entity from the lower-layer management entity.

4. If the base station is not yet ready to transmit a confirmation of the setup to the mobile station, it sends a Wait command. The mobile terminal receives this command and likewise responds with Wait.

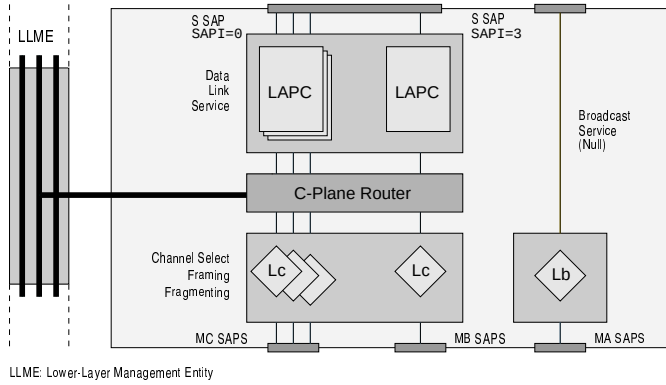


Figure 9.30: Control plane of the DLC layer

5. After completion of the protocol process on the fixed side, the base station sends a confirmation (Bearer_Conf).

6. The mobile station receives the confirmation and directly sends Other in the next frame.

7. The base station receives the Other message, and likewise responds immediately with this primitive.

8. The mobile station receives the Other message, and the TBC informs its MBC that the setup has been successful (Bearer_Established_ind).

If an error occurs at any time during the connection setup, the setup attempt is discontinued and the setup procedure is then repeated. A maximum of 10 setup attempts (N200) not exceeding the connection setup timer (T200 = 3 s) is permitted.

9.4.3 Data Link Control Layer (DLC)

The DLC layer [10] is divided into a U-plane and a C-plane (see Figure 9.16). The C-plane constitutes the control layer for all internal DECT protocol processes relating to signalling (see Figure 9.30).

It provides two independent services:

- Data link services (LAPC+Lc)
- Broadcast services (Lb)

The U-plane controls the transmission of user data and offers the following services, which are accessible over its own service access points (see Figure 9.31):

- LU1 *TRansparent UnProtected service* (TRUP)

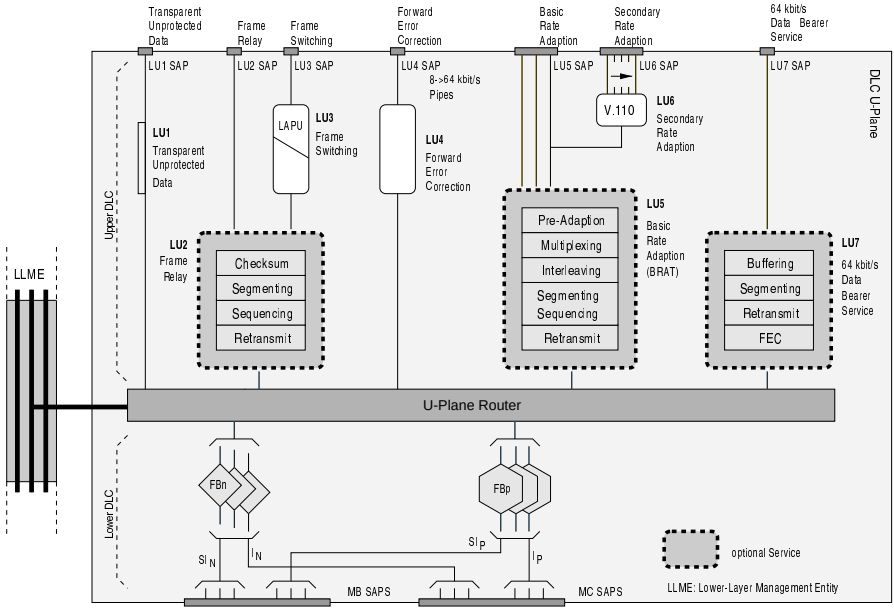


Figure 9.31: User plane of the DLC layer

- LU2 *Frame RELay service* (FREL)
- LU3 *Frame SWItching service* (FSWI)
- LU4 *Forward Error Correction service* (FEC)
- LU5 *Basic RATe adaption service* (BRAT)
- LU6 *Secondary RATe adaption service* (SRAT)
- LU7 *64 kbit/s data bearer service*
- LU8-15 reserved for future services
- LU16 *ESCape* (ESC) for services other than standardized services

The specifications for services LU3 and LU4 have not yet been finalized. Subordinate frame structures that allow direct mapping of the MAC services are available for each of the LUx services (see Tables 9.2 and 9.3).

Each LUx service can occur in different classes of transmission:

Class 0 No LUx transmission repeat and no sequential control in the DLC layer. This means that error messages from the MAC layer are forwarded to the higher layers.

Class 1 No LUx transmission repeat; the DLC layer distributes the DLC frames in the correct sequence.

Class 2 Variable throughput with LUx transmission repeat.

Class 3 Fixed throughput with LUx transmission repeat.

The LUx services are described below.

9.4.3.1 Transparent Unprotected Service LU1

The LU1 service can only occur as a *Class 0* service, and is the most basic of the services. Although envisaged for voice connections, it can also be used for data services. No error protection is provided. FU1 is the subordinate frame service.

9.4.3.2 Frame Relay Service LU2

The LU2 service is a protected service for the transmission of data blocks (*frame relay*) that is accessed through the LU2-SAP. It operates in a generic field of user data, which is transferred in and out of the DLC U-plane as service data units (SDU).

LU2 supports the reliable transport of the SDUs and also respects the SDU boundaries. Three basic procedures are supported:

1. Addition of a checksum to each service data unit (SDU).
2. Distribution of the resultant data-SDU+checksum to data fields of protocol data units (PDU).
3. Peer-to-peer transmission of these PDUs.

A distinction is made between external and internal blocks (frames) (see Figure 9.32):

- SDUs are external blocks.
- PDUs are internal blocks.

Checksum procedures The 16-bit long checksum supports error detection for an entire SDU frame (see Figure 9.33).

A defective SDU is not retransmitted. The user can designate external error-handling protocols.

9.4.3.3 Frame Switching Service LU3

This service should be mentioned here, but the standardization for it has not been completed.

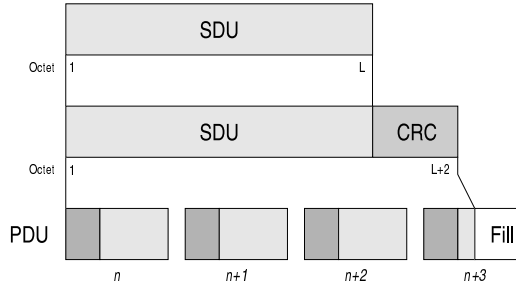


Figure 9.32: SDU division into PDUs

8	7	6	5	4	3	2	1	Bit
		Checksum Octet X						Frame length 1
X8	X7	X6	X5	X4	X3	X2	X1	
		Checksum Octet Y						Frame length 2
X8	X7	X6	X5	X4	X3	X2	X1	

Figure 9.33: Field format of checksum

9.4.3.4 Service with Forward Error Correction LU4

The specifications for this service have also not yet been completed. It will include forward error correction and possibly also an ARQ entity.

9.4.3.5 Data Rate Adaptation Service LU5

The service LU5 is being introduced to support synchronous data streams with fixed data rates (64, 32, 16 and 8 kbit/s). A differentiation is made between protected and unprotected transmission. The protected service offers considerably more reliable transmission.

The difference lies mainly in the use of the different MAC services: I_P services are used for the protected service and I_N services for the unprotected one. This produces differences in the combination of MAC services for the data adaptation service, which justifies the separate treatment of protected and unprotected services. In general, the principle of these services can be described as follows. The service supports up to three independent data channels. There is only a limited combination of different possible data rates. Rate adaptation is carried out for the individual data connections. These data connections are then multiplexed onto a channel. An option is offered here for interleaving the data. The channel is segmented into individual frames that are provided with control information and transmitted over the radio channel. The entire process is performed in reverse on the receiver side so that up to three independent data channels are again available at the service access point.

Table 9.4: Synchronous and asynchronous rate adaptation in the LU6 service

Input data	Output data rate [kbit/s]						
	Synchronous				Asynchronous		
rate [kbit/s]	8	16	32	64	8	16	32
0.05					×		
0.075					×		
0.11					×		
0.15					×		
0.2					×		
0.3					×		
0.6	×				×		
1.2	×				×		
2.4	×				×		
3.6					×		
4.8	×				×		
7.2		×				×	
9.6		×				×	
12			×				×
14.4			×				×
19.2			×				×
48				×			
56				×			

9.4.3.6 Supplementary Data Rate Adaptation Service LU6

The LU6 service can only operate in combination with LU5. It offers data adaptation for terminals that comply with the V-series. This service converts data rates in accordance with CCITT Recommendation V.110. The rate adaptations executed in the LU5 service are presented in Table 9.4.

9.4.3.7 64 kbit/s Data Service LU7

This service was specifically developed to support the 64 kbit/s data service of ISDN. Since ISDN fixed networks have a lower bit-error ratio than DECT radio connections (with ISDN a BER $\leq 10^{-6}$ is expected; with DECT a handover is initiated at a BER $= 10^{-3}$), the aim of the protocol is the reduction of bit-error ratio. This is accomplished through a combination of forward error correction (FEC) with an RS code and ARQ procedures. An increased data rate and buffer storage are necessary in order to maintain the ISDN data rate of 64 kbit/s. Realistically it is possible to transmit with a net data rate of 64 kbit/s or 72 kbit/s; the buffer storage produces a delay of 80 ms (see Figure 9.34).

The transmit buffer contains the last eight transmitted blocks. The receive buffer stores the arriving (maximum eight) blocks, thereby producing the time delay of 80 ms. If a defective block is received, it is rerequested and for eight

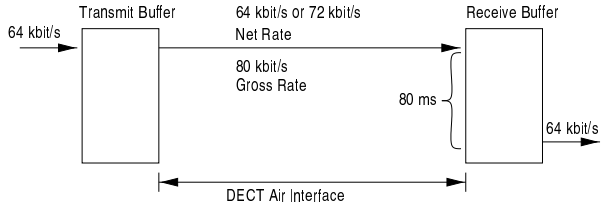


Figure 9.34: Implementation of ISDN data rate through error handling

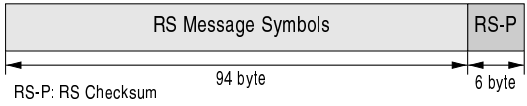


Figure 9.35: Content of a double slot B-field from the standpoint of BS coding

transmit/receive cycles there is a possibility to receive the block again. A system of counters ensures that the block is written at the correct place in the receive buffer. A temporary change to the increased transmission rate of 72 kbit/s compensates for the time lost through the repeated transmission.

Details of the algorithms as they are specified in the standard are discussed below.

The data fields The B-field of a double slot (see Figure 9.35) uses a (100,94) Reed–Solomon code with 94 bytes for the ARQ mechanism and 6 bytes for the checksum.

The ARQ mechanism divides the 94 message symbols into three groups (control, information and checksum, CS); see Figure 9.36.

The checksum field contains a checksum which the ARQ mechanism used to determine whether forward error correction was successful. The information field contains 90 bytes of user data when there is a 72 kbit/s transmission format; with a 64-kbit/s transmission format the last 10 bytes are filled with zeros (see Figures 9.37 and 9.38).

The control field contains the counters and format control parameters important for the ARQ mechanism (see Figure 9.39).

The send and receive buffer storage and status variables $V(O)$, $V(R)$, $V(S)$, $V(A)$ and $V_i(T)$ can be seen in Figure 9.40. These regulate the ARQ process by indicating which frame has been sent, received or acknowledged and from

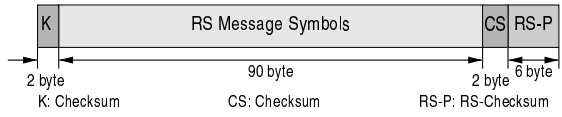


Figure 9.36: Content of an RS code word

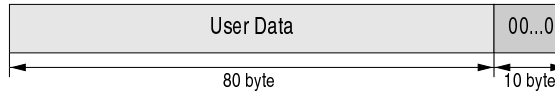


Figure 9.37: ARQ information field for 64 kbit/s user data rate

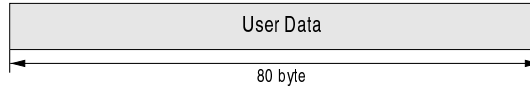


Figure 9.38: ARQ information field for 72 kbit/s user data rate

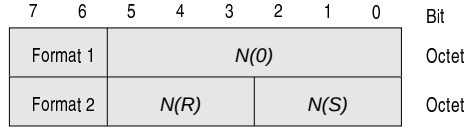


Figure 9.39: Content of ARQ control field

which storage position in the send storage read should take place or at which storage position in the receive storage write should take place. The parameters $N(O)$, $N(R)$, $N(S)$ transmitted in the ARQ control field and formats derived from the status variables are used in the communication between ARQ entities.

ARQ Control Fields

Format field The transmission format is coded with a total of 4 bits that are divided over two 2-bit fields (format 1, format 2); see Figure 9.39. The receiver must be informed of which transmission format is being used so that it can interpret the last 10 bytes as user data or zeros (see Figure 9.37).

ReTransmit requests (RTR) are sometimes also sent in these fields. Table 9.5 shows the significance of the bits.

The Offset variable V(O) indicates how many additional bytes (in units of 10 byte) are still required in order to refill completely the receive buffer of the partner instance. It also determines the respective transmission format (64 kbit/s or 72 kbit/s).

When a frame is retransmitted, $V(O)$ is increased by 8 (corresponding to the deficit of 80 bytes in the occupation of the receive buffer to be compensated for). Retransmission is only permitted as long as $V(O) \leq 48$. If $V(O)$ is greater then retransmission is not possible because the data requested is no longer available in its entirety in the transmit buffer and, furthermore, would no longer have a chance of arriving in time at

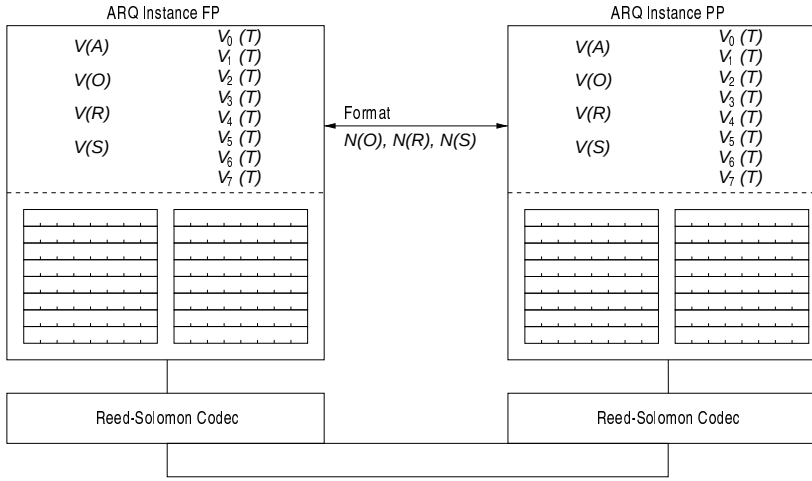


Figure 9.40: Components of the ARQ mechanism

Table 9.5: Significance of bit values in ARQ format fields

	Format 1		Format 2		
Bit	8	7	8	7	Significance
	0	0	0	0	Format 64 kbit/s
	0	0	0	1	Format 64 kbit/s, RTR
	0	1	0	0	Format 72 kbit/s
	0	1	0	1	Format 72 kbit/s, RTR

the partner entity. Instead, the next frame waiting to be sent in the transmit buffer is transmitted.

If $V(O) = 0$ then transmission is at the 64 kbit/s format and $V(O)$ is not altered. If $V(O) > 0$, then the format of 72 kbit/s is used. Because of the additional 10 bytes of user data also being transmitted, $V(O)$ can be reduced by 1 if transmission is successful, which corresponds to the reduction by 1/8 of the missing $8 \cdot 10$ bytes in the receive buffer of the partner entity.

The time variables $V_n(T)$ indicate the position in the transmit buffer of the entity and in the receive buffer of the partner entity for the last eight transmitted blocks. When the transmission is repeated, the value $V_n(T)$ of the requested block is transmitted as $N(O)$ to inform the partner entity of the position in the receive buffer where it must write the block.

The offset number N(O) gives the position in the receive buffer in which the block is to be written. *N(O)* here corresponds to the distance of the last

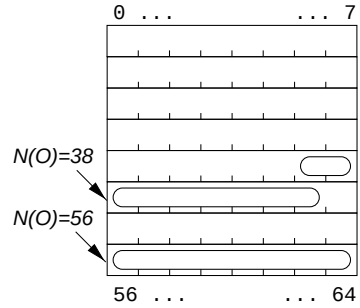


Figure 9.41: Relationship between $N(O)$ and the position of a block to be written in storage

bit from the start of storage in units of 10 bytes. Figure 9.41 shows examples of two different *N(O)s*.

The send status variable V(S) is the sequence number for each new initial transmission of a block. It is increased modulo 8 by 1 and therefore can assume values between 0 and 7. $V(S)$ is the send sequence number $N(S)$ of the next block waiting to be transmitted.

The acknowledgement status variable V(A) identifies each block being acknowledged by the partner entity. Here $V(A) - 1$ is the send sequence number $N(S)$ of the last block to be acknowledged by the partner entity.

The send sequence number N(S) is the sequence number of the transmitted block. For an initial transmission it is set at $V(S)$, and for a repeated transmission it is set at $V(A)$. $N(S)$ is transmitted as a binary-coded number (3 bits).

The receive status variable V(R) contains the send sequence number $N(R)$ of the next block that is expected. If an error-free block with $N(S) = V(R)$ is received then $V(R)$ is increased by 1, and by an additional 1 for each following error-free block in the receive buffer.

The receive sequence number N(R) indicates to the receiver the send sequence number of the next expected block. $N(R)$ is set at $V(R)$ before a block is sent, and consequently acknowledges all frames including $N(R) - 1$ to the partner entity.

The ARQ information field contains 80 or 90 bytes of user data, depending on the transmission format (64 kbit/s or 72 kbit/s); with the format 64 kbit/s the last 10 bytes are zero.

The ARQ checksum field CS (16 bits) is calculated as follows (see Figure 9.36):

$$\text{checksum}(x) = \text{EK}(a(x) + b(x))$$

with EK( ) representing the ones complement (bit inversion) and

$$\begin{aligned} a(x) &= x^{736}(x^{15} + x^{14} + x^{13} + x^{12} + x^{11} + x^{10} + x^9 + x^8 + x^7 + x^6 \\ &\quad + x^5 + x^4 + x^3 + x^2 + x + 1)\text{mod}(x^{16} + x^{12} + x^5 + 1) \end{aligned}$$

as well as

$$b(x) = x^{16}(ci(x))\text{mod}(x^{16} + x^{12} + x^5 + 1) \tag{9.2}$$

Here $ci(x)$ is the polynomial representation in the $GF(2)$ of the combined ARQ control and information field.

9.4.3.8 ESCAPE Service LU16

This service permits the development of a separate user plane in which use of the different terminals is only possible if an additional agreement exists outside the standard. The LU16 service can use each subordinate DLC service and any transmission category. However, a protected MAC service must be selected. The transparent LU1 service is available for unprotected services. No service may be implemented that is already covered by a standard service.

9.4.4 Network Layer

The network layer (NWL) (see Figure 9.16) controls the connections between the mobile part (*Portable Radio Termination*, PT) and the network (*Fixed Radio Termination*, FT); see Figure 9.9. The U-plane in DECT has no tasks in the network layer, and forwards all data unprocessed in the vertical direction. The C-plane carries out signalling, and is responsible for controlling data exchange. Five protocols with corresponding tasks and based on the *link control entity* (LCE) are provided for this purpose (see Figure 9.42).

MM *Mobility Management* is responsible for all tasks required for the mobility of mobile stations. It processes authentication messages and encryption data relating to the location area.

CC *Call Control* is the entity used for establishing, managing and releasing connections.

CISS *Call Independent Supplementary Service* is a supplementary service that sets up and releases call-independent connections for supplementary services.

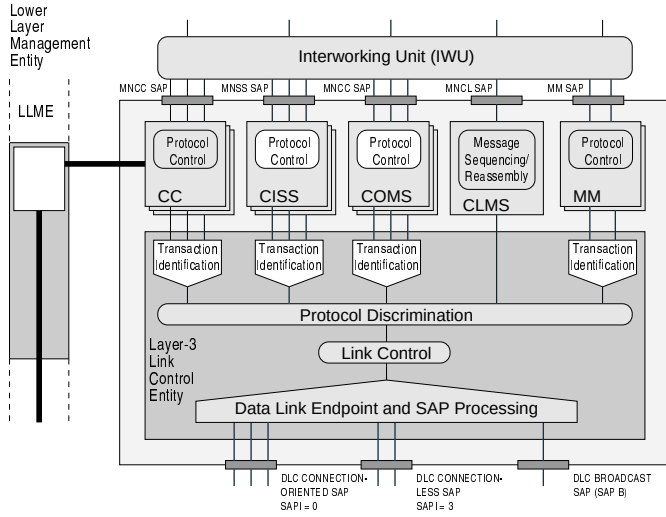


Figure 9.42: Services and protocols of the C-plane, layer 3 in DECT

COMS *Connection-Oriented Message Service* is a service that transmits packets between two end points. An end-to-end connection must first exist.

CLMS *Connectionless Message Service* is a connectionless message-switched data service.

LCE *Link Control Entity Messages* carries out logical connections over which all five higher-ranking entities exchange messages.

9.4.4.1 Mobility Management

This service manages the functions of user identity, authentication and location area updating. All the messages needed for mobility management (MM) are discussed below.

Identity Messages

- Temporary_Identity_Assign
- Temporary_Identity_Assign_ack
- Temporary_Identity_Assign_Reject
- Identity_Request
- Identity_Reply

These messages can be used to query the FT about specific identification data of the PT and to assign a *Temporary Portable User Identity* (`TPUI`). The identification is always initiated by the FT. A Temporary_Identity_Assign message is sent by the FT to the PT to agree on a TPUI with the PT. After successful execution, the PT acknowledges the assignment with

Temporary_Identity_Assign_ack. If an error occurs, the PT sends a Temporary_Identity_Assign_Reject. The FT uses Identity_Request to request an identity from the PT, to which the PT responds with an Identity_Reply.

Authentication Messages

- Authentication_Request
- Authentication_Reply
- Authentication_Reject

Authentication is carried out in two directions to guarantee to the network and to the PT that the PT or the network (as the case may be) has authorization. A new encryption parameter can be provided at the same time (see Sections 9.10, 9.16.1 and 3.3). The procedure is initiated with an Authentication_Request message. The calculated reply is returned with Authentication_Reply. Should the authentication not succeed or another error occur, the partner entity is informed through Authentication_Reject, and is able to take the appropriate action.

Location Messages

- Locate_Request
- Locate_Accept
- Locate_Reject
- Detach

These messages inform the network of the current location area of a PT. This enables an incoming call to be forwarded quickly to the respective PT. The mobile station sends a Locate_Request message to the FT in order to establish a connection or to request a handover to another base station. The FT responds to a successful connection or handover with Locate_Accept, and to an error with Locate_Reject. If a mobile station is switched off, it informs the network with a Detach message.

Access Rights Messages

- Access_Rights_Request
- Access_Rights_Terminate_Request
- Access_Rights_Accept
- Access_Rights_Terminate_Accept
- Access_Rights_Reject
- Access_Rights_Terminate_Reject

With these messages the *International Portable User Identity* (IPUI) and the *Portable Access Rights Key* (PARK) can be stored or cleared again in the PT. There are the four PARK classes A to D. An Access_Rights_Request is sent by the PT to the FT to start the procedure. The FT responds with Access_Rights_Accept and the corresponding parameters, or with Access_Rights_Reject if there has been an error and the requested parameters cannot be transmitted. The Access_Rights_Terminate messages are the reverse of the messages described above.

Key Allocation Messages

- Key_Allocate

This message is used to replace an authentication key (AC) with a user authentication key (UAC).

Parameter Retrieval Messages

- MM_Info_Suggest
- MM_Info_Request
- MM_Info_Accept
- MM_Info_Reject

These messages supply information, e.g., for external handovers, which is transmitted with the help of existing connections. Using MM_Info_Suggest, the FT can propose to a PT that it carry out a handover or location update. The PT uses MM_Info_Request to request information from the FT. The FT supplies this information sending an MM_Info_Accept message. If the information requested by the PT cannot be sent, the FT informs the PT accordingly with MM_Info_Reject.

Ciphering Messages

- Cipher_Suggest
- Cipher_Request
- Cipher_Reject

Ciphering messages are used to transmit the ciphering parameters and to initiate or terminate the ciphering. The PT can make this recommendation to the FT with Cipher_Suggest. This then takes place through a Cipher_Request from the FT to the PT. In the event that the PT is unable to carry out the ciphering, it responds accordingly with Cipher_Reject.

9.4.4.2 Call Control (CC)

Call control is an instance of an entity responsible for call handling. It is used to set up, maintain and release calls.

Call Establishment Messages

- CC_Setup
- CC_Setup_ack
- CC_Info
- CC_Call_Proc
- CC_Alerting
- CC_Notify
- CC_Connect
- CC_Connect_ack

CC_Setup commences the call setup of call control, and can be initiated by both sides (FT and PT). Some of the setup parameters are sent at the same time; the other data is transmitted with CC_Info. A CC_Setup_ack message is optional and is only sent in the direction of a mobile station. CC_Call_Proc indicates to the mobile station that the call is being processed in the network. A CC_Alerting message indicates to the network that the mobile station is

calling or to the mobile station that the user terminal being called is being paged. CC_Connect signals the connection to the partner layer of the U-plane and CC_Connect_ack acknowledges it. CC_Notify is a message that can be sent during an existing connection in order to convey a message (such as a Hold) to the called party.

Call Information Phase Messages

- CC_Info (see above)
- CC_Service_Change
- CC_Service_Accept
- CC_Service_Reject
- IWU_Info

A CC_Info message can be used to establish an end-to-end connection (more precisely from a PT to an IWU) for the exchange of information. With CC_Service_Change the communication parameters can be changed during the setup phase or during the active phase. A CC_Service_Accept message constitutes the corresponding acknowledgement and a CC_Service_Reject message a rejection of the parameter changes.

IWU_Information is used to transport `IWU Packet Elements` or `IWU to IWU Elements` when corresponding data is to be transmitted but cannot be attached to any other message.

Call-Related Supplementary Services

- Facility
- Hold
- Hold_ack
- Hold_Reject
- Retrieve
- Retrieve_ack
- Retrieve_Reject

Facility is used to request or acknowledge a supplementary service. The desired service is supplied within the messages with the appropriate parameters.

An MS can use Hold to transfer an existing connection to waiting status. The MSC acknowledges with Hold_ack. This waiting status can be rejected by the MSC with Hold_Reject. The Retrieve messages are the reverse of these commands. Retrieve is used by the MS to reactivate a call in waiting status. The MSC replies with Retrieve_ack or with Retrieve_Reject.

Call Release Messages

- CC_Release
- CC_Info
- CC_Release_com

CC_Release releases all U-plane and C-plane calls in the network layer. A CC_Release message is transferred to the link control entity (LCE), which then decides which procedure should be followed for releasing a call. CC_Release_com is the acknowledgement for the release of the call.

9.4.4.3 Call Independent Supplementary Services (CISS)

CISS Establishment Messages

- CISS_Register

CISS Information Phase Messages

- Facility

CISS Release Messages

- CISS_Release_com

Supplementary services consist of call-independent supplementary services (CISS) and call-related supplementary services (CRSS). CISS connections are call-independent. CISS_Register is used to set up new calls without end-to-end connections. CISS_Release_com indicates that the sender of a message is releasing the connection and also instructing the receiver to release the connection.

Facility is used to request or acknowledge a supplementary service. The desired service is provided within the message with the necessary parameters.

9.4.4.4 Connection-Oriented Message Service (COMS)

COMS Establishment Messages

- COMS_Setup
- COMS_Notify
- COMS_Connect

COMS Information Phase Messages

- COMS_Info
- COMS_ack

COMS Release Messages

- COMS_Release
- COMS_Release_com

COMS offers a point-to-point connection that can be used for transmitting packets. This connection can be established at any time using COMS_Setup. A successful connection is acknowledged with COMS_Connect. The connection phase uses a COMS_Info message to transmit the information of a COMS connection. A COMS_ack message acknowledges that one or more COMS_Info messages have been correctly received. The COMS connection is released through a COMS_Release command and acknowledged through COMS_Release_com.

9.4.4.5 Connectionless Message Services (CLMS)

Call Information Phase Messages

- CLMS_Variable
- CLMS_Fixed

CLMS can be used to transmit data when an end-to-end connection does not exist. The FT uses CLMS_Fixed and CLMS_Variable to transmit applications-related data to the PTs. In contrast to all the others, which have the S-format (see Section 9.4.4.8), a CLMS_Fixed message is in the B-format.

9.4.4.6 Link Control Entity Messages (LCE)

LCE Establishment Messages

- LCE_Request_Page
- LCE_Page_Response
- LCE_Page_Reject

The LCE entity in the physical layer is below the five entities (MM, CC, CISS, COMS, CLMS) above it, and is used to set up connections to the partner layer with LCE_Request_Page. The positive response to a connection setup request is LCE_Page_Response; the negative one is LCE_Page_Reject. Like CLMS_Fixed, LCE_Request_Page is a B-format message.

9.4.4.7 State Diagrams for Connection Control

Along with the operating states of the mobile and base station (see Figures 9.20 and 9.21), each protocol can be represented by a state automaton. All the possible states and transitions of a system are listed in a corresponding state diagram. New states are achieved whereby an existing state is transferred through transition into another state. Associations exist between the states and transitions in a way that some states cannot be directly transferred into any other state.

These states can be generated for each protocol, and are of particular interest here for call control because it too uses the protocols below it and establishes complete connections. It is possible for more than one connection to exist between a mobile station and the network, but in this context a simple connection is considered only. Figure 9.43 shows the states of a base station (FT) and Figure 9.44 the states of a mobile station (PT).

State diagram for a mobile station (PT) Table 9.6 presents a summary of the most important messages and their effect on the state diagram (see Figure 9.43).

The states are divided into four groups.

Central calling states (PT)

- `T-00 Zero` No call exists.

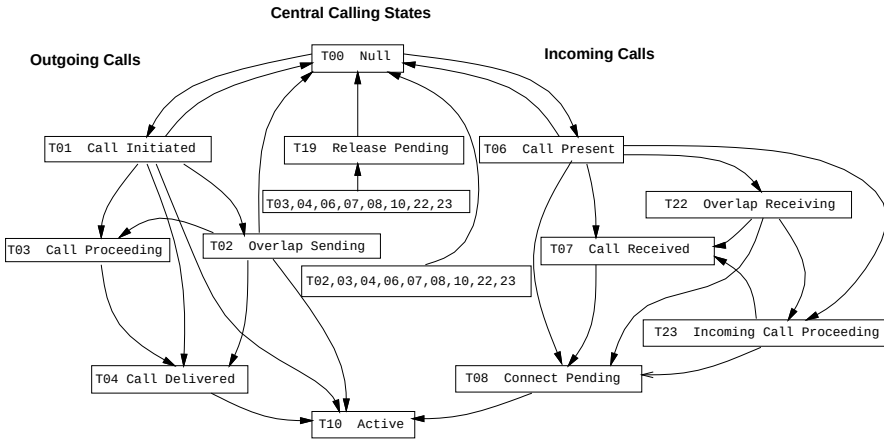


Figure 9.43: State diagram for a mobile station

Table 9.6: The transitions in the CC state diagram for mobile stations

From T-state	00	01	02	03	04	06	07	08	10	19	to
CC_Setup_sent	×										T01
CC_Setup_rcvd	×										T06
Setup_ack_rcvd		×									T02
Call_Proc._rcvd		×	×								T03
Alerting_rcvd		×	×	×							T04
Connect_rcvd		×	×	×	×						T10
Connect_ack_rcvd								×			T10
Setup_Accept						×					T07
Connect_sent						×	×				T08
Release_sent			×	×	×			×	×		T19
Release_rcvd				×	×	×	×	×	×		T00
Release_Compl._rcvd		×	×							×	T00
Release_Compl._sent						×					T00

- `T-19 Release Pending` The mobile station has sent a Release message to the base station but has not yet received a reply.
- `T-10 Active` A call (outgoing or incoming) exists and a connection has been established.

States for outgoing call (PT)

- `T-01 Call Initiated` The mobile station has sent a Setup message to the base station in order to initiate a call.

- T-02 Overlap Sending Another outgoing connection is being established via *overlap sending*.
- T-03 Call Proceeding The base station has acknowledged receipt of the Setup message.
- T-04 Call Delivered The mobile station receives an Alerting message informing it that the terminal it is calling is being paged.

States for incoming calls (PT)

- T-06 Call Present The mobile station has received a Setup message but has not yet responded to it.
- T-07 Call Received The mobile station indicates to the base station through Alerting that it is ringing.
- T-08 Connect Pending The user of the mobile station has taken the call, but the mobile station is waiting for an acknowledgement that the U-plane connection is being established.

Optional calling states (PT)

- T-22 Overlap Receiving Another incoming call is being set up in an overlapped state.
- T-23 Incoming Call Proceeding The mobile station acknowledges receipt of the setup information from the base station.

State diagram for a base station (FT) Table 9.7 presents a summary of the most important messages and their effect on the state diagram (see Figure 9.44).

As with mobile stations, the states are divided into four groups.

Central calling states (FT)

- F-00 Null No call exists. With a Setup message received from the PT, the FT moves into state F-06. If the FT itself sends a Setup message to the PT in order to set up a call, it then moves into the state F-01.
- F-19 Release Pending The base station has sent a Release message to the mobile station but has not yet received a response. A Release_Complete or the running out of a timer leads to state F-00.
- F-10 Active A call (outgoing or incoming) exists and a connection is established. A Release message transfers the FT to state F-19, and a Release_Complete_rcvd message received from there leads to F-00.

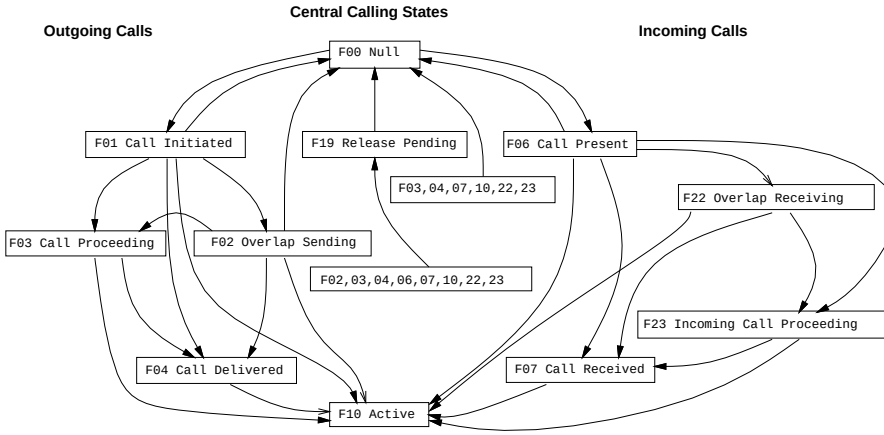


Figure 9.44: State diagram for a base station

Table 9.7: The transitions of a CC state diagram for base stations

From F-state	00	01	02	03	04	06	07	10	19	Final state
Setup_sent	×									F06
Setup_rcvd	×									F01
Setup_ack_sent		×								F02
Call_Proceeding_sent		×	×							F03
Alerting_sent		×	×	×						F04
Alerting_rcvd						×				F07
Connect_sent		×	×	×	×					F10
Connect_rcvd						×	×			F10
Release_sent						×	×	×		F19
Release_rcvd		×	×	×	×	×	×	×		F00
Release_Complete_rcvd						×			×	F00
Release_Complete_sent		×	×							F00

States of outgoing calls (FT)

- `F-01 Call-Initiated` The mobile station has sent a Setup message to the base station in order to establish a connection. The base station has not yet responded to it.
- `F-02 Overlap Sending` An outgoing call is being set up overlapping another.
- `F-03 Call Proceeding` The base station acknowledges that the Setup message from the mobile station has been received.

- `F-04 Call Delivered` The base station notifies the mobile station that the user terminal called is ringing.

States of incoming calls (FT)

- `F-06 Call Present` The base station has sent a Setup message, but the mobile station has not yet responded to it.
- `F-07 Call Received` The base station receives a message that the mobile station is ringing but the incoming call from the user is not yet being picked up.

Optional calling states (FT)

- `F-22 Overlap Receiving` An incoming call is set up overlapping with others.
- `F-23 Incoming Call Proceeding` The base station receives an acknowledgement that the setup information from the mobile station has been received.

9.4.4.8 Setup of Signalling Messages

A message is a block that consists of a number of bits and contains related data. Each eight bits are combined to form an octet organized from an MSB (*most significant bit*) to an LSB (*least significant bit*), and the bits are numbered from 8 (MSB) to 1 (LSB). Octets form the elements from which a message is created, with the structure the same for all messages and a differentiation made in DECT between a B-format and an S-format.

B-format messages These may only be sent by the LCE or the CLMS protocol.

- LCE_Request_Page
- CLMS_Fixed

All messages have a fixed length to make it easier for them to be processed at the lower levels (MAC level). Different formats are available for this (see Figure 9.45):

- Short format (3 octets)
- Long format (5 octets)
- Extended format (5, 10, 15, 20, 25 or 30 octets)

LCE_Request_Page uses either the short or the long format. In both cases they are structured in the same way.

Bits 1 to 3 in the first octet contain the header. Bit 4 is a flag for the `TPUI` address. In the long format the length of the `TPUI` address is indicated in bit

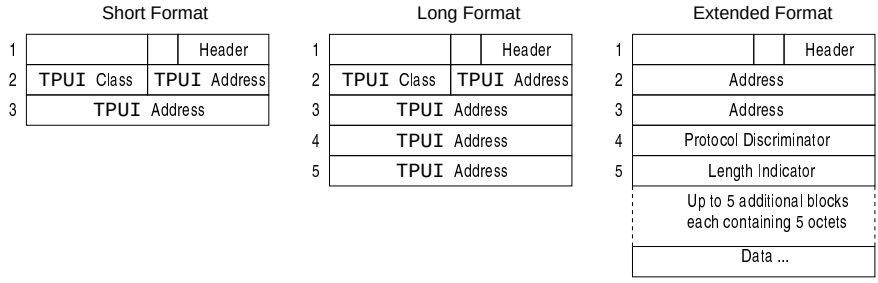


Figure 9.45: The structure of B-format messages

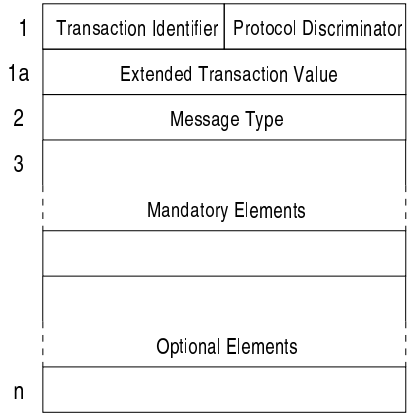


Figure 9.46: Structure of S-format messages

4. Bits 5 to 8 have no significance. Depending on the information in the first octet, the IPUI address begins in the second octet.

CLMS_Fixed uses either the long or the extended format. The extended format has a header consisting of five octets. In addition to the target address, the length of the data which belongs to this message is provided. This data is combined into blocks, each consisting of five octets.

S-format messages Each S-format message consists of the following parts (see Figure 9.46):

- Transaction identifier
- Protocol discriminator
- Message type
- Mandatory elements
- Optional elements

The first three elements are mandatory for each message, with the two elements protocol discriminator and transaction identifier contained in the first octet, each with four bits, and the message type in the second octet. The last

two elements are specifically allocated to individual message types, and can be of different lengths.

The protocol discriminator occupies bits 1 to 4, and describes the type of connection (MM, CC, CISS, COMS, CLMS, LCE) to which the message belongs.

The transaction identifier indicates in bit 8 the direction in which the message is being transmitted. A `0` signifies the direction from the calling to the called user; a `1` the other direction. A transmission value is assigned to bits 5 to 7. If it is the value `1 1 1`, the following octet is treated as an expanded transmission value (octet 1a).

The message type is identified in the subsequent octet: bits 1 to 7. This is where the code is for all MM, CC, CISS, COMS, CLMS and LCE messages; bit 8 is always `0`.

The octets 3 to n can contain mandatory and optional elements. Some elements have a firmly fixed length (*Fixed Length Information Elements*) and often a variable length in the second octet (*Variable Length Information Elements*).

Locate_Request is a typical example of an MM message (see Figure 9.47). A typical CC message is CC_Setup (see Figure 9.48).

9.5 Dynamic Channel Selection

The DECT system is designed to cope with high voice and data traffic loads. Since the load on the cells is usually distributed unevenly and the peak loads are extremely variable timewise and geographically, the process of *Dynamic Channel Selection* (DCS) is applied. This means that basically the entire frequency spectrum with all (120) channels is available in each cell, and a mobile station is able to select a suitable channel for itself.

In cellular mobile radio systems channel assignments are made in accordance with a fixed plan in which certain frequencies are allocated to the base stations on the basis of anticipated traffic volumes. This technique, which is referred to as *Fixed Channel Allocation* (FCA), requires very careful cell planning because it is difficult to make any changes once the installation has been implemented. The system makes it difficult to absorb short-term dynamic load changes in individual cells.

Because of the DCS technique, DECT systems do not require frequency planning, but instead a planning of the base station locations, just to guarantee a complete radio coverage of the service area. The system is capable of adjusting independently to changes in traffic load, and therefore can offer a lower blocking probability than FCA networks (see Section 9.13).

9.5.1 Blind Time Slots

The actual maximum number of channels in a DECT cell depends on the available number of transceivers at the base station. If it has only one transceiver

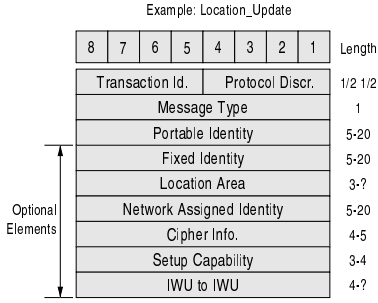


Figure 9.47: MM message structure based on the example of Locate_Update

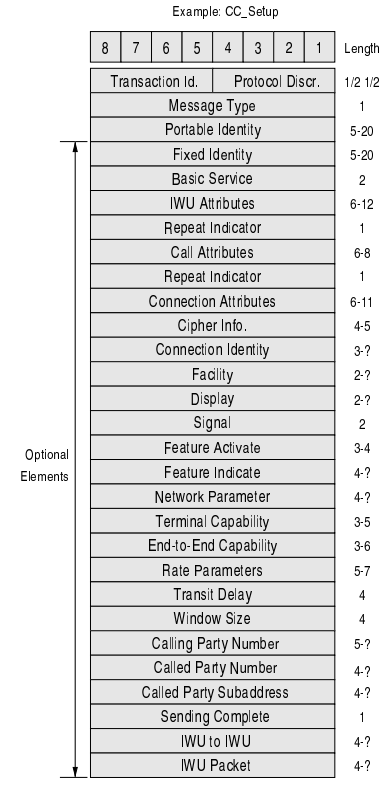


Figure 9.48: CC message structure based on the example CC_Setup

then it can only serve one mobile station within each time slot, because the transceiver is not able to work with two different frequencies at the same time. Therefore this means that only 12 duplex channels can be operated. If a terminal occupies a channel, the remaining channels in the same time slot (on the other frequencies) are flagged as *blind slots*.

Figure 9.49 illustrates the effects of a blind slot on a transceiver in a base station. Connections to frequencies 5 and 1 (black marking) exist in time slots 2 and 8. All other frequencies in these time slots are flagged as blind (shaded marking).

To inform the other mobile stations in the cell of the current blind slots, the base station periodically transmits this information as control information in the A-field.

A mobile station searches for the possible channels of a frequency in each frame in order to acquire information on the quality of the channels (see Section 9.5.1.1). If the terminal is occupying a channel for transmission, it can no longer monitor the channels of other frequencies in this time slot.

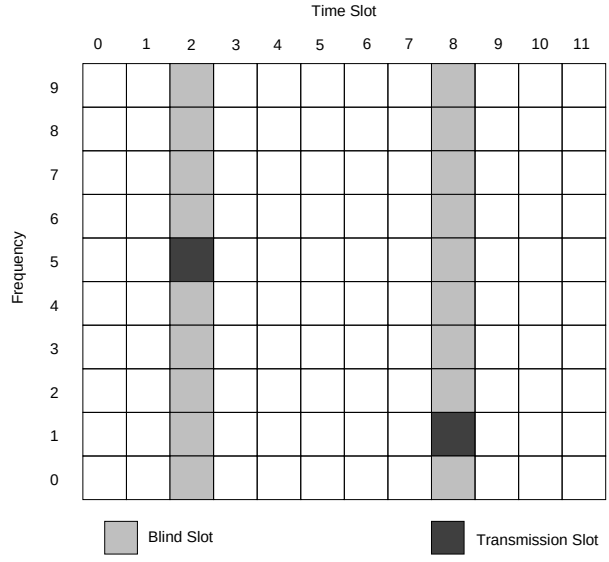


Figure 9.49: Blind slots in a base station with a transceiver operating two connections

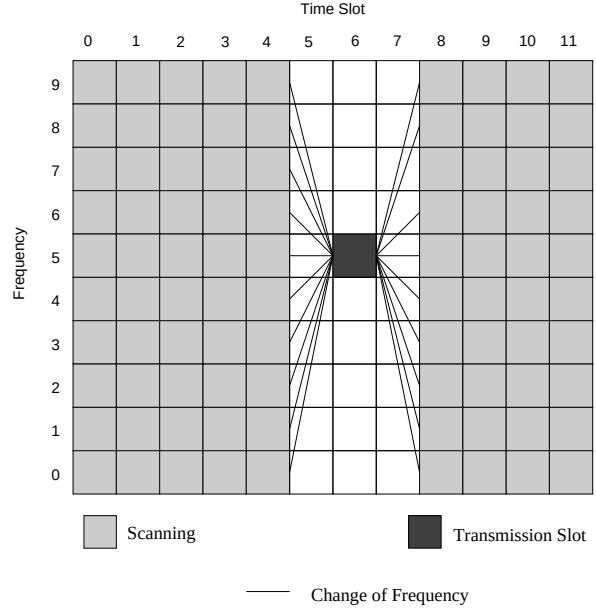


Figure 9.50: Channels not usable because of time required for PT to switch from communication to channel measurement

Because of the more cost-effective and simpler hardware used, it is usually not in a position to check the quality of the time slots that appear before and after the selected traffic channel. Likewise these blocked channels are no longer available for use by the mobile station (see Figure 9.50).

9.5.1.1 Channel Selection by Mobile Stations

The base station and the mobile station must select a physical channel before the initial transmission of information by a bearer. First the mobile station undertakes a measurement of the signal levels of the individual base stations and assigns itself to the one with the strongest level. The mobile station makes up to three attempts to set up a connection before it moves to the base station with the next-highest signal level.

The signal level (*Radio Signal Strength Indicator*, RSSI) determines the suitability of a channel. Relative to the reference level of 1 mW, it indicates the measured signalling power on the channel. Depending on the type of bearer requested, a number of different RSSI measurements are relevant to the decision.

- For a duplex bearer, the receiving channel in a terminal, in other words the downlink, is critical for the evaluation of a slot pair. As an example, using the RSSI measurement the mobile station establishes, e.g., in slot 2 with a frequency of f_x, the accepted interference power level for using the slot pair (2/14) on this frequency (see Figure 9.49).
- For a double simplex bearer, the relevant measurement is the one in that section of a slot pair that indicates a higher signal level.
- When it comes to simplex bearers, mobile and base stations have different measurement approaches. For a base station, the stronger of the two slots is relevant, similarly to the double simplex case. A mobile station evaluates the TDD half of a channel in which it is not transmitting its simplex bearer. An uplink bearer in slot 14 would be based on a positive measurement in slot 2 (see Figure 9.49).

The measured RSSI values are entered on a channel list (see Figure 9.51). The resolution of the measurement values should be 6 dB or better in order to achieve as fine a graduation as possible between qualitively good and bad channels. The lower limit of the list should be −93 dBm or lower. All channels with a lower signal level than the lowest limit of the list are classified as *quiet* and can always be used for bearer setup. The upper limit (*busy*) defines the area of the occupied channels, and remains variable. It can be adapted to the respective environment in which the DECT system is being operated and is normally at −33 dBm. Under no circumstances can the channels above this level be used in setup.

The channel list shown in Figure 9.51 has a granularity of 6 dB between the quiet and busy areas to allow channels to be entered appropriately in one

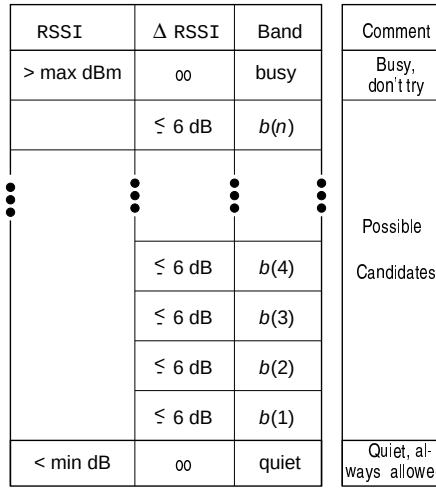


Figure 9.51: Channel list in MAC layer

of n rows. The following procedure is followed for the selection of a channel from one of the rows $b(1)$, $b(2)$ ... $b(n)$.

When the quietest channel is to be selected, the *quiet* area must be checked to ensure that a sufficient number of channels is available. If this is the case, then no channel is selected from the area b directly above it. On the other hand, if no *quiet* entry is available, a switch is made to the $b(1)$ area. If a channel is not available in this area either, then the $b(2)$ area is checked. In general, a transition is made from $b(x)$ to $b(x+1)$. If no channel can be found, the channel selection process is terminated and the base station is identified as *busy*.

Channel selection must be completed within two seconds (T210 = 2 s). A maximum of 10 attempts (N202 = 10) can be carried out during this time. If more than one bearer is being set up for a connection, up to five times as many more attempts are allowed.

Channels other than those with strong signals can also be entered as *busy* on an expanded channel list. One reason for this restriction could be that a transceiver unit is not able to support all frequencies. Blind slots can also be included in the busy area to prevent the selection of these channels.

9.5.1.2 Power Control

The standard provides the option of two levels of power control, with 250 mW as the maximum permitted power.

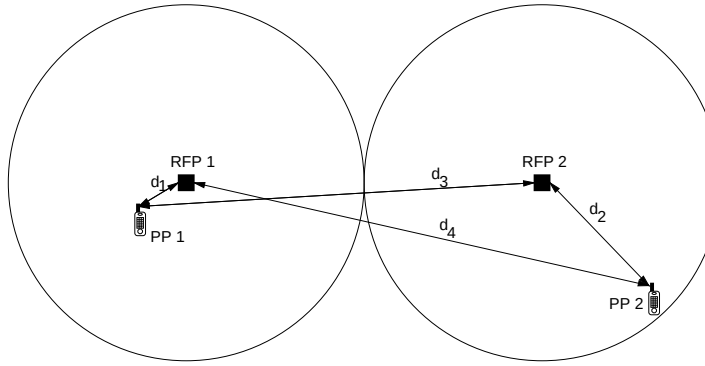


Figure 9.52: Channel displacement on the basis of the near/far effect

9.5.2 Channel Selection and the Near/Far Effect

It is easy to understand that with the DCS technique under high load in an area with many base stations the service areas of the base stations are breathing. The reason for this is that under high load the near/far effect dominates the decision as to which terminal will be able to open a new connection, with preference given to the terminals close to a base station. More distant terminals might even be pushed out of the system when a terminal near to a base station decides to set up a new connection; see also Section 4.3.3.

When a connection is first being established, the mobile station (*portable part*, PP) attaches itself to the base station with the highest signal level. A condition for the success of the channel setup is an adequate C/I in one of the 120 channels. A mobile station that is located in the close proximity of its base station will receive a higher signal level than one that is located further away. Consequently, the likelihood of a successful connection setup is greater with the mobile station that is closer to the base station than with the one further away. This is referred to as the near/far effect, which is clarified further in Figure 9.52 [24].

If a connection has been successfully established from the nearby mobile station (PP1) to its base station (*radio fixed part*, RFP1), it is possible that another mobile station (PP2) located further away and occupying the same channel to a different base station (RFP2) will be displaced from the channel due to high interference.

When two connections exist on one channel, the carrier-to-interference ratio C/I at the location of the nearest mobile station PP1 is

$$\left(\frac{C}{I}\right)_{PP_1} = \frac{{d_3}^{\gamma}}{{d_1}^{\gamma}}$$

with γ representing the attenuation coefficient of the radio propagation. At the location of mobile station PP2 that is located further away the following applies:

$$\left(\frac{C}{I}\right)_{PP_2} = \frac{{d_4}^{\gamma}}{{d_2}^{\gamma}}$$

If $d_1 \ll d_2$ and $d_3 \simeq d_4$ then it follows that:

$$\left(\frac{C}{I}\right)_{PP_1} > \left(\frac{C}{I}\right)_{PP_2} \tag{9.3}$$

In a DECT system a specific carrier-to-interference ratio C/I of, e.g., x dB, is required in order to set up and operate a connection on a channel. In the case considered it is possible that the nearer mobile station PP1 is measuring a carrier-to-interference ratio of more than x dB during its connection setup, and consequently will be successful in setting up a connection on this channel. This means that the carrier-to-interference ratio for the more remote mobile station (PP2) will will then fall below the required value of x dB, i.e., this mobile station will be displaced from its channel. As a result, it is even possible that the PP2 will be cut off from its connection if it is not able to find an alternative channel.

9.6 Speech Coding Using ADPCM

The 300–3400 Hz frequency range is usually used for the transmission of voice signals over telephone lines. Therefore the voice base frequency, which is between 120 and 160 Hz for men and between 220 and 330 Hz for women and children, is not transmitted. The hearing range for a person is between 16 Hz and 16 kHz. The highest perceptible frequency decreases from 20 kHz for a child to 10 kHz as one ages. The covered dynamic area is 130 dB. Despite the band limitations of a voice signal in the 300–3400 Hz frequency range, the level of syllable intelligibility is 91 %. Phrase intelligibility is even at 99 %.

In DECT systems *Adaptive Differential Pulse Code Modulation* (ADPCM) is employed in the digital signal coding of voice signals [12]. The assumption is that the analogue signal $x(t)$ has already been converted into the digital scanning sequence $\{x(t = k\,T)\}$. An analogue/digital conversion of the voice signal is required in order to produce this scanning sequence. First a band limitation of the signal to $f \leq f_g$ is carried out using an analogue filter.

The frequency f_g then becomes the cutoff frequency of the filter. This is followed by sampling with a frequency $f_A = 1/T \geq 2f_g$.

The objective of ADPCM is to reduce the number of binary decisions required in the quantization without reducing the voice quality. This can be done because of the correlation of the successive sampling values. There is a relatively small change in the signal $x(k)$ from sampling value to sampling

value. Successive values $x(k)$ are dependent on one another. With this dependency of the signal values being exploited, the prediction is that the point in time $k = k_0$ is predicted from n past sample values.

ADPCM was standardized as ITU-T Rec. G.721 in 1984. The coding is based on sampling with $f_A = 8$ kHz and a quantization of the residual signal with $w = 4$ bits into 2^w levels. This produces the data rate $f_A w = 8\ \text{kHz} \cdot 4 = 32$ kbit/s.

The voice quality is only negligibly worse than with logarithmically companded *Pulse Code Modulation* (PCM). Companding is the combination of a compressor and an expander — two characteristics used to produce an uneven quantization.

With PCM a distinction is made between an A-characteristic (used in Europe), which is approximated on the basis of 13 segments, and a μ-characteristic (used in the USA and in Japan). The compressor characteristics enable an irregular quantization to maintain a constant relative quantization error. This involves quantizing low signal values more finely than higher ones.

The quantizer adaptation is halted for stationary signals, thereby making the system also suitable for modem signals.

The achievable voice quality with 32 kbit/s ADPCM is comparable to the quality of 64 kbit/s-PCM (A-characteristic, μ-characteristic).

9.7 Handover

A handover takes place when a connection is switched to another base station because of poor quality, or a channel change within a cell is effected by the same base station. Because of the small cell sizes in microcellular systems it is assumed that roaming mobile stations will initiate a large number of handovers. Because a handover involves a great deal of signalling and the occupation of additional channels, there is an interest in minimizing the number of needed handovers.

The *Radio Resource Management* (RR) of a DECT system provides a decentral mobile-controlled handover (MCHO) algorithm that decides whether and when handover is necessary. *Seamless handover* (SH) in which the mobile station does not leave its current channel until a new one has already been set up is usually possible. In contrast to a *non-seamless handover*, the user usually does not notice that a channel or cell change has taken place.

There are two different physical forms of handover (see Figure 9.53):

Intracell Handover This occurs when the time and/or frequency channel within a cell changes. When this happens, the selected base station is retained.

Intercell Handover If the original base station is released when a channel change takes place and a new channel is set up by a new base station

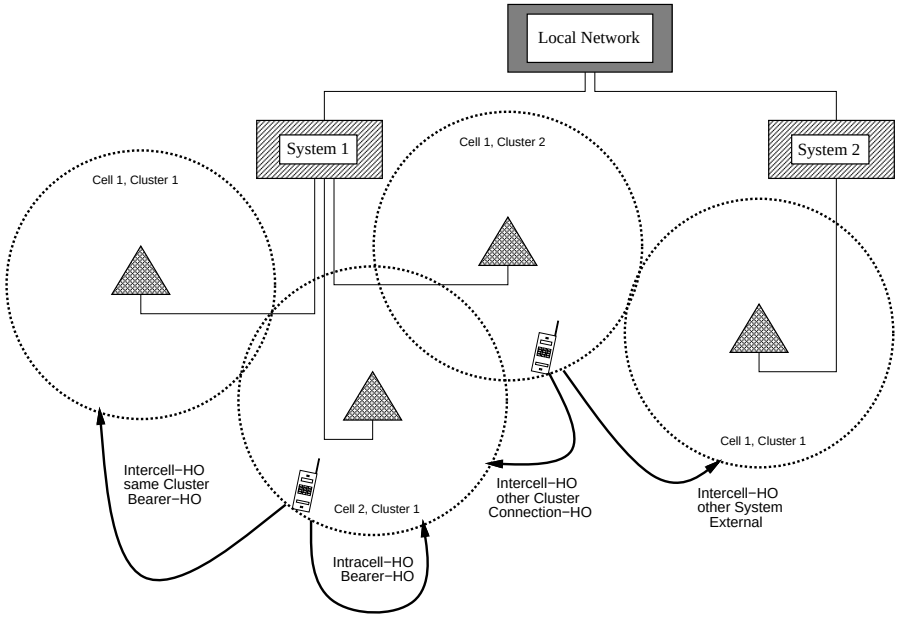


Figure 9.53: Physical and logical DECT handovers

then this is referred to as an intercell handover. When this happens, a new frequency and/or slot selection can be made in the new cell.

In addition to the differentiation between two types of handover because of physical characteristics, it is also possible to define two logical steps in a channel change:

Internal Handover If the handover takes place within a closed DECT system (*fixed part*, see Section 9.2.2), this is an internal channel change or handover. In the protocol, this handover technically occurs either in the MAC layer or in the DLC layer.

External Handover A changeover between two independent DECT systems (*Fixed Part*) is referred to as an external handover. This is a high-level handover that is executed in the mobility management of the network layer. A short-term loss of service (*non-seamless handover*) can occur with this handover.

9.7.1 Bearer Handover

A bearer handover is an internal handover, and requires fewer protocol efforts than any other handover process. It takes place within the MAC layer, and

only affects certain slots. Along with an intracell handover, an intercell handover can also take place at the MAC level, as long as the two cells involved are within one cluster (see Figure 9.53).

While the process is taking place, the current bearer is retained until the new one has been completely set up. There are briefly two bearers at the same time jointly supporting the same connection. Bearer handovers are initiated on the basis of quality tests carried out by the MAC layer of the mobile device, taking the following into account:

- A-field: 16-bit R-CRC
- B-field (unprotected): 4-bit X-CRC
- Synchronization impulse
- Information for connection identity
- B-field (protected): four 16-bit R-CRCs in the field parts
- Comparison of X-field and Z-field (*sliding collision*)
- Signal level (RSSI value)
- Frequency fluctuations
- Quality observations of the base station in the Q1 and Q2 bits of the A-field header

During the handover of a duplex bearer, the corresponding MBC must be able to recognize at least one new channel. It produces a TBC that sets up the new channel using bearer setup. Once the new bearer has been set up, the MBC of the base station decides which channel is to be maintained. The bearer no longer being used is then released.

Because the aim is to keep the number of channel changes to a minimum, a bearer handover cannot be reattempted until an interval of `T202` = 3 s has passed after a successful handover.

9.7.2 Connection Handover

A connection handover is an internal intercell handover between two base stations that belong to different clusters. It is executed in the DLC layer on the basis of its protocol. The service for the network layer is maintained during this process.

A connection handover occurs for one of the following reasons:

- Involuntary handover due to loss of service in the MAC layer
- Voluntary handover because of the poor quality of a connection

The LLME triggers a connection handover in which a new MAC connection is established. Similarly to a bearer handover, two MAC connections are operated during a voluntary handover process in order to guarantee a *seamless handover*. A seamless change of channel is not possible in an involuntary handover, because the old connection no longer exists at the start of the handover. The original type of connection (basic, advanced) is retained during the change in connection.

9.7.3 External Handover

The change from one DECT system to another is controlled by the mobility management of the network layer. This high-level handover from one network node to another should only be carried out in special cases, because of the amount of signalling required. A common management entity that coordinates the handover process must exist above the network layer for the execution of the change. During the transition, important processes such as encryption must be stopped and reactivated in the new system. There is no guarantee of a seamless handover, and it is possible that a connection will be broken off.

9.7.4 Handover Criteria

Loss of channel quality in a DECT system is usually attributable to one of two reasons. One is that a user's mobility can cause a reduction in the signal level of the base station when the terminal moves out of the supply area. Another reason is that the somewhat random reuse distance of a channel results in more or less heavy interference. A handover is initiated early in order to counter these two effects. Both situations are described in detail below.

9.7.4.1 RSSI Handover

There can be several reasons for the decrease in a base station's signal level in the location where a mobile station is operating. The main reason is the change of location caused by the mobility of the terminal. If a mobile station begins to roam too far away from its immediate base station or moves into radio shadowing, there is the chance of a sharp drop in signal level described by the *Radio Signal Strength Indicator* (RSSI). A sudden loss of *line-of-sight* (LOS) to the base station can cause a loss of 15–30 dB.

If a mobile station roams beyond the logical cell boundary into a neighbouring cell, the signal strength of its own base station decreases more and more, while that of the neighbouring station increases. A change in cell is desirable in this case in order to improve the quality of the connection. The measured RSSI values are transmitted over a time window in order to filter out the effects of short-term signal loss. An intercell handover should not be initiated until the signal level averaged for the neighbouring station is higher by a certain threshold value than that for the terminal's own base station

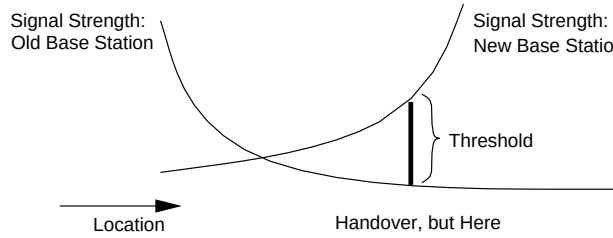


Figure 9.54: Hysteresis curve in an RSSI handover

(see Figure 9.54), in order to prevent frequent intercell handovers on the cell boundaries without a significant improvement to the receive level.

9.7.4.2 C/I Handover

If, in spite of a high signal level, the quality of a channel deteriorates owing to interference from other radio connections then a handover must be carried out within the same cell or to a more favourable cell. This kind of disturbance is caused by co-channel interferers which are occupying the same channel in other cells. The parameters used for characterizing this event are the *Carrier to Interference* (C/I) ratio and the *Bit-Error Ratio* (BER). Limit values that take into account the technology used and the requirements for connection quality are defined for the C/I ratio and/or bit-error ratio.

A cyclical code for forming check bits for error detection (CRC) is used for the measurement of connection quality in the DECT system. With the 32 kbit/s ADPCM voice coding used, the maximum bit-error ratio allowed is 0.001. For GFSK modulation with $BT = 0.5$ this corresponds to a minimum C/I ratio of 11 dB, which does not yet take into account field strength fluctuations (fading effects) originating in the radio channel [27]. If shading caused by obstacles in the path between sender and receiver are also considered in addition to the multipath propagation, the result is a required threshold value of 31 dB for the C/I ratio.

9.8 Protocol Stacks for Multicell Systems

A DECT fixed part in a multicell network consists of explicitly defined subgroups that can be transferred to the radio fixed parts. The result is that some of the protocol operations are only working in the RFP, thereby not burdening the whole system. However, many of the logical functions must more or less be centralized into one unit, thereby forming the gateway to a local network through the use of an internetworking unit. Since most multicell networks only possess one cluster, the layer 2 (MAC layer) of the DECT system can physically be divided into two parts; see Figure 9.23. The cell site functions (e.g., TBC) are allocated to each RFP, whereas the cluster control functions

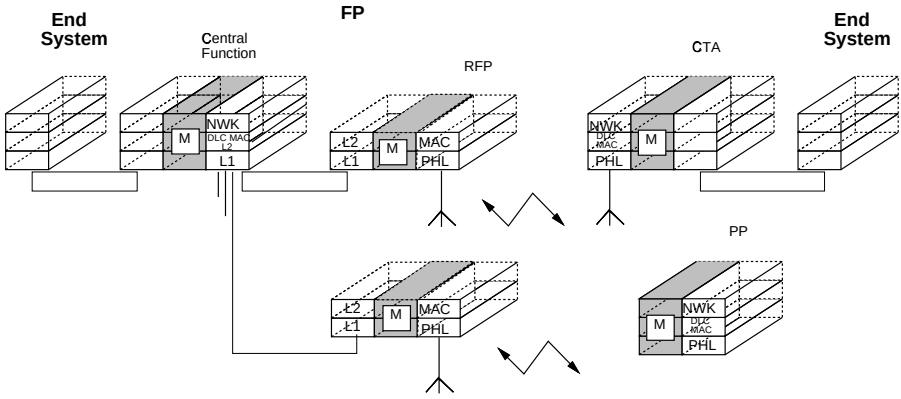


Figure 9.55: Protocol architecture for multicell systems (NWK= Network)

(e.g., MBC) remain in the central network element. The internal protocols L1 and L2, as indicated in Figure 9.55, can be specified and configured for their use by the operators themselves.

A DECT system requires a control unit, called the *DECT Central System* (DCS), which controls and links together all the base stations (FP). All the data necessary for authentication and encryption must also be filed in this unit. The DCS must also provide access to external networks through the interconnection of an *interworking unit* (IWU). The fixed network side is configured differently by each of the individual manufacturers of DECT systems.

9.9 The DECT Network Gateway Unit

In order to transport information between DECT and other systems, the DECT system uses an interworking unit at the interface to other networks (see Figure 9.42). The interworking unit works at two parallel planes. All signalling data and protocols such as *call control* and *mobility management* are converted in the signalling plane (*C-plane*). The user data for an existing connection is transmitted over the user data plane (*U-plane*). This data must be converted to conform with the requirements of other networks.

9.9.1 Signalling Data

DECT [16] provides service primitives for the interworking unit at layer 3 that relate to the protocols *mobility management* (MM) and *call control* (CC). These service primitives are used to convert the DECT protocols into the protocols of external networks. The services are made available to the higher layers of the interworking unit by DECT layer 3.

Table 9.8: CC service primitives of the interworking unit

CC service primitive	Request	Confirm	Indication	Response
Setup	×		×	
Setup_Acknowledge	×		×	
Reject	×		×	
Call_Proceeding	×			
Alert	×		×	
Connect	×	×	×	
Release	×	×	×	
Facility	×		×	
Info	×		×	
Modify	×	×	×	
Notify	×		×	

Table 9.9: MM service primitives of the interworking unit

MM service primitive	Request	Confirm	Indication	Response
Identity	×	×	×	×
Identity_Assign	×	×	×	×
Authenticate	×	×	×	×
Locate	×	×	×	×
Detach	×		×	
Access_Rights	×	×	×	×
Access_Terminate	×	×	×	×
Key_Allocate	×		×	
Info	×	×	×	×
Cipher	×	×	×	×

The service primitives listed in Tables 9.8 and 9.9 consist of elements for the actual transport of information. They correspond to the elements of protocol messages in the DECT standard [16], which are described in the next section.

9.9.2 User Data

User data can consist of coded ADPCM speech. The data must be converted if the connected external network requires the user data in another form. The DECT user data is provided in the interworking unit at a uniform PCM-interface point. The user data is provided in PCM-coded form at this reference point, and can be accepted by external networks.

9.10 Security in DECT

Mobile telecommunications networks must be protected against unauthorized access and misuse. Criteria relating to system security were therefore established for DECT [17]:

- User identification
- Unauthorized use of a mobile station
- Base station identification
- Unauthorized use of a base station
- Illegal eavesdropping of user or signalling data

The security measures described below relate to these criteria.

9.10.1 User Identification

At the beginning of a call a user is requested by the mobile station (PP) to input the *Personal Identity Number* (`PIN`). This number is checked locally by the mobile station. Each subscriber has an *International Portable User Identity* (`IPUI`), which is clearly defined in an authorized area (*Portable Access Rights Key*, `PARK`). Together with the `PARK`, the `IPUI` provides a unique identification for the mobile station [11]. Every `IPUI` can be used with several different `PARKs`, with the costs calculated over the network authorized by the `PARKs`. A mobile station also requires an `IPUI` and one or more `PARKs` in order to initiate a call.

9.10.2 Portable Access Rights Key (`PARK`)

There are four categories of `PARKs`, depending on the size of the DECT system:

`PARK Category A` is the access right to small single- or multicell systems in private households.

`PARK Category B` is the access right to more complex private branch exchanges and local area networks (LANs).

`PARK Category C` corresponds to the public version of categories A and B.

`PARK Category D` is designated for public use when a DECT system is connected directly to GSM.

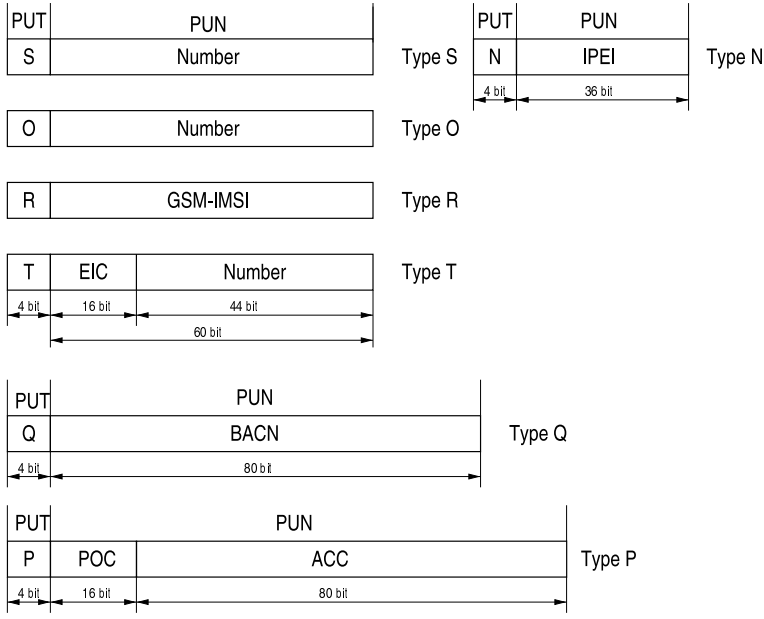


Figure 9.56: The structure of an `IPUI`

9.10.3 IPUI

An IPUI contains the *Portable User Type* (`PUT`) and the *Portable User Number* (`PUN`). A `PUT` is 4 bits long; the length of a `PUN` depends on the type of `IPUI`. For example, seven different types of `IPUI` for use in different applications are defined in DECT (see Figure 9.56).

`IPUI Type N` is the simplest type of `IPUI`. It consists of a `PUT` and a 36-bit long `PUN` (see Figure 9.56) and is used in conjunction with `PARK Class A` in private households.

`IPUI Type S` is provided for DECT connection to the ISDN because the 60-bit long `PUN` is coded like a PSTN or an ISDN number. It is used together with `PARK Category A`.

`IPUI Type O` is a local unique number in a private branch exchange (PABX or LAN) based on `PARK Category B`. The `PUN` can be specified by the operator of the branch exchange so that it is a locally unique number corresponding to the PSTN number. The `PUN` is 60 bits in length.

`IPUI Type T` is used for expanded branch exchanges. The 60-bit long `PUN` consists of a 16-bit long *Equipment Installer's Code* (`EIC`) and the additional number (44 bits). Large companies with a network of subsidiaries have the possibility of assigning each location with its own `EIC`. This

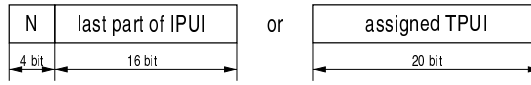


Figure 9.57: The structure of a TPUI

enables each mobile station within the subsidiaries to be used just as it would be in the home subsidiary.

`IPUI Type P` is available for public Telepoint use or radio local-loop applications. The 96-bit `PUN` consists of a *Public Operator Code* (`POC`, 16 bits) and a *Telepoint Account Number* (`ACC`, 80 bits), which are used in the calculation of call charges. The operator is coded in the `POC`. The uniqueness of the `IPUI` is only guaranteed through the `POC` and the `ACC`.

`IPUI Type Q` corresponds to Telepoint use, similarly to `Type P`, but is calculated using an 80-bit long BCD-coded *bank account number* (`BACN`).

`IPUI Type R` is provided for public access to GSM via a DECT mobile station. The identity is unique, and allows for DECT and GSM charges to be calculated together. The 60-bit long `PUN` corresponds to the GSM network BCD-coded `IMSI`, which can be up to 15 digits long [3]. This `Type R` is only available to operators of GSM networks and is used in conjunction with the `PARK Category D`.

9.10.4 TPUI

In local applications a temporary telephone number (*Temporary Portable User Identity*, `TPUI`) agreed between FP and PP is used for contacting a mobile station. This protects the `IPUI`, which in this case is not transmitted, from eavesdropping. The `TPUI` consists of 20 bits and is either the last part of an `IPUI` or an assigned `TPUI` (see Figure 9.57).

9.10.5 Authentication of a Mobile Station

Before a connection is set up, a mobile station is checked to ensure that the equipment is not recognized as stolen and that the user has authorized access to the network. This check is undertaken on the basis of an *Authentication Key* (`K`), which is stored in the mobile station (DAM) and in the base station (see Figures 9.58 and 9.82). Through the use of algorithm A11, a temporary key (*Session Authentication Key*, `KS`) is calculated from `K` and a value transmitted by the FP (`RS`). A random value (`RAND F`) selected by the FP is transmitted to the PP where on the basis of the KS and algorithm A12 an *encryption result* (`RES 1`) is calculated. This result is sent back as proof of identification of the `KS` to the FP, where the result is compared with that calculated in the FP. If there is any disparity between the two, the connection is immediately

terminated. This identification check protects a mobile station from using an unauthorized or manipulated base station.

The advantage of a temporary key is that there is no need for the PP to transmit its secret authentication key when it is visiting a different coverage area. K is calculated from an authentication code (AC), user authentication key (UAK), or a combination of UAK and *user personal identity*.

9.10.6 Authentication of a Base Station

The process is the same as that described above for a mobile station, except that the roles of the PP and FP are reversed. The PP calculates a *Reverse Authentication Key* (KS') from the value RS received from the FP and the authentication key K using algorithm A21. A random value (RAND P) sent from the PP to the FP is calculated to produce the result RES 2 using A22 and KS' and then transmitted back to the PP (see Figure 9.58).

9.10.7 Equivalent Authentication Between Mobile and Base Stations

There are three different methods for carrying out this authentication: the direct method (see above) and two indirect methods.

The indirect methods rest on the premise that ciphering of the data is not possible unless the ciphering parameters are known. With the first method the PP is authenticated as described above. Using the data transmitted, the PP calculates a *Ciphering Key* (CK) and uses it to encipher all data. If the FP does not have the correct authentication key K, it will calculate the ciphering key CK incorrectly, and therefore not understand the data sent by the PP — which will cause the connection to be terminated. The authentication is completed when the FP understands and is able to respond to the ciphered data.

With the second indirect method, a static ciphering key that is used by all PPs and FPs is agreed [17].

9.10.8 Ciphering of User and/or Signalling Data

When data is transmitted on the radio interface, it is sent in ciphered form to protect it from unauthorized eavesdropping. For this purpose a ciphering parameter (*Derived Cipher Key*, DCK) is calculated during the authentication. A *Static Cipher Key* (SCK) can also be used in the ciphering. A key stream that can encipher the data on the radio channel is calculated in a *Key Stream Generator* (KSG) (see Figure 9.58). The parameters and algorithms for security are listed in the top part of Figure 9.58.

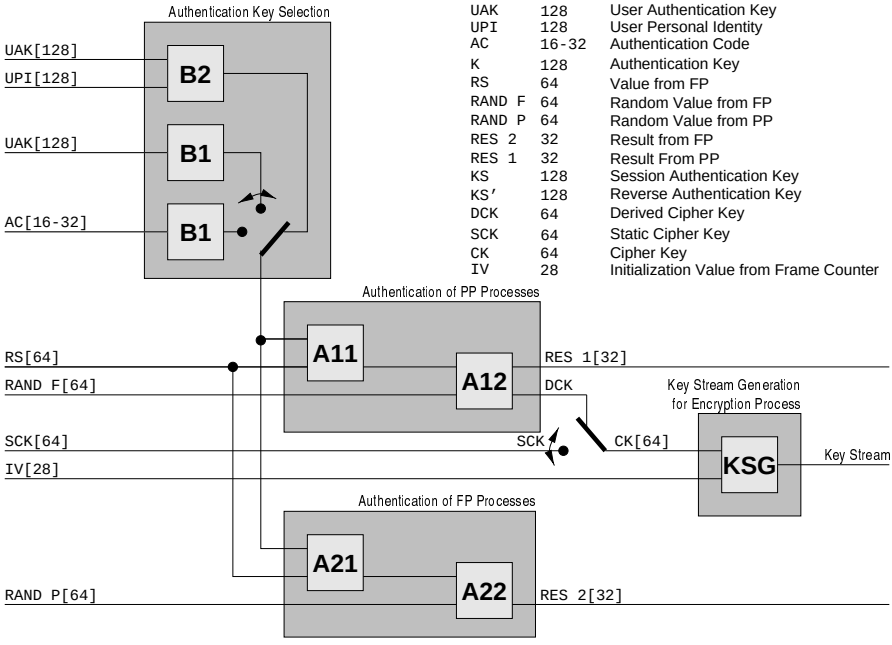


Figure 9.58: Authentication and ciphering

9.11 ISDN Services

In discussions on the suitability of the DECT system as a public access network *(Radio in the Local Loop, RLL)* a stipulation was made that ISDN services would have to be available to subscribers. Appropriate solutions were specified at ETSI/RES 3, and these included taking advantage of the possibility of using parallel DECT bearers for the same connection.

9.11.1 End System and Intermediate System

If a user wants to connect a DECT terminal to the ISDN, a conversion unit (*Interworking Unit*, IWU) must ensure that the protocols are converted into one another (see Figure 9.14).

The IWU converts the functions of one system into the functions of the other system. One application is a DECT terminal, which is connected to an ISDN basic connection and allows the use of certain ISDN services. This type of constellation is referred to as an *end system* [15].

Another application of the interworking unit is in linking ISDN terminals to an ISDN over a DECT radio interface, with the IWU on the network side undertaking the conversion ISDN↔DECT and the IWU on the user side the conversion DECT↔ISDN. This involves transmitting transparent ISDN

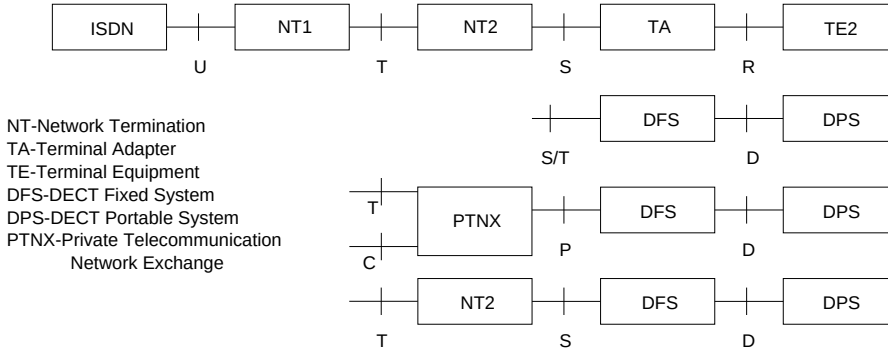


Figure 9.59: Reference configuration of an end-system (first line: ISDN reference model)

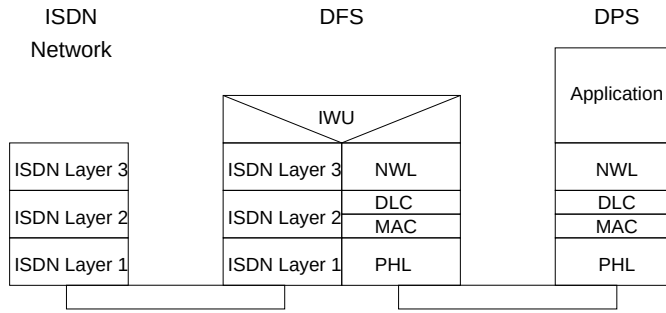


Figure 9.60: Control Plane of IWU in an end-system configuration (NWL = DECT network layer)

channels over radio. This constellation is referred to as an *intermediate system* [19].

9.11.1.1 Reference Configuration of an End System

A reference configuration describes a functional system constellation on the basis of the reference points of the DECT or ISDN system (see Figure 9.59). The significance of these interfaces is explained in [23].

An end-system configuration exists when the *DECT fixed system* (DFS) and the *DECT portable system* (DPS) together assume the role of an ISDN terminal.

Figure 9.60 presents the IWU protocol stack for the C-plane. It shows that the lower three layers of the OSI reference model are converted, whereas the right side corresponds to the protocol stack in Figure 9.16.

Figure 9.61 illustrates the protocol stack for the IWU for the U-plane in the voice service. It shows that layer 1 of ISDN is converted to the physical

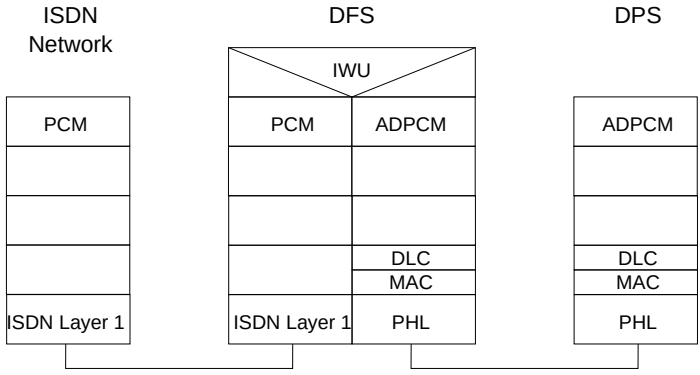


Figure 9.61: User plane of IWU in an end-system configuration using the voice service as an example

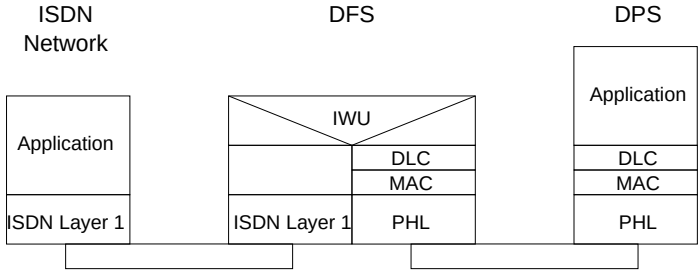


Figure 9.62: User plane of IWU in an end-system configuration using the 64 kbit/s data service as an example

DECT layer and that the DLC and MAC layers are also contained in the IWU on the DECT side. Furthermore, it is clear that the ISDN speech coding PCM must be converted to the DECT coding ADPCM above layer 3.

Figure 9.62 presents the reference model specified for the 64 kbit/s data service. It shows that on the DECT side of the IWU the data link layer (DLC) carries out the backup of the 64 kbit/s-data service described in Section 9.4.3.7.

9.11.1.2 Intermediate System Configuration

For comparison purposes, the reference model for the C-plane of an intermediate system is illustrated in Figure 9.63. In contrast to the C-plane in an end system (see Figure 9.60), it can be seen that the layer-3 messages in ISDN are forwarded from the intermediate system to the partner entity without being mapped to the DECT functions in the IWU.

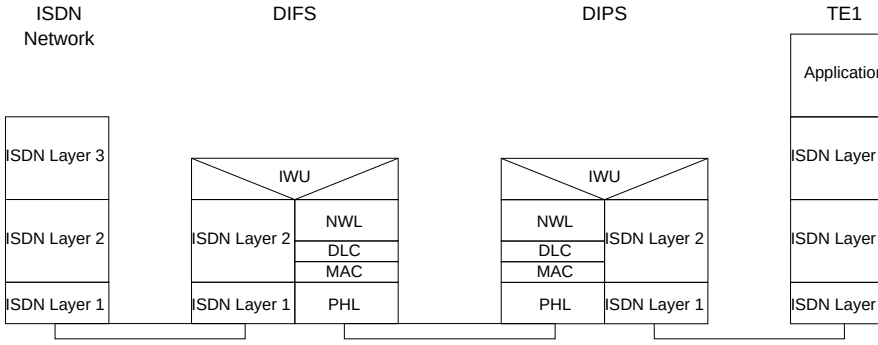


Figure 9.63: Control plane of IWU in an intermediate system configuration

To provide a 64 kbit/s data service, a DECT station must be able to transmit 640 bits — which corresponds to 80 bytes — of user data per sending cycle, because a new DECT frame is transmitted every 10 ms.

However, because a slot only contains 480 bits, additional channel capacity must be created through a bundling of slots. Two consecutive slots are used for this purpose. This solution is referred to as a double slot and is used in the standard [15] for the 64 kbit/s data service.

For comparison purposes, Figure 9.35 shows the distribution of a double slot (packet `P80`) in the different fields. The distribution of the D-field that is also shown is explained in detail in Section 9.4.3.7.

9.12 DECT Relays

A DECT relay is a wireless DECT base station (WBS) that provides coverage over radio to partial areas in DECT networks without requiring a direct connection of the WBS to a PSTN/ISDN. A draft standard Wireless Base Station (WBS) has been available since March 1997 [18].

The use of DECT relays opens the door to a variety of different types of applications. However, these can all be divided into two categories:

- Applications outside buildings (*outdoors*)
- Applications inside buildings (*indoors*)

These two areas of applications are analyzed in detail and examples of special fields of application for the relay concept are provided in the following sections.

9.12.1 Outdoor Applications

As an extension to DECT base stations (FT), operators use DECT relays in wireless local networks (RLL) to replace the previously wirelined *last mile*

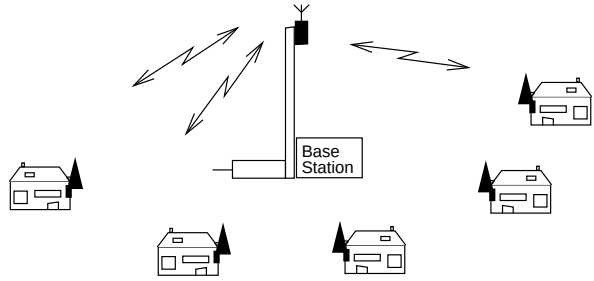


Figure 9.64: Radio local loop (RLL) system

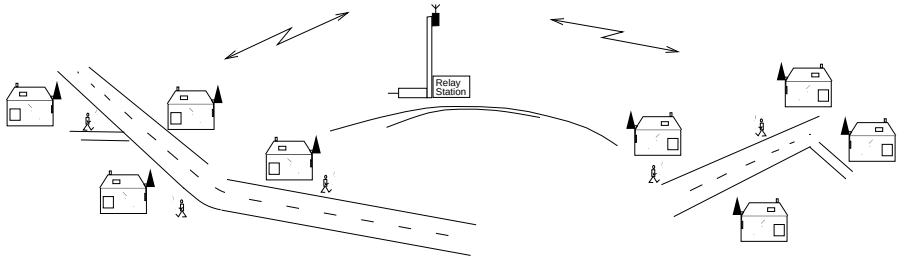


Figure 9.65: Radio local loop (RLL) with public access

from the local exchange, concentrators or cabling distribution panels up to the home location of the customer with a DECT radio path (see Figure 9.64). The system therefore leaves the indoor area for which it had originally been designed, and takes over network functions outdoors. The elimination of customers' wireline connections first provides the flexibility needed to establish a coverage area quickly, and secondly offers a cost-effective option through the use of relays. The savings in cabling costs in the local network in which each household is connected to the local exchange through a two-wire line makes the wireless local network especially attractive.

Whereas these measures in the distribution network of a network operator can equate to lower charges for the customer, the next step attempts to make these operator routes available for public access. This means that a mobile DECT handset would not only be used for private use to communicate over DECT systems installed in homes but could also access the public network with its relays and base stations. Consequently, the Telepoint concept, which has already been tested but not proven successful in Germany, would be reintroduced in a different version over the RLL system.

This would create a local or regional wide-area network that could guarantee comprehensive secure mobile coverage to subscribers. An example of this network is illustrated in Figure 9.65. Mobile subscribers can be reached anywhere in the whole scheme, and are able to access the fixed network not

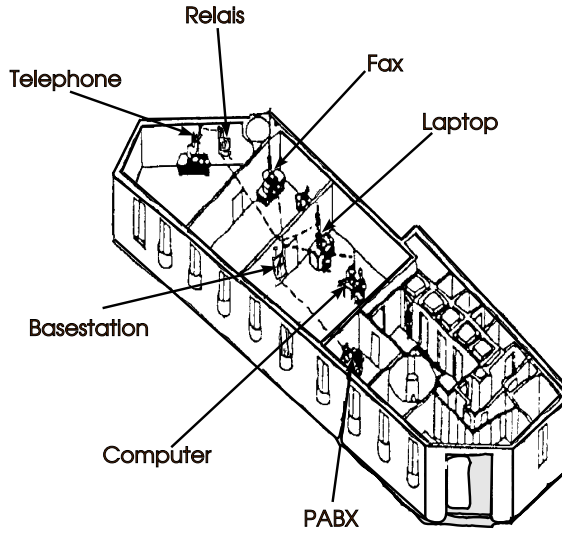


Figure 9.66: Indoor relay system

only directly through the base station but also over the stationary DECT end systems if they are implemented as relays. The *Generic Access Profile* (GAP) [7] approved by ETSI in 1996 and the *Public Access Profile* (PAP) [13] lay down a manufacturer-independent access interface for DECT systems ensuring that public access to other systems is also possible from every PP.

9.12.2 Indoor Applications

Similar ground constellations as outdoors are also found indoors. Relays can be added to existing DECT systems in order to improve the coverage to certain areas. Also, when consideration is being given to installing a new DECT branch exchange, the savings on cabling a house can be an important factor in favour of DECT. Relay stations particularly lend themselves for use by companies with scattered sites and assembly plants, as well as places like construction sites where a work location is used for a limited period of time (see Figure 9.66). Except for access to power supply sources, relays eliminate the need for any cabled connections, and sometimes even solar power supplies can be used.

In contrast to RLL, in indoor use relays are considered integral components of the system. As with RLL systems with public access, the capacity of an indoor network is determined by the mutual interference of the cells among themselves. Similarly to outdoor situations, indoor relay systems, with a

given number of send/receive facilities, are not restricted by the number of channels available (*trunk-limited*). However, the multiple occupancy of the radio medium through the use of relays causes a deterioration in the C/I values on the cell boundaries. This creates a capacity loss that cannot be offset even by an increase in the number of transceivers.

The strength of interference in both indoor and outdoor applications is similarly dependent on the spacing selected for the relays and base stations, the transmitter power and the antennas used. It is therefore possible to make a direct comparison of the results from indoor and outdoor studies if the dimensions of the different scenarios are taken into account.

9.12.3 Relay Concept

Relay systems offer the following advantages:

Savings on cabling This is especially attractive to users of DECT indoor systems. Existing cabling is often not sufficient for linking an adequate number of base stations to the fixed network.

Wider radio coverage The radio coverage for a situation can be improved considerably through the use of relays. Areas that later turn out to have insufficient supply (shadowing) can be integrated into an overall system through the use of relays.

DECT implementation Relays allow users to connect to base stations and relay stations using conventional DECT-compatible terminals. Subsequent expansions to a system using relays therefore do not affect the mobile stations.

Retention of frequency and transmitter power Relays in a DECT implementation use the frequency channels authorized for the DECT system. The 250 mW maximum transmitter power specified also applies to the relay systems, and therefore no additional approval is required for the operation of relays.

Mobility of relays for changing locations of use Because relays are not restricted to a particular network, the operator is able to adjust the system manually to cope with long-term load fluctuations in selected areas.

These advantages carry different weight in every network installation. Depending on the particular application, the indoor and outdoor areas can overlap to the extent that a system is referred to as a combined one.

In principle, all concepts can be divided into two categories.

9.12.3.1 Stationary Relay Stations

As the name implies, these are fixed relay stations (see Figure 9.67). Although they can be moved manually, they usually remain fixed to one location.

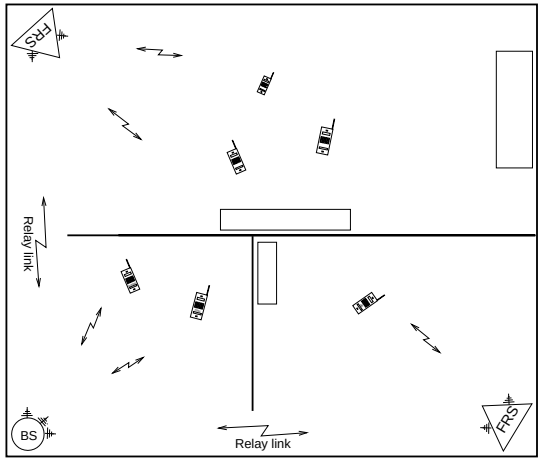


Figure 9.67: Indoor application with fixed relay stations (FRS) and RFP (= BS)

The DECT radio interface to the user is maintained, and no modifications are required to the hardware and software in the user's terminal. Therefore the user is not even aware that a *Fixed Relay Station* (FRS) is being used. The mechanism for forwarding a call from a relay to a base station (BS) that is linked to a network is a matter for the supplier of the system, and can be controlled and changed within the network. A high transmission quality will be achieved if an optimal location is selected and complex antenna systems for the link FRS↔BS are used.

9.12.3.2 Mobile Relay Stations

Mobile relay stations (MRS) are not in fixed positions like the FRS, but are movable. One possible implementation of a mobile relay could be the use of appropriately modified handsets (see Figure 9.68). Because each mobile terminal is able to offer relay operations along with the traffic channels it uses itself, this type of operation provides all the mobiles in a system with a considerably improved quality of service compared with fixed relay stations, albeit clearly placing higher demands on the terminals, particularly on battery use.

The quality of a connection depends on the mobility of the relays and the terminal, accompanied by all the associated risks, including the possibility of a connection being terminated. The issues that must be taken into account are the amount of routing required and the time delays that can occur because of the lack of control over the growing number of *hops* with mobile relay systems. A comparable concept for multihop communication is found with ETSI/HIPERLAN/1 (see Section 13.3).

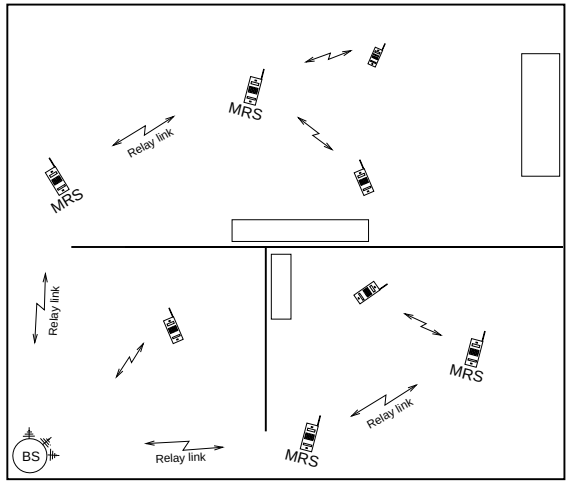


Figure 9.68: Indoor application with mobile relay stations (MRS)

The following discussion focusses exclusively on fixed relay concepts. An important advantage of the FRS variation is that the location can be determined precisely in advance and that there is a finite definable number of occurring sequential radio hops of connections.

9.12.4 Setting up a Relay Station

A DECT system that uses relays places a multiple burden on the radio medium. The mobile stations that are directly linked to the system in the other RFPs must share the available channels with the relays in the immediate vicinity.

Relay systems are characterized as follows:

A relay acts the same as an RFP for each mobile station It offers the PP the same services and is therefore completely interchangeable with an RFP.

Unrestricted configurability A relay can be adapted to prevailing local conditions. In addition to a specification of the frequencies and the time slots allowed, independent *Dynamic Channel Selection* (DCS) from the base stations for access to the fixed network is also possible.

Coverage to undersupplied areas Optimal coverage is possible because of the flexibility allowed in specifying the location of the relays. It is recommended that the corresponding signal strengths selected should be higher than average to help eliminate interference to channels between the relays and the base stations.

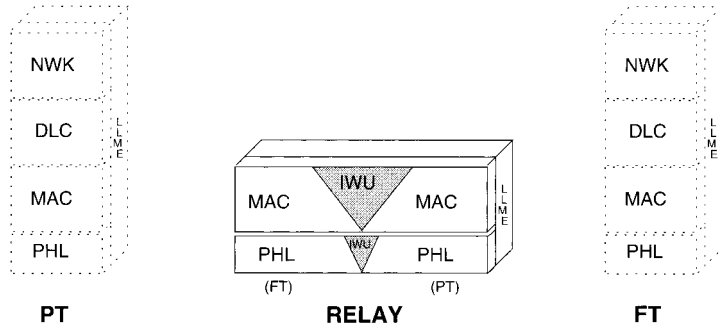


Figure 9.69: Reference model for relays

Interference due to relays Because they change incoming radio channels to outgoing ones, relay stations cause a higher level of interference and a reduction in the capacity of a total system compared with wireline fixed stations (RFP). With unimpeded free-space propagation in outdoor situations, the quality of service in neighbouring cells can drop considerably. This loss can be reduced through the installation of antenna systems with directional beams in relays and RFPs.

9.12.4.1 Basic Relay Structure

A DECT relay station (*wireless relay station*, WRS) contains the key functions of the *radio fixed part* (RFP) and of the *portable part* (PP). But there is no provision for things like network gateways, user interfaces and speech coding of the PP and the RFP. To a base station the relay appears as a mobile station with up to 12 traffic channels. A mobile handset perceives the relay as an RFP. The relay functions required are responsible for conversion of the PP $\leftrightarrow$ relay connection to the allocated relay $\leftrightarrow$ RFP connection (see Figure 9.69), where a relay is provided between the portable and the fixed radio termination (PT and FT).

If the relay operates as a pure repeater station then the time slots within the physical layer are interlinked through the relay functions in the physical layer. This transparent service allows data to be transmitted within the same half frame. If the relaying is carried out in the MAC layer, each connection in the relay is assigned its own separate MAC layer entity. This allows it to evaluate independently the quality of the channels and to execute a bearer setup or a handover using dynamic channel selection.

Each slot in a relay (FRS) can be used as either a send or a receive slot (see Figure 9.70). The allocation of slot areas in a relay is presented in Table 9.10.

This splitting-up of the transmitting and receiving areas makes it possible to retain the standard uplink and downlink separation in the RFP and in the mobile station. Consequently, no modifications are required to the terminals.

Table 9.10: Slot occupancy in relay systems

	Uplink Relay – RFP	Downlink Mob. term. – Relay
Relay slots 0–11	Receive (RX)	Transmit (TX)
Relay slots 12–23	Transmit (TX)	Receive (RX)

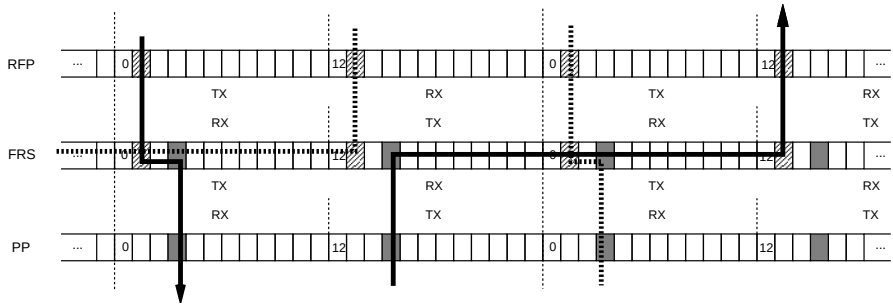


Figure 9.70: Example of slot occupancy with use of a relay station

This type of WRS is called a *cordless radio fixed part* (CRFP) [6]. A *repeater part* (REP) is a technically more complicated solution.

Figure 9.70 shows that a connection exists between the RFP and the relay (FRS) in slot pair 1/13. The corresponding downlink path between the relay and the mobile terminal (PP) is occupying slot pair 3/15. This means that an information exchange (black line in Figure 9.70) between the base station to the mobile station and back no longer only requires a half frame, but is taking up one-and-a-half frame lengths. When several relay stages are used, this time delay can cause unacceptable transmission delays (and resultant echo interference) for a connection.

Switching from transmitting to receiving mode between two slots requires high-speed transceiver hardware in the relay. If this kind of hopping is not possible, it is necessary for some of the slots to be set as *blind* so that the transceiver unit can be switched from sending to receiving and vice versa (see Figure 9.49). However, this reduces the slot capacity in the relay.

9.12.4.2 Connection Setup and Handover

When a mobile user is being connected to a network-coupled base transceiver over a relay system, several connections must be set up in sequence. As usual, the terminal first sends a request to what it knows to be the strongest base station, which in this case is considered as a relay (see Figure 9.29). If the bearer request has been received correctly, the base station responds with a standard WAIT command, to indicate that the request for a connection has been accepted. The next step is for the relay to establish a connection

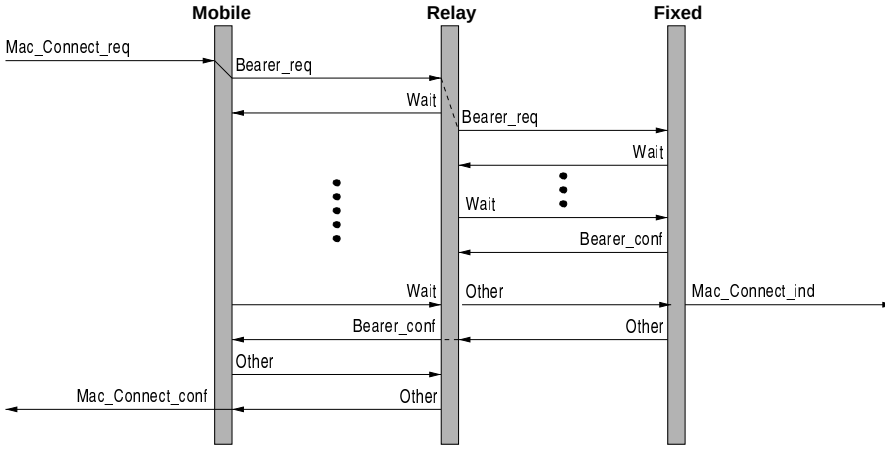


Figure 9.71: Connection setup using a relay station

to the base station coupled to the network. The connection cycle for the mobile station can continue as soon as this connection has been implemented successfully (see Figure 9.71).

If a setup error occurs between the relay station and the RFP, another attempt is made to set up a connection between the two, with no reinitiation of the WAIT connection to the mobile station required. However, if the connection timer or the counter in one of the systems (the mobile unit, the relay or the RFP) runs out, the connection setup will be considered to have failed and all the existing hops will be terminated.

A change of channel during an actual connection (*handover*) can occur for the traffic channel between the relay and the PP as well as between the RFP and the relay. If an intracell handover is required for a mobile station because of co-channel interference then a new connection is only established between the PP and the relay. The corresponding relay $\leftrightarrow$ RFP link is not affected by this process.

If a terminal initiates an intercell handover — in other words a change to a different base station — then not only the PP $\leftrightarrow$ relay connection but also the corresponding relay $\leftrightarrow$ RFP connection will be terminated once a completely new transmission path has been established over a qualitatively more favourable base station. The relay is normally responsible for initiating handovers on the uplink path (relay $\leftrightarrow$ RFP). In the same way as a PP, it measures the appropriate quality parameters and decides when an intracell or an intercell handover should be executed. This change of channel is not noticed by a downlink connection.

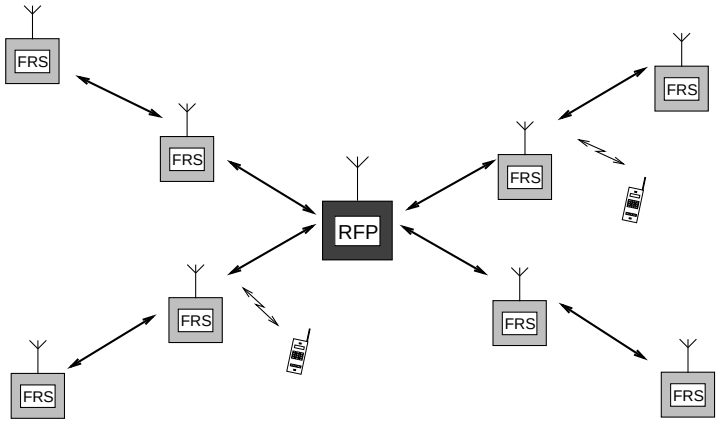


Figure 9.72: Star-shaped multirelay structure

9.12.4.3 Multirelay Structures

Until now the discussion has centred on the installation of relays between mobile terminals and wireline base transceiver stations. Figure 9.72 illustrates the simple layout of a multirelay structure. Here a number of relays at different levels are arranged around a fixed base station that has responsibility not only for its own internal cellular traffic but also for the traffic between the different relay levels. This star-shaped structure incorporates the following advantages and disadvantages:

- The star structure allows a particularly easy hierarchical classification of the relay stations. Depending on their level in the hierarchy, the FRSs have special features such as transceiver equipment, antenna configurations and routing characteristics.
- The classification according to level makes it easier for mobile stations to seek out the base station with the best performance characteristics. The mobile station should keep the routing in the relay network to a minimum.
- A potential disadvantage of the star-shaped structure is the heavy network use of the RFP. Because the base station supports all the traffic in the relay arrangement, it requires a large number of transceivers. Considering the relatively high level of interference in the central area, this is something that should be avoided.
- Network security is limited because there is only one gateway over the base station to the fixed network for the entire system. The system can collapse if parts of this important station become inoperative.

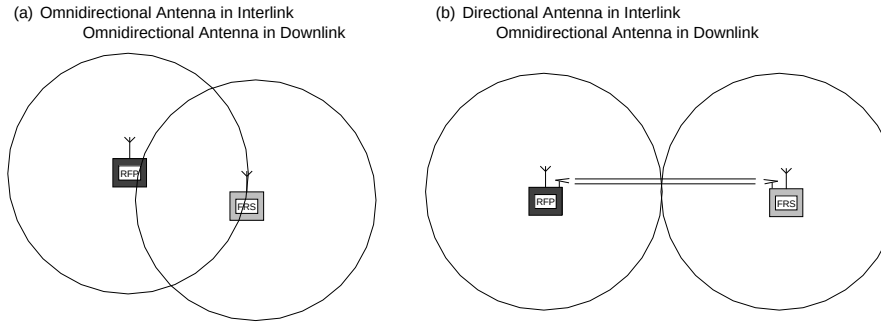


Figure 9.73: Range of a relay system with omnidirectional and directional antennas in the interlink of RFP and FRS

Other more complex physical arrangements or logical structures such as ring-shaped and bus-type layouts and hexagonal arrangements are possible in addition to the star-shaped one.

A system's range can be increased through the use of relays on the edge of an existing network. However, since the relay must be located within the transmitting area of a higher-level base station, the system only expands by half the diameter of the relay cell. Figure 9.73 (a) shows that a large part of the relay field overlaps with that of the base station when omnidirectional antennas are used. The actual increase in the system's range is smaller than half the cell radius. As can be seen in Figure 9.73 (b), the range of radio coverage can be improved considerably through the use of directional antennas for the link between the RFP and the relay, and omnidirectional antennas for supplying the respective PPs. In this scenario the receiving radius of the relay station is utilized almost completely to provide coverage to one of its own cells. The directional beam can be realized in the DECT frequency band.

The use of sequential relay systems increases the signal propagation time. Whether these factors have any effect on the functioning of a system or whether there should be an upper limit on how many relay systems in sequence are used can be determined based on the following considerations:

1. A delayed transmission of information can particularly cause timers of the signalling protocols to expire if acknowledgements for transmitted messages take a too long time to arrive. This necessitates limiting the number of relay levels.

2. The handover algorithms for multilevel relay systems are very complex. Although this does not affect intracell handovers, it makes the intercell handover process clearly more complex. Figure 9.74 illustrates the handover process between two relay systems. It shows that all higher-level connections must be reestablished and operated in parallel for a short time before the original connection can be terminated.

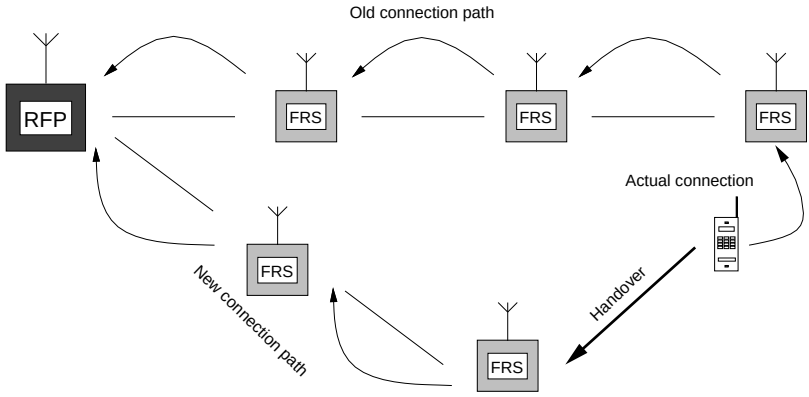


Figure 9.74: Handover process for relay systems

3. Synchronization of the individual relays is a problem in many relay systems. Whereas an impulse for synchronization over the appropriate hardware port normally guarantees the same cycle for all neighbouring base stations (RFP) connected to the same local network, this is not possible with relay systems coupled over radio. The only solution is for the system to synchronize itself following its own timing using the signal received from the higher level. This can lead to a shift in the synchronization, *sliding collisions* — in other words to a blocking of individual slots by non-synchronous stations.

4. Relays increase the channel load of a system. Interference in the network increases considerably with each relay level. Not even directional antennas can always reduce this interference. Relays are therefore especially suitable for improving coverage in situations involving a small or medium amount of traffic.

The relay in Figure 9.70 only has one transceiver unit; two slot pairs are required for each connection: one pair as the downlink from the mobile terminal to the relay station, and one pair as the uplink from the relay to the next RFP. This reduces the amount of traffic that the entire system is able to support. The relay can only accept 6 instead of 12 independent bearers per RFP. The installation of a second receiver in the relay helps to solve the problem. The uplink side (RFP $\leftrightarrow$ relay) and the downlink side (relay $\leftrightarrow$ mobile) are then physically separated on two transmit/receive facilities.

9.12.5 Parameters for the Performance Evaluation of DECT Systems

A specification of appropriate performance characteristics is required before the quality of a concept can be described.

System load of a configuration When an RFP is replaced by a relay or a radio local loop application is installed, there is an increase in interference in the total system according to the number of relay systems and fixed base stations, and the capacity falls. An analysis of the individual average traffic loads per base station or relay produces information on performance requirements, equipment and suitable locations for RFPs and relay stations.

Change to signal propagation times The delay times in a system are increased with the addition of each relay level. It is necessary to analyse various time parameters, e.g., for setting up a connection.

Number of blocked setup attempts and terminated handovers These two parameters can be used to define the *grade of service* (GoS) [14] of a system, which is an important measure of a network's quality of service:

$$\text{GoS} = \frac{\text{number of blocked calls} + 10 \cdot \text{number of interrupted calls}}{\text{total number of calls}} \qquad (9.4)$$

Therefore a lower grade of service means that the system has a high quality of service. As the GoS value increases, so too does the share of aborted connection setup attempts and interrupted connections.

Handover behaviour With an uneven distribution of system load within a network, it can be expected that the handover activity in the area of the relay locations will increase. It is there that interference caused by the double burden of uplink and downlink traffic is the greatest. The aim should be to strive for a relatively even distribution of the locations used for changing channels.

Radio Signal Strength Indicator (RSSI) and Carrier-to-Interference (C/I) ratio at terminal The actual quality of the supply to the user can be determined by the C/I ratio calculated by the mobile station on the basis of the RSSI level of the base stations at the PP location.

9.13 Traffic Performance of DECT systems

Most of the following also applies to the *Personal Handyphone System* (PHS).

There is a familiar model from traffic theory that can be used to calculate the traffic performance of DECT systems. It is derived from a large number of independent traffic sources (PPs) and a known number of identical radio channels accessible by each station. According to Appendix A.2.2, an $M/M/n-s$ model can be used for modelling, and allows the calculation of loss probability p_l (see Equation A.13). At $s = 0$ waiting positions in the control buffer the duration of a connection can be generally distributed, and is only calculated on the basis of its mean value. The requests for connections by the

PPs are modelled on the basis of a Poisson arrival process at the rate λ (calls per time unit): the product of the average duration of a call and the arrival rate is called offered traffic ρ and describes the average channel utilization rate minus the lost traffic rate (see Equation A.4).

From the probability p_l in Figure A.7, it can be inferred that with n available channels a connection will not be possible, because the channels of the base station concerned are temporarily occupied. The loss probability depends on the average utilization rate of the channels ρ of the base station. For example, with $n = 20$ channels the loss probability $p_l = 0.01$ would occur if each channel were used to 62 % capacity.

Simulation studies confirm that in some situations it is possible to make a reliable prediction of DECT traffic capacity (i.e., allowable percentage of utilization) with a given grade of service (see Equation 9.4) using the Erlang model shown in Appendix A.2.2. The requirement is that the arrangement should consist either of geographically isolated base stations (without neighbouring cells) or of stationary user terminals (as is the case with RLL systems) that can only be assigned to a specific base station. With a general base station layout, the correct assumption in terms of appropriate cluster size in the scenario considered is the one that determines the average number of available channels per cell.

9.13.1 Equipment and Interference-Limited Capacity

If the channels switched by the base station are not sufficient for dealing with a particular traffic load, in other words all 12 channels of the only transceiver of an RFP are occupied, and if the interference conditions would allow use of a 13th channel, then this is referred to as an equipment-limited capacity restriction of the base station. This capacity restriction can occur with geographically isolated or poorly equipped base stations in the cellular layout.

If the base stations are well equipped with transceivers then capacity is restricted by co-channel interference. Interference-limited capacity occurs when the hardware of the base stations would permit additional channels to be operated but other channels are not able to achieve the C/I value required for connection setup and therefore are not usable.

Owing to dynamic channel selection in interference-limited DECT systems, an average reuse distance between co-channel cells is created that depends on the path loss or morpho structure (see Section 2.3) and is very difficult to determine through measurement. It is known from simulation tests [24] of large-scale urban scenarios that average reuse distances are set at $p_l = 0.01$, which corresponds to a cluster size of about five.

9.13.2 Estimating the Capacity of DECT Systems

When occupied, each of the total of 120 available channel pairs produces an interference field that does not allow it to be reused locally until adequate

Table 9.11: Traffic capacity per RFP depending on cluster size with 10 transceivers/RFP

Clusters	Channels/ RFP	$\rho(p_l = 1\,\%)$	Erl./ RFP	$\rho\,(p_l = 0.5\,\%)$	Erl./ RFP
4	30	0.68	20.4	0.63	18.9
5[a]	24	0.64	15.36	0.60	14.4
6[a]	20	0.62	12.4	0.56	11.2
7	17	0.58	9.86	0.53	9.01
9	13	0.525	6.8	0.46	6.0
12	10	0.455	4.55	0.39	3.9
15	8	0.39	3.1	0.32	2.5

[a]Cluster sizes 5 and 6 are to be seen as mean values, which are not possible in cellular systems but can occur in DECT systems.

propagation attenuation has taken place. The path-loss models to be used depend on each individual situation, with the result that cluster sizes of 7 and below are achievable if the RFPs are not installed within the range of visibility. (In cellular systems a cluster describes the number of neighbouring cells in which each frequency channel may only occur once. These clusters are differentiated from the group of RFPs/FRSs that are connected to the fixed network at the same access point and are also referred to as clusters [22].)

For simplification purposes, p_l=0.01 is equated with a grade of service (GoS) of GOS=1% in the following. The aim would be to seek a value of GOS = 0.5 % in order to catch up with the grade of service of wireline local networks.

If one assumes that there is only enough radio coverage by RFPs over an area that the supply areas of neighbouring cells barely overlap (or not at all), and that each RFP is equipped with 10 transceivers (thereby guaranteeing that each frequency/time channel on each RFP is available), then, depending on the cluster size assumed or the resulting average number of available channels per RFP, the following capacity utilization ρ is produced per channel with the given value p_l, which can be derived from Figure A.7 (see Table 9.11). The traffic capacity (Erl./RFP), i.e., the number of connections allowed at the same time, can be established from ρ and the number of channels n per RFP, depending on the cluster.

Instead of being equipped with 10 transceivers, RFPs are clearly adequately equipped with, for example, only one or two transceivers per RFP if the supply zones of the neighbouring RFPs overlap sufficiently so that a PP is able to reach approximately 17 to 24 time channels (of its own and all the neighbouring RFPs together), as is the case for the 7-cell to 5-cell clusters in Table 9.11).

Table 9.12: Traffic capacity per RFP with one transceiver per RFP

Channels/RFP	ρ/N ($p_l = 1\%$)	Erl./RFP	ρ/N ($p_l = 0.5\,\%$)	Erl./RFP
12	0.5	6.0	0.44	5.3

If the radio coverage by RFPs is assumed so that the supply areas of the neighbouring cells barely overlap (or not at all), as is the case with fixed radio local loop systems (RLL), and each RFP is equipped with only one transceiver, then the results are those shown in Table 9.12, which can easily be understood from Figure A.7.

According to Table 9.12, the traffic capacity per RFP represents a lower capacity limit (with a given loss probability).

If the radio coverage areas of the RFPs equipped with one transceiver overlap then, depending on the situation, there is a corresponding increase in the number of possible channels per PP, and the traffic capacity per RFP achieves the values shown in Table 9.11. Overlapping helps to achieve a load distribution between sites with little or high traffic, and is therefore sensible; it also produces savings in the hardware needed for transceivers.

The cluster size appropriate for a given scenario depends heavily on the path loss (model): with morpho structures with little shadowing, 7-cell clusters are likely for modelling local channel reuse; with strong shadowing caused by buildings, channel reuse would be expected to be smaller, corresponding to a cluster size down to five.

If the RFPs/FRSs are very close together (e.g., <50 m because of the high amount of traffic), the probability of line-of-sight links between RFPs increases dramatically, and the cluster size needed increases to values of 20 or higher (with a correspondingly lower number of available channels/RFP and consequently less traffic capacity per RFP). Under these conditions, the DECT system is pushed to the capacity limits due to interference.

From experience, the traffic capacity per RFP for the physical arrangement of RFPs in multilevel buildings is approximately 20 – 30 % less than indicated in Table 9.11 owing to the occurrence of three-dimensional interference.

Reference [24] presents calculation methods that have been validated through simulation experiments for interference-limited DECT systems; these methods use the C/I ratio and the path-loss factor as parameters, and examine the uplink and the downlink separately. A complete theory covering the calculation of capacity of systems with dynamic channel allocation does not yet exist.

9.14 Traffic Capacity of DECT RLL Systems with Several Competing Operators

There is a discussion in Europe about awarding licenses to several competing DECT-RLL operators in the same coverage area in the DECT 1880–1900 MHz frequency band. In addition to the cases mentioned above, consideration must here be given to how much the different RLL operators would influence each other.

From a global traffic-theoretical standpoint, the explanations given in Section 9.13.2 continue to apply. Independently of the RFP/FRS density of the operators, the same overall capacity of the DECT systems, as calculated in Tables 9.11 and 9.12, would be available. This capacity has to be shared if there is more than one operator in the same coverage area.

It was pointed out in Section 9.13.2 that

- the transmitter power used per RFP/FRS and subsequent overlapping of the neighbouring cells of an operator, and
- the number of transceivers per base station

have a profound influence on the number of channels (with trunking gain) available to a PP. The achievable quality of service of each operator can be affected through the same measures. Simulation results show that when there are competing operators, each operator is forced to equip his RFPs with more transceivers than would be necessary for a single operator to take full advantage of the system capacity and to achieve the same quality of service [24]. Having a large number of transceivers also allows an operator to gain an advantage (on demand or continuously) at a cost to the others. The list that follows is certainly incomplete.

9.14.1 Using a Higher Density of Base Stations

In high traffic situations that (owing to the equipment installed and base station density) exceed the locally available system capacity of DECT, the operator who benefits at a cost to the other competitors is the one who installs his RFP/FRS stations closely together and hence close to the PPs of his subscribers (see Section 9.5.2). A PP that is situated close to its operator's base station is generally more apt to succeed in making a connection than a PP located further away. Depending on its signal strength, through causing interference it can even displace other PPs that are already connected but are operating relatively far away from their RFP/FRS (and therefore are probably served by competitors with smaller station density) from their channels, with the consequence that it will not be possible for them to find an alternative channel, and their connections will be interrupted.

The operator with higher base station density can have a noticeable affect on the quality of service of his competitors locally. This observation corre-

sponds to the internationally accepted view that a frequency etiquette in the form of *spectrum sharing rules* is necessary when different radio systems are being licensed in the same radio spectrum in order to ensure that the spectrum capacity is divided evenly among the interfering systems.

9.14.2 Use of More than One Transceiver per Base Station

In Section 9.13.2 it was shown that the use of several transceivers per RFP/FRS increases the number of channels available for connection setup or handover of a PP, and consequently has a favourable effect on the loss probability p_l.

For operators with low station density the installation of a sufficient number of transceivers per RFP/FRS compensates for the disadvantages created by the near/far effect if the capacity limits of the system have not yet been reached locally.

9.14.3 Channel Reservation

The effect of RFPs transmitting, irrespective of station density, at maximum transmit power in order to achieve a greater overlapping of their coverage areas is the same as it is when RFPs are equipped with many transceivers per RFP.

According to the DECT standard, PPs generally only attempt to set up connections or handovers on those channels for which the required carrier-to-interference ratio (e.g., $C/I = 11$ dB plus fading reserve) has been measured. The framework of the standard allows for technical measures to give the impression that the channels are being occupied (with the aim of reservation) to prevent the competitor's PPs from accessing them.

The (load-dependent/dynamic) use of channels (e.g., as beacon channels) with the aim of reserving these channels can distort the competition between operators.

According to a wise old saying that can be applied to the use of radio systems with multiple access, "anyone who is too considerate or careful will never find a channel". Accordingly, an operator could specify that a lower carrier-to-interference ratio than the usual 20 dB is adequate for call setup for his system when the other operator's PP is currently attempting to use a channel (in order to displace the competitor's PP from the channel through deliberate interference). The competitor could then grab the freed-up channel for his own use. The DECT system is based on a system of cooperation between PPs: the law of the jungle rules with all its consequences when the competitors' PPs are not cooperative.

9.14.4 Problems Anticipated through Mutual Interaction

When operating in the same band, competing RLL/PCS operators should be required to ensure that their PPs and base stations are cooperative and fair

with one another. It is difficult to define this concept, and almost impossible to delineate between what is fair and what is unfair. Moreover, the behaviour of a DECT system is controlled dynamically through parameters over the beacon channel; in other words, it cannot be monitored externally.

On this basis, it can be assumed that competition in the operation of public DECT systems will produce problems because of the coexistence of the systems at the same location. This also applies to competition in the operation of private and public systems — however, only at the location where they overlap.

It is possible to prevent the interference anticipated from competing public systems if only one public network operator is allowed per region or the operators (at least at the moment) use different frequency bands.

It is possible that the competitors in the DECT band in the future will be general licensed private and office systems and operators licensed to run public services. If this is the case, the following could apply:

1. Private systems and office systems will experience noticeable interference from RLL systems at so-called *hot spots*. Private operators have a right to confidentiality protection because public operators were licensed in *their* bands later.

2. In some cases public operators will adversely affect the services of private operators more than they should (see the comments in Section 9.14.5).

3. The quality of service of public systems will be adversely affected by private systems in so-called *hot spots*.

4. Competing operators of public systems in the same frequency band will take technical steps in an attempt to gain advantages from the standpoint of capacity and quality of service.

   It is known from a field study carried out in Sweden that users typically judged the quality of service of RLL systems to be lower than that of the copper wire lines of the local network [2]. Therefore each operator has a high motivation (within cost limitations) to compete with as high a quality of service as possible.

5. Because of the constraints of licensed systems for public use and the generally approved systems for private use being operated in the same frequency band, the DECT system is pushed to its technical limits more quickly than would be the case if these conditions did not exist. Therefore there is a danger with DECT systems that they appear to be less efficient than would be the case if normal conditions applied, that they are losing their reputation as systems with international applicability, and that they are suffering in the competition with other concepts.

9.14.5 Separating Competing Operators in the Spectrum

There are different solutions for alleviating or avoiding the problems mentioned:

1. Restricting public operators to a part of the DECT band. This then provides private operators with the confidentiality protection expected. The frequencies reserved for their use should be measured so that they are sufficient for normal use (e.g., three carriers). This proposal reduces the trunking gain of public operators.

2. Licensing of only one public DECT network operator on the condition that he must allow access to at least two competitive service providers per regional area to his network. The result of this proposal could be that some regions only have one service provider. The DECT licensee is in a similar situation as, e.g., an incumbent local loop operator awarding use of its local network to a third party.

3. Licensing in the DECT band, but partially separated from competitive public operators because of a restriction to a subset of available carrier frequencies, with some shared frequencies (e.g., allocation of 7 instead of 10 carriers per operator, with 3 of them for the exclusive use of each operator). The exclusively allocated frequencies are shared with the private systems.

   This proposal noticeably reduces trunking gain, because only 84 channels instead of 120 are available per operator, and, according to the calculation in Table 9.11, these are to be divided among clusters. However, there is a part of the band (36 channels) that each operator does not have to share with the competitors.

   With proposal 1 each of two operators would only have exclusive use of three carrier frequencies and shared use of one (48 channels together). This would produce a sharp drop in trunking gain.

4. Licensing in the DECT band but partial separation from competing public operators through the availability of additional DECT extension bands (adjacent to the DECT band).

   According to this proposal:

   - The extension band will be shared by both operators (e.g., $x = 10$ MHz).
   - Each operator will have exclusive use of part of the band (e.g., $x = 5$ MHz).

5. Licensing public operators in a DECT extension band (FPLMTS band) with a bandwidth of approximately 20 – 30 MHz and with two options:

- Both operators are competitors in the same band.
- Each operator is allocated exclusive use of part of the band.

This solution corresponds to the licensing arrangements expected for third-generation cellular systems (FPLMTS). The advantages and disadvantages of both solutions have already been covered.

6. Licensing in the DECT band in accordance with one of the models (1, 2, 3) listed above for the introductory phase (in which traffic loads are still light), with the option of providing an extension band later for shared or (partially) separate use by competitive public operators, as the need arises.

Of all the options mentioned, under current conditions the last solution appears to be practical and the most viable.

9.15 DECT Abbreviations

ADPCM	Adaptive Differential Pulse Code Modulation
ARQ	Automatic Repeat Request
BER	Bit-Error Ratio
BMC	Broadcast Message Control
BSC	Base Station Controller
CBC	Connectionless Bearer Control
CeBIT	Centrum Büro Information Telekommunikation
CCITT	Comité Consultatif International des Télégraphes et Téléphones
CEPT	Conference of European Posts and Telecommunications Administration
CIR	Carrier Interferer Ratio
CMC	Connectionless Message Control
C-Plane	Control Plane
CRC	Cyclic Redundancy Check
CT	Cordless Telephone
DBC	Dummy Bearer Control
DCA	Dynamic Channel Allocation
DCS	Dynamic Channel Selection
DECT	Digital Enhanced Cordless Telecommunications
DLC	Data Link Control
DSP	Digital Signal Processor
ETSI	European Telecommunications Standards Institute
FCA	Fixed Channel Allocation
FDMA	Frequency-Division Multiple Access
FEC	Forward Error Correction

FRS	Fixed Relay Station
FSK	Frequency Shift Keying
FT	Fixed Radio Termination
GMSK	Gaussian Minimum Shift Keying
GP	Guard Period
GSM	Global System for Mobile Communications
HO	Handover
IRC	Idle Receiver Control
ISDN	Integrated Services Digital Network
ISO	International Standards Organization
IWU	Interworking Unit
LLME	Lower-Layer Management Entity
LMS	Least Mean Square
LOS	Line-of-Sight
MAC	Medium-Access Control
MAHO	Mobile Assisted Handover
MBC	Multiple Bearer Control
MCHO	Mobile Controlled Handover
MSC	Mobile Services Switching Center
MSK	Minimum Shift Keying
NLOS	Non-Line-of-Sight
NSH	Non-Seamless Handover
NWK	NetWorK (layer)
OSI	Open Systems Interconnection
PABX	Private Automatic Branch Exchange
PAM	Pulse Amplitude Modulation
PCM	Pulse Code Modulation
PDU	Packet Data Unit
PER	Packet-Error Ratio
PHL	Physical Layer
PP	Portable Part
PT	Portable Radio Termination
RFP	Radio Fixed Part
RSSI	Radio Signal Strength Indicator
SAP	Service Access Point
SH	Seamless Handover
TBC	Traffic Bearer Control
TDD	Time-Division Duplex
TDMA	Time-Division Multiple Access
U-Plane	User Plane
WRS	Wireless Relay Station

9.16 Integration of DECT Systems into GSM900/1800

This section presents concepts for linking DECT systems to the GSM network. Thereby, an example discussion and outline of the solutions are given for the problems arising when integrating two different systems in a way that they can both be used from a so-called *dual-mode mobile station*. Examples of systems integration not covered here that can be similarily solved are integration of

- GSM and mobile satellite service
- GSM and UMTS
- Any second-generation mobile radio systems.

The DECT fixed network side is not specified in the respective standards, but instead is manufacturer-dependent implemented.

DECT and GSM protocols are designed to establish, to maintain and to disconnect connections (*call control*) and to support mobile station mobility (*mobility management*).

An *interworking unit* (IWU) that establishes connections to external systems is provided in all DECT systems. Manufacturers are therefore required to provide a suitable protocol, corresponding to the BSSAP protocol in the GSM network, between the *DECT fixed system* (DFS) and the IWU in order to translate the DECT system *mobility management* (MM) and *call control* (CC) protocols into those of GSM and vice versa (see Section 3.5). An interworking function is used in the conversion process.

A separation between the DECT and GSM functions is important for this purpose, to differentiate between which functions are to be carried out independently by the DECT fixed network side and which functions should be provided by GSM user administration. The IWU must provide all the data required by the GSM network in a way that enables the GSM network to communicate with the IWU as well as with a BSS. As a result, the IWU on the GSM side must simulate a BSS with all its protocols and timers, and interconvert the MM and CC protocols in the DECT system and the corresponding GSM protocols. To do so, the IWU needs the appropriate information, which is supplied by the DECT system. The IWU on the DECT side must be designed so that it is able to convert all the GSM network protocols into the corresponding DECT protocols. The same applies on the GSM side.

9.16.1 Details on Integration of DECT into GSM 900/1800

9.16.1.1 DECT–GSM Interface

The two systems are interconnected via interfaces. Of particular importance here is the connection of the DECT interworking unit to GSM. It is important

that no modification to the existing GSM and DECT standards should be required and that connection should take place with as minimal a technical effort as possible. Different GSM interfaces will be examined for this purpose.

U_m radio interface The DECT and GSM radio interfaces are structured differently from one another. In addition to having different carrier frequencies, the systems have different channel structures, voice coding and protocol structures. A common DECT and GSM radio interface is not possible, because both systems are rigidly specified at the radio interface. This also would not be desirable because the advantages of DECT over GSM in local area use stem from the fact that it has a different radio interface. The dynamic frequency and timechannel allocation of DECT must be used in order to take useful advantage of the benefits of DECT integration into GSM.

A_{bis}-interface In GSM the BSS is a logical unit consisting of several BTSs and one BSC, with the BTS and BSC connected to each other over the A_{bis}-interface (see Figure 3.1). The BSC controls different BTS functions such as radio channel management, sequence of frequency changes and handover functions over the A_{bis}-interface. Because of the different channel access procedures and channel structures, this information cannot be processed by the DECT system and control information required by the BSC is not provided by DECT. Consequently, this interface also is not suitable for the interconnection of DECT and GSM.

A-interface No control information for the radio interface is transmitted in the GSM over the A-interface between MSC and BSC. Data that is sent over the network layer is from the CM, MM and BSSAP protocols, which are responsible for connection setup and routing (see Figure 3.29). Data is transmitted over ISDN-compliant (ITU-T-Series G.732) PCM-30 systems, which provide 15 full-duplex B-channels with a total of 2,048 Mbit/s. There is also a signalling and a synchronization channel (64 kbit/s each).

In the DECT system dynamic radio channel access and intracell handover control is carried out independently by the mobile stations. The information that the DECT system must transmit over an interworking unit in external networks such as GSM includes the messages of the signalling protocols CC and MM and user data. The user data is either digital voice data that is ADPCM-coded in DECT and RPE-LTP-coded in GSM, or is data from computer-supported applications.

The tasks of the interworking unit at the A-interface are as follows:

Conversion of CC and MM protocols The call control (CC) and mobility management (MM) protocols for the tasks of connection establishment and clearance and for the functions of mobility support are comparable in DECT and in GSM. Protocol conversion in the

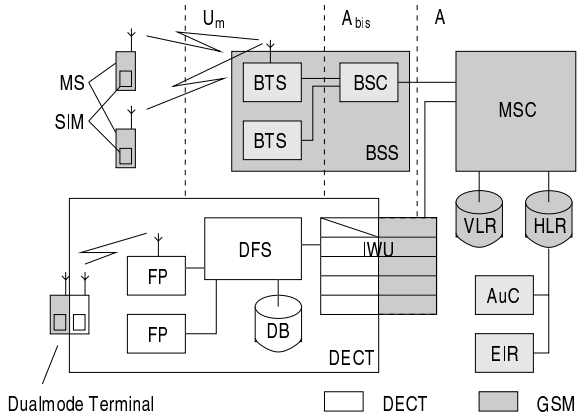


Figure 9.75: DECT connection to GSM A-interface

IWU means converting the content of transmitted messages into the syntax of the other protocol; this will be described in more detail later in this section.

Conversion of voice and data DECT transmits voice ADPCM-coded at 32 kbit/s. The GSM network codes the data at the interface into RPE-LTP at 13 kbit/s. The voice data is transmitted at 64 kbit/s based on the ISDN standard behind a *Transcoder/Rate Adaptor Unit* (TRAU) (see Section 3.2.1.2). Depending on the position of the TRAU, the voice data must be converted from 32 kbit/s to 13 kbit/s (before the TRAU) or 64 kbit/s (after the TRAU). This problem is covered in greater detail in Section 9.16.2.2. Conversion of the data from the format at the DECT radio interface to a normal ISDN format is carried out at the A-interface, in a similar fashion to that for GSM, and has no effect on GSM.

The A-interface is obviously particularly suitable for connection of the DECT and GSM fixed networks. In the DECT Reference Document [4] as well as in various other publications it is referred to as the reference interface [25]. An example of this is illustrated in Figure 9.75.

Other available GSM interfaces Another GSM interface that will be covered briefly is the E-interface between two MSCs. Security-relevant data, such as data sets for authentication that contain RAND and SRES, is transmitted over this interface. This data is not allowed to leave the network of the operator or to be transmitted to a private branch exchange. Along with security-related problems, DECT also cannot assume responsibility for the user administration of an MSC or replace it. This interface therefore does not lend itself to the interconnection of DECT and GSM.

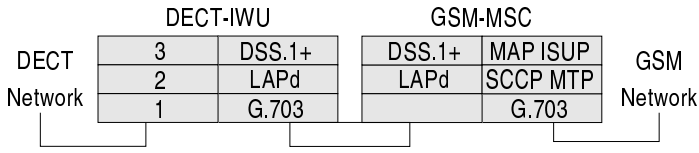


Figure 9.76: DSS.1+ connection between DECT and GSM

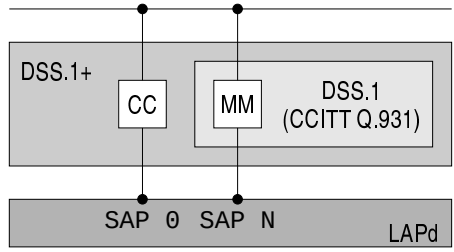


Figure 9.77: DSS.1+ protocol

The other interfaces available in the GSM network are not suitable either, because no user data or signalling data is transmitted over them.

New interface to MSC ETSI is currently examining the feasibility of defining a new standardized interface to MSC. This interface will be based on ISDN and incorporate the following functions:

Layer 1 based on ISDN layer 1.

Layer 2 based on ISDN LAPD.

Layer 3 based on existing ISDN-DSS.1 protocol (in compliance with ITU-T Q.931). The DSS.1 protocol assumes the functions of call control. Functions for mobility management will also be required. This extension to the DSS.1 protocol to include mobility management functions was drawn up by ETSI in 1994. The DSS.1+ protocol therefore has a good chance of becoming a standard for the integration of mobile communications networks at common interfaces [25]; see Figure 9.76 and 9.77.

9.16.1.2 Layer Model and Protocols

The A-interface presents a good option for connecting DECT to GSM. Figure 9.78 illustrates a possible interworking unit at the A-interface of GSM.

Call control in the ISO/OSI reference model is handled by the network layer (layer 3), which provides all the relevant call control and mobility management services needed for these functions.

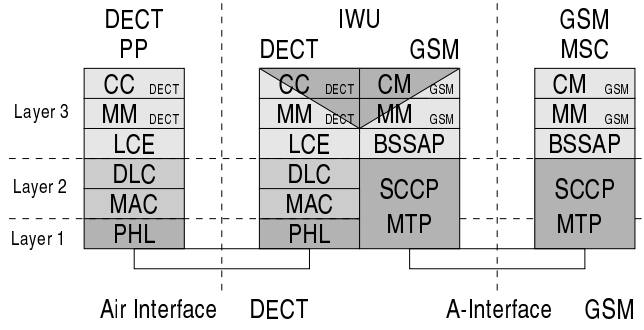


Figure 9.78: Layers of a GSM–DECT complex

GSM is based on protocols SCCP and MTP in layer 1 and layer 2 of the SS 7 protocol stack and on layer-3 protocol BSSAP (*Base Station Subsystem Application Part*). BSSAP is the lowest layer-3 protocol and supports all higher protocols. These are mobility management (MM) and call management (CM), requiring the existence of an MM connection before a CM message can be transmitted. CM consists of three parallel entities: *call control* (CC), *short-message service* (SMS) and *supplementary services* (SS) (see Section 3.5).

The network layer in the DECT system, above the *Data Link Control* (DLC) layer, corresponds in the interworking unit to the GSM network layer (see Figure 9.78). As the lowest sublayer of layer 3, the *Link Control Entity* (LCE) lies above the DLC sublayer, which protects layer-3 messages from transmission error. In addition to the layer-3 CC and MM services shown in Figure 9.78, *Call-Independent Supplementary Services* (CISS), *Connection-Oriented Message Services* (COMS) and *Connectionless Message Service* (CLMS) should also be mentioned.

On each of its two sides, the interworking unit must work with the prescribed protocols, which it converts internally. During the first phase only, the signalling data for connection setup and clearance along with the mobility management data is converted. Plans are underway to extend the functions of the interworking unit to all the supplementary services in GSM.

9.16.1.3 User Data Administration

When two systems are interacting together the tasks of each must be precisely defined in advance to allow consideration of the different possibilities for fulfilling certain tasks.

To enable a *dual-mode unit* to operate in compliance with both the DECT and GSM standards, a mobile station must use an appropriate algorithm to distinguish which mode is being selected. This takes place on the basis of periodically received DECT and GSM frequencies and their evaluation by the mobile station. The decision concerning which system is being used lies decentrally with the mobile station, with priority given to the connection

being made over DECT if the mobile station finds itself in an area covered by DECT. If the mobile station changes location area, it independently informs the network (DECT or GSM).

The network is responsible for user administration and for the mobility management of the users. The design of the DECT system determines exactly how user administration in DECT is to be implemented in each individual case. Three basic systems are considered for the task assignment in user administration.

DECT systems without user administration are the simplest from the viewpoint of design. Mobile stations (PP) are supplied by one or more base stations (FP), which are connected to a control (DFS) without user administration (see, e.g., Figure 9.2). All data is sent to the MSC over an interworking unit and the A-interface (see Figure 9.78). User administration for the DECT mobile stations is carried out in the HLR and VLR databases of GSM, where the GSM mobile stations are also managed.

Advantages Extending GSM user administration to the DECT system simplifies the DFS. This solution is of interest to operators of GSM networks because it allows them to connect DECT indoor systems with little effort, thereby allowing them to make better use of existing GSM resources.

Disadvantages All simple DECT equipment and dual-mode units must be managed by the user administration of the GSM network, where data on the current location area is also being managed. This increases the processing and signalling for GSM.

DECT systems with limited user administration are those systems with a control (DFS) with integrated user administration in which all DECT mobile stations are managed. The dual-mode devices are managed over the interworking function and the A-interface by GSM. The management of the dual-mode devices is similar to the administration of the GSM mobile stations in the GSM network.

Advantages DECT mobile stations are subject to the user administration within DECT, which reduces the signalling required between GSM and DECT. The GSM resources in the fixed network are not overburdened by the DECT mobile stations.

Disadvantages DECT user administration must be integrated into the DFS: security-related files must be protected against unauthorized access. A distinction has to be made between a DECT device and a dual-mode device in DECT mode for connection establishment and authentication.

DECT systems with full user administration manage all DECT and dual-mode equipment within the coverage area.

Advantages These systems are those that relieve GSM the most from the administration of the dual-mode equipment. Information on the equipment is stored in the GSM-VLR, whereas the location area stored in the VLR relates to the DECT coverage area. With its own complete user administration, DECT is assured of being differentiated from other external systems such as GSM. It means that this system constitutes its own branch exchange, which can be linked to any open system through the selection of an appropriate interworking unit.

Disadvantages An independent system requires its own user administration. Despite this, a link must exist to GSM for the dual-mode devices to allow them to be authenticated for communication. GSM treats the DECT system behind the interworking unit as a GSM base station system (BSS). Consequently, the mobile switching centre in GSM will not allow any calls from mobile stations that have not been authenticated beforehand. In this case each dual-mode device from the DECT system must be registered and authenticated in the GSM network.

The self-sufficiency of branch exchange systems that manage their users themselves is a definite advantage. The user administration function incorporated within DECT enables DECT systems to be connected to any telecommunications network through an interworking unit. Connection to GSM is a special case, and will be dealt with accordingly. GSM takes responsibility for global user administration in wide-area roaming — something that is not possible with DECT.

9.16.1.4 Security Requirements

Numbering Only one single calling number should be required within and outside the DECT coverage area for the entire system. In DECT there are seven different possibilities for assigning an identity to a mobile station.

`IMSI↔IPUI` As shown in Figure 9.79, an *International Portable User Identity* (`IPUI`) `Type R` is used for DECT. It consists of a *Portable User Type*, `PUT`) 4 bits in length, and a *Portable User Number* (`PUN`). The `PUN` consists of 60 bits and contains a GSM-compliant identity number called an *IMSI*. This IMSI can be used to reach a dual-mode device over GSM (see Section 9.2.1.2). When the DECT coverage area is entered, DECT uses an `IPUI Type R` with the same `IMSI`.

`TMSI↔TPUI` In GSM, mobile stations are assigned a temporary identity `TMSI`. This number is stored in the VLR and is used to contact a mobile station.

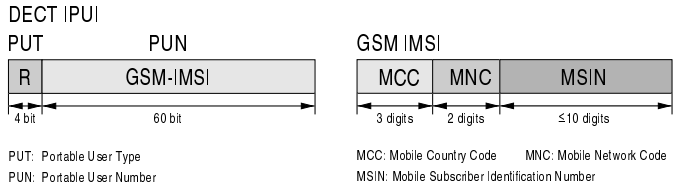


Figure 9.79: Comparison of `IMSI` and `IPUI`

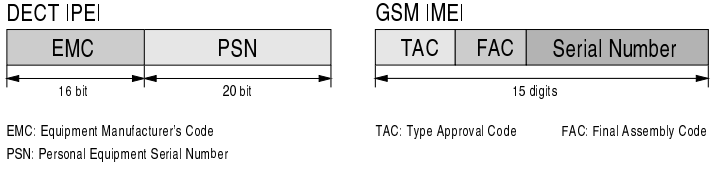


Figure 9.80: Comparison of `IMEI` and `IPEI`

The number can also be used under DECT, in which case the `TMSI` is regarded as an identity that has been assigned by the DECT system. An element in DECT allows the use of the GSM-TMSI: The TMSI accepts the *Network Assigned Identity* element described below with a maximum of four octets (octets 5–8) if the type indicated in octet 3 with `1110100` has been declared as a GSM *Temporary Mobile Subscriber Identity*. In the DECT system the `TMSI` becomes a *Temporary Portable User Identity* `(TPUI)` [17].

`IMEI`↔`IPEI` It is difficult to identify dual-mode devices using the equipment identity. DECT and GSM use two different structures for the `IMEI` (GSM) (see Section 3.2.1.4) and the *International Portable Equipment Identity* `IPEI` (DECT); see Figure 9.80. In GSM the equipment identity consists of a 15-digit number, which is made up of a *Type Approval Code* (`TAP`), a *Final Assembly Code* (`FAC`) and a serial number. In contrast to the `IMEI`, the `IPEI` consists of 36 bits. The first 16 bits contain the *Equipment Manufacturers Code* (`EMC`), followed by a 20-bit long *Portable Equipment Serial Number* (`PSN`).

Location area codes A DECT system can be regarded by GSM as a location area. It is possible to differentiate between several location areas within a DECT system, although the DECT location areas are not allowed to extend beyond the GSM location areas [25]. The reason is to avoid any unnecessary updating traffic in GSM. The location areas have their own identification, which is continuously transmitted. This identification is the *Location Area Code* (`LAC`) in GSM. Within DECT, a `LAC` corresponding to the GSM `LAC` can be assigned to the DECT location areas. The `LAC` is transported with the `Location Area Element`, which

is also provided for the transport of a GSM LAC. DECT therefore uses a GSM-based LAC.

Encryption Coding on the GSM radio channel begins in the mobile station and ends in the BTS or BSC (see Figure 3.1). No coded data is transmitted at the A-interface. The ciphering key is calculated in the MSC and in the MS (see Section 3.13), and the coding is undertaken in three gradual stages. DECT codes its data itself on the radio channel, and requires no external control mechanism to do so.

When DECT and GSM are connected at the A-interface, it is not necessary for the DECT system to make a request for encryption. However, the MSC attempts to initiate encryption on the basis of GSM guidelines. An interworking unit must respond accordingly to prevent the MSC from cutting off the connection.

The following steps can be taken:

Change to MSC control A change to its software will allow the MSC to detect the A-interfaces to which the DECT system is connected. When this happens, no encryption is begun by the MSC. This option requires a modification to the GSM system.

Encryption refusal Encryption is initiated by the BSSMAP message Cipher_Mode_Command from the network (see Figure 3.87). The interworking unit can then refuse the encryption using Cipher_Mode_Reject. The reason is sent as Ciphering_Algorithm_not_supported, with the BSS usually indicating that encryption cannot be carried out. Since the BSS has been replaced by an interworking unit, the latter can send an appropriate response.

Encryption of an imaginary radio channel If the interworking unit receives a GSM-BSSMAP_Cipher_Mode_Command message over the A-interface, it can forward a DECT-MM_Cipher_Request message to the DECT mobile station. The mobile station responds to this request with a coded message. Upon receipt of the first coded message, the IWU can send the MSC the reply GSM-BSSMAP_Cipher_Mode_Complete. This confirms to the network that the radio interface is coded, with the DECT system using its internal encryption mechanism.

Authentication Authentication of dual-mode devices must be assured for both GSM and DECT. The systems use different procedures for the authentication process, although smart cards on which information such as algorithms for identification are stored are common to both. DECT uses a form of mutual authentication in which both the mobile station and the network must be authenticated. With GSM authentication is only required of the mobile station. The preferred form of authentication when DECT and GSM are interconnected is the use of an appropriate smart card to authenticate both

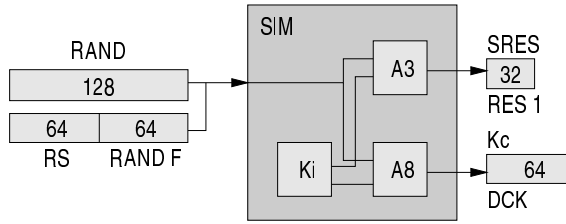


Figure 9.81: Authentication in DECT using SIM

systems. There are two possibilities for comparison: the use of the SIM of GSM also for DECT, or the use of the DAM of DECT also for GSM. Both methods are presented here.

DECT authentication using SIM Under certain conditions, the *Subscriber Identity Module* (SIM) of GSM (see Section 9.2.3) can also be used for DECT authentication:

- The *fixed terminal* (FT) in the DECT system must be made aware that GSM authentication is being carried out in a dual-mode unit. This is conveyed through the use of DECT encryption algorithms for the radio channel and DECT protocols.
- The standard lengths of all available DECT authentication parameters are used. The parameters `RS`, `RAND F` and `DCK` have a length of 64 bits; `RES 1` consists of 32 bits.
- Authentication of SIM and DAM (*DECT Authentication Module*) are called up by the same command RUN_GSM_ALGORITHM.
- The protocol and the interface between DAM and the DECT *portable part* (PP) are identical to the corresponding protocol and interface between SIM and a GSM mobile station in the GSM network (see Figures 3.84, 3.85 and 9.81).
- A *session key* (`SK`) is not allowed to be used in DECT.

The processor of the PP commences authentication using a command similar to the RUN_GSM_Algorithm one. It sequentially inputs `RS` (64 bits) and `RAND F` (64 bits) from the SIM, and interprets them as `RAND` (128 bits). The SIM carries out the GSM algorithms A3 and A8. The results `SRES` (32 bits) and `Kc` (64 bits) are produced and interpreted as `RES 1` (32 bits) and `DCK` (64 bits), and used accordingly in the DECT system.

GSM authentication using DAM Another option is the use of the DAM for GSM authentication. Dual-mode units operating outside a DECT coverage

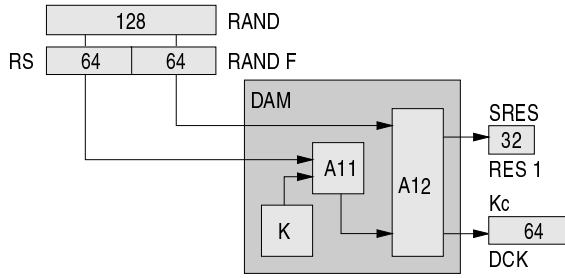


Figure 9.82: Authentication in DECT using DAM

area must be authenticated in GSM, and a possibility using DAM for this purpose is provided. The following requirements must be met:

- The GSM network side must be made aware that DECT authentication is taking place. The dual-mode unit uses the GSM protocols and GSM coding on the radio channel to convey this. The authentication key `K` must be known to the GSM network.
- Only the standard lengths of the DECT parameters are used. These include 64 bits for `RS`, `RAND F` and `DCK`, and 32 bits for `RES 1`.
- Authentication by SIM and DAM is called up through the same procedure.
- The protocol and the interface between DAM and the DECT portable part are identical to the corresponding protocol and interface between SIM and the GSM mobile station in GSM.

The processor in the GSM mobile station uses Run_GSM_Algorithm to initiate the authentication. It sequentially inputs the number `RAND` (128 bits) into the DAM, with the first 64 bits considered as `RS` and the next 64 bits as `RAND F` (see Figure 9.82). The DAM carries out the DECT calculations and obtains `RES 1` (32 bits) and `DCK` (64 bits). `RES 1` is interpreted as `SRES` and `DCK` as `Kc`, and as such also used in the GSM network.

In practice, the first method involving authentication with the GSM SIM for DECT is the one used when the requirements described above are met.

9.16.1.5 Handover

Handovers are not a problem as long as they take place within a system (within GSM or within DECT).

GSM to GSM In GSM a distinction is made between Intra-BSC and Inter-BSC handovers, and both are controlled by the BSS (see Section 3.6). The Inter-BSC handover is of particular interest here because at the

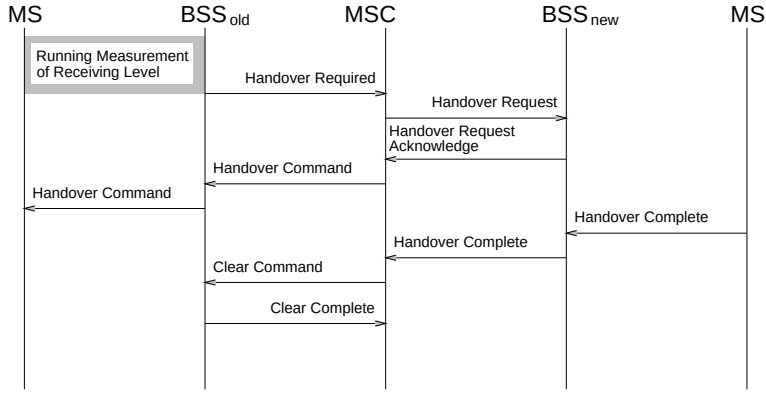


Figure 9.83: Inter-BSC Handover in GSM

MSC level it can be switched between different BSCs or to a DECT coverage area. The signal levels of the mobile stations are measured and evaluated, and additional information is passed on to the network in the form of corresponding receive signal level values. If the signal level is too low, the BSS sends a Handover_Required to the MSC (see Figures 9.83 and 3.52). The MSC then sends a Handover_Request to the new BSS, which responds with Handover_Request_Acknowledge. During this process, a handover reference number is agreed that allows the mobile station to be recognized by the new BSS and receive its calls. The MSC now sends a Handover_Command message to the former BSS and from there to the mobile station. The mobile station thereupon reports to the new BSS with Handover_Complete and the agreed reference number. The new BSS forwards a Handover_Complete to the MSC, which then proceeds to clear the old connection using Clear_Command. The old BSS acknowledges this message with Clear_Complete.

DECT to DECT DECT makes use of a decentral algorithm that is controlled by the mobile station. The DECT mobile station compares the receive levels of the base stations capable of receiving, and switches as quickly as possible to the base station with the higher receive level. The process continues with the establishment of a new connection to the new base station. As soon as the new connection has taken place, the call from the old connection is rerouted to the new one, and the old one is terminated. The handover process in the DECT system must be capable of switching ongoing calls over a new base station and a newly established connection without any loss of data. The DECT mobile station must also be capable of switching over from an old connection to a new one for a call in progress. The DECT control (DFS) must be informed any time a mobile station changes to a new location area.

When a connection exists between DECT and GSM, handovers can also be executed between the systems as described above. Two cases should be noted:

GSM to DECT A handover from GSM to DECT is only possible if the signal quality of the GSM connection is high enough. The dual-mode unit sets up a connection to DECT and gives notification of its request for handover. Furthermore, the BSS in the GSM that makes the handover decision must be instructed to carry out the handover. One possibility is for the mobile station to report to the network that the receive signal is poor. If the BSS has decided to carry out the handover, the same procedure is followed as illustrated in Figure 9.83. As soon as the GSM mobile station receives a Handover_Command from the old BSS, it registers itself as a DECT-PP in the DECT system. The IWU of DECT must now send a Handover_Complete message to the GSM-MSC so that the old connection can be terminated. One way of reporting this to the IWU is to send `IWU_Packet` elements. The IWU recognizes the executed handover and notifies the GSM accordingly. The GSM-MSC switches the connection to the new BSS, which is replaced here by a DECT-IWU.

DECT to GSM A handover from DECT to GSM is initiated when a mobile station exits a DECT coverage area. Since DECT has a smaller range of coverage, a call is more likely to be broken off owing to shadowing than in the other direction if a handover is not executed in time. Any deterioration in the DECT receive signal must therefore be detected by the dual-mode unit, which then requests a handover from the IWU. The IWU then sends a Handover_Required to the MSC. The procedure continues as described above. After a Handover_Command has been sent from the MSC to the IWU, the IWU then sends a Handover_Complete to the new BSS. The connection to DECT is terminated.

A closer look at handover control is required to determine what is involved in terms of hardware and software for a handover.

It is not clear to what extent handovers that are executed beyond system boundaries should be considered. These handovers do not represent a crucial function [25], and for the time being are not being planned for the first implementations.

9.16.1.6 DECT Elements for GSM Integration

The connection to GSM has already been prepared in various DECT messages. This is advantageous because it means that some of the message elements in the IWU no longer need to be converted and can be forwarded from the GSM to the mobile station without any further processing. This applies in particular to the following:

`Auth-Type Element` The `Auth-Type Element` is used to convey the authentication algorithm of the DECT system, with the `0000 0001` coding identifying the DECT standard authentication algorithm. A `0100 0000` identifies authentication in accordance with the GSM standard, which means that messages in which the `Auth-Type Element` is transmitted can be referred to the GSM standard algorithm. These messages are:

- MM_Authentication_Request
- MM_Authentication_Reject
- MM_Access_Rights_Request
- MM_Access_Rights_Accept

`Identity Type Element` This describes the type of identity used by the network to contact mobile stations. This element of the messages consists of four octets. The identity group is specified in bits 1–4 in octet 3 of this element. For DECT systems `0000` is entered here, which means *portable identity*. Based on this, the type is specified in bits 1–7 in octet 4. `000 0000` stands for *International Portable User Identity* (`IPUI`), `001 0000` for *International Portable Equipment Identity* (`IPEI`) and `010 0000` for *Temporary Portable User Identity* (*TPUI*).

For DECT systems linked to GSM, identity group `0001` is provided in octet 3, which refers to the *Network Assigned Identity* group. This is followed by a `111 0100` in bits 1–7 in octet 4, which means that a mobile station is being contacted via the GSM *Temporary Mobile Subscriber Identity* (`TMSI`). The identity type is used in the following messages:

- CC_Release_Complete
- MM_Identity_Request

`IWU Attributes Element` These information elements are used to exchange compatibility information between mobile stations and the interworking unit. The information is exchanged through the use of `IWU Packet Elements` and is not relevant to the DECT system; in other words, the DECT system is transparent to this element.

The GSM 04.06 LAPDm protocol, which is coded with `1 0010` in bits 1–5 and is an ISO/OSI layer-2 protocol, can be indicated in octet 8 for GSM. This element is used in the following messages:

- CC_Setup
- CC_Call_Complete
- MM_Identity_Request

`IWU Packet Element` The `IWU Packet Elements` can accept all external structured or unstructured data and transport it between the mobile stations and the interworking unit. When the external data is too voluminous to be transported in an `IWU Packet Element`, it can be divided into several `IWU Packet Elements` through the use of `Segment Info Elements`. `IWU Packet` is used as an optional element in almost all messages. The IWU-Information message, which then has the sole task of transporting the `IWU Packet Element`, can be used for transmission.

`IWU to IWU Element` The `IWU to IWU Elements` are used to transport any messages that cannot be sent by any other DECT element. As with the `IWU Packet`, this data too can be segmented. `IWU to IWU` is used as an optional element in almost all messages. For transmission, the `IWU to IWU Element` can be attached to the `IWU-Information` message if no other messages are being sent. The task of the `IWU-Information` message is to transport the `IWU-Elements`.

`Location Area Element` The `Location Area Element` transmits the identification of a location area. The *GSM Location Information* can be transmitted with the coding `1111` in bits 5–8 in octet 4. This consists of the *Mobile Country Code* (`MCC`), the GSM *Mobile Network Code* (`MNC`), the *GSM Location Area Code* (`LAC`) (see Figure 3.57) and the *GSM Cell Identity* (`CI`). The `CI` is required for external handovers. The location element is transported in the following messages (in some of them optionally):

- CC_Information
- CC_Release_Complete
- MM_Access_Rights_Accept
- MM_Info_Accept
- MM_Info_Suggest
- CC_Setup_Acknowledge
- MM_Locate_Accept
- MM_Locate_Request
- MM_Info_Request

`Network Assigned Identity Element` This element is used to transmit the identity assigned by the network. With the coding `111 0100` of the bits 1–7 in octet 3 of the element, it corresponds to the GSM *Temporary Mobile Subscriber Identity* (`TMSI`) number used by the network. The length of the `TMSI` should not be longer than four octets. This element is transmitted in the following messages:

- CC_Information
- MM_Identity_Reply
- MM_Locate_Request
- MM_Info_Request
- MM_Temporary_Identity_Assign
- MM_Detach
- MM_Locate_Accept
- MM_Info_Accept
- MM_Info_Suggest

`Portable Identity` This element transmits the unique identity of a mobile station within DECT. The identity can be more accurately specified through a type in octet 3. `000 0000` stands for `IMSI`, `001 0000` for `IPEI` and `010 0000` for `TPUI`. If properly allocated, the identities mentioned above can correspond to the numbers of GSM. Type R of the `IPUI` is especially designed to accept a GSM `IMSI`. The `TPUI` can accept an identity assigned by the network that corresponds to a GSM `TMSI`.

`Network Parameter Element` This is used to transmit network data such as handover reference numbers that are required by GSM for handover execution. The `Network Parameter Element` is transmitted within the following messages:

- CC_Information
- MM_Info_Accept
- MM_Info_Suggest
- CC_Release_Complete
- MM_Info_Request

9.16.2 Interworking Unit DECT-GSM

An interworking unit connects two different systems together, converts protocols and recodes data. The interworking unit between DECT and GSM is responsible for the correct conversion of the call control (CC) and mobility management (MM) protocols that are used during the signalling phases. In addition, it is necessary for the voice signals of the U-plane to be recoded and the data transmission protocols of the radio interface to be carried out correctly and in some cases converted.

9.16.2.1 Conversion of Signalling Messages

DECT and GSM work with similar protocols, whose protocol data units (PDU) are converted by the interworking unit. This conversion takes into account the structure and content of individual messages, and determines how they are schematically processed in synchronization with each other.

A comparison of the content and structure of the MM and CC messages follows.

MM Procedures The individual MM messages are structured differently, but in some cases have the same content and similar functions. Comparable messages are listed in Table 9.14. They have the same tasks and to the extent possible are merged together in an interworking unit.

The different messages can be converted into each other relatively easily because their content is similar and they have the same tasks. The different messages carry out the same functions in their systems, and when converted are merged together. The structure of GSM messages is as shown in Figure 9.84; the structure of DECT messages is as shown in Figures 9.85 and 9.47. The header of both types of messages consists of a `Protocol Discriminator` (4 bits) and a `Transaction Identifier` (4 bits). This is followed by the message type, which is coded differently in each system. The messages in Table 9.14 that are not applicable to both systems are special functions that relate to the respective system and cannot be converted because no comparable messages exist in the other system.

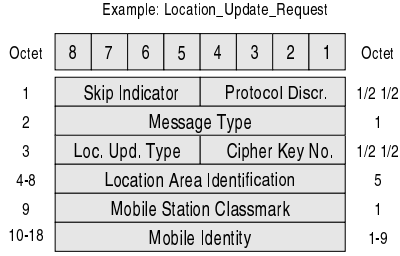


Figure 9.84: MM message structure in GSM using Location_Update_Request

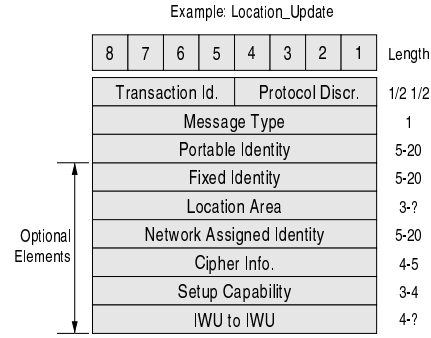


Figure 9.85: MM message structure in DECT using Locate_Update

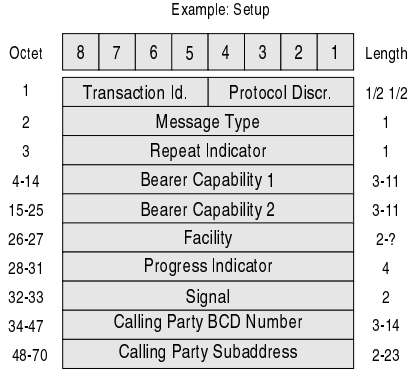


Figure 9.86: CC message structure in GSM using Setup

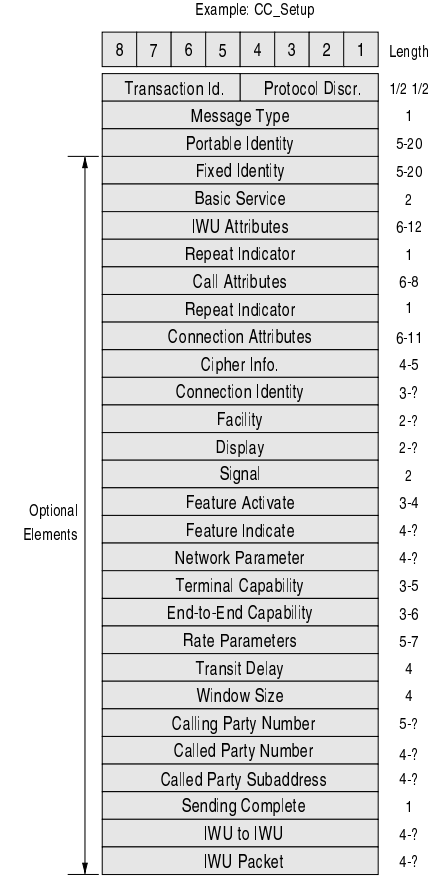


Figure 9.87: CC message structure in DECT using CC_Setup

Table 9.14: Comparison of MM messages in GSM-DECT

GSM message	Direction	DECT message	Direction
Location_Update_Req.	MS-Network	Locate_Request	PP-Network
Location_Update_Acc.	Network-MS	Locate_Accept	Network-PP
Location_Update_Rej.	Network-MS	Locate_Reject	Network-PP
IMSI_Detach_Ind.	MS-Network	Detach	PP-Network
Authentication_Req.	Network-MS	Authentication_Req.	both
Authentication_Res.	MS-Network	Authentication_Reply	both
Authentication_Reject	Network-MS	Authentication_Reject	both
Identity_Request	Network-MS	Identity_Request	Network-PP
Identity_Response	MS-Network	Identity_Reply	PP-Network
TMSI_Realloc._Compl.	Network-MS	Temp._Ident._Assign	Network-PP
TMSI_Realloc._Compl.	MS-Network	Temp._Ident._Assign_Ack.	PP-Network
MM_Status[a]	both	Temp._Ident._Assign_Rej.	PP-Network
CM_Service_Request	MS-Network	LCE	
CM_Service_Accept	Network-MS	LCE	
CM_Service_Abort	MS-Network	LCE	
CM_Service_Reject	Network-MS	LCE	
CM_Reestab._Request	MS-Network	LCE	
Abort	Network-MS	CC	
		Access_Rights_Request	PP-Network
		Access_Rights_Accept	Network-PP
		Access_Rights_Reject	Network-PP
		Acc._Rights_Terminate_Req.	PP-Network
		Acc._Rights_Terminate_Acc.	Network-PP
		Acc._Rights_Terminate_Rej.	Network-PP
		Key_Allocate	Network-PP
		MM_Info_Suggest	Network-PP
		MM_Info_Request	PP-Network
		MM_Info_Accept	Network-PP
		MM_Info_Reject	Network-PP
BSSMAP		Cipher_Suggest	PP-Network
BSSMAP		Cipher_Request	Network-PP
BSSMAP		Cipher_Reject	PP-Network

[a]MM Status provides general information about a state

CC Procedures As is the case with the MM service, there are messages in the call control service that are comparable to one another, i.e., have the same tasks. These are listed in Table 9.15.

The individual messages listed next to each other in Table 9.15 can be interconverted relatively easily because they have the same content and serve the same task. However, the structure also varies here; see Figures 9.86 and 9.87 (same as Figure 9.48). Moreover, certain messages in GSM and DECT refer to system-specific characteristics and cannot be converted.

It should be noted that the objective with both services and the related protocols (MM and CC) was to limit the functions of DECT to the functions

Table 9.15: Comparison of CC messages in GSM-DECT

GSM message	Direction	DECT message	Direction
Setup	both	CC_Setup	both
Emergency_Setup	MS-Network	CC_Setup	both
Alerting	both	CC_Alerting	both
Connect	both	CC_Connect	both
Connect_Acknowledge	both	CC_Connect_Acknowledge	Network-PP
Call_Proceeding	Network-MS	CC_Call_Proceeding	Network-PP
Release	both	Release	both
Release_Complete	both	Release Complete	both
Facility	both	Facility	both
Hold	MS-Network	Hold	both
Hold_Acknowledge	Network-MS	Hold_Acknowledge	both
Hold_Reject	Network-MS	Hold_Reject	both
Retrieve	MS-Network	Retrieve	both
Retrieve_Acknowledge	Network-MS	Retrieve_Acknowledge	both
Retrieve_Reject	Network-MS	Retrieve Reject	both
Notify	both	CC_Notify	Network-PP
Start_DTMF	MS-Network	contained in all messages	both
Stop_DTMF	MS-Network	contained in all messages	both
Modify	both	CC_Service_Change	both
Modify_Complete	both	CC_Service_Accept	both
Modify_Reject	both	CC_Service_Reject	both
User_Information	both	IWU_Information	both
Congestion_Control	both		
Progress	Network-MS		
Call_Confirmed	MS-Network		
Disconnect	both		
Start_DTMF_Ack.	Network-MS		
Start_DTMF_Reject	Network-MS		
Stop_DTMF_Ack.	Network-MS		
Stop_DTMF_Reject	Network-MS		
Status_Enquiry	both		
Status	both		
		CC_Setup_Acknowledge	Network-MS
		CC_Information	both

of GSM. This applies to the messages which are sent from the network to the IWU. Problems occur when a message from the IWU cannot be converted to a DECT message. This is the case when no corresponding partner message exists for a GSM message (Network-MS) in Tables 9.14 and 9.15.

The GSM functions become important for the interworking unit when GSM requests a response from the IWU or when DECT requires the IWU to request a service from the GSM network. This affects DECT messages that are sent to the IWU but cannot be converted there.

9.16.2.2 Transmission of Voice Data

In DECT voice data is coded in accordance with the ADPCM standard at 32 kbit/s [12]. This speech must be transcoded by the interworking unit into a standard that is acceptable for external networks. A reference point where digital voice data is available in a specified form in compliance with the PCM standard is defined for digital external networks.

In GSM digital voice data is coded in accordance with RPE-LTP at 13 kbit/s and transmitted over the radio channel. A transcoder and rate adaptor unit (TRAU) that recodes the GSM-coded speech into PCM at 64 kbit/s is found in the BTS. RPE stands for *Regular Pulse Excitation* and LTP for *Long-Term Prediction*. Although the TRAU is functionally part of the BTS, it is physically located in the MSC. The advantage of this is that voice data can be transmitted over the A- and A_{bis}-interfaces at a lower data rate of 16 kbit/s.

GSM conforms with ISDN, and uses an ISDN multiplex channel with 2 Mbit/s in accordance with ITU-T G.703 at the A-interface. Thus 30 data channels with 64 kbit/s and two signalling channels are available at this interface. These 64 kbit/s channels are divided through submultiplexing into four submultiplex channels with 16 kbit/s each over which the 13 kbit/s voice data is transmitted.

The interworking unit between DECT and GSM must guarantee that the voice data is routed to GSM with the correct coding and at the correct data rate, and vice versa. GSM is able to process PCM-coded voice data with 64 kbit/s if the multiplex line is not connected to a TRAU in the MSC. However, GSM can also process RPE-LTP-coded voice data at 13 kbit/s if it is converted into 64 kbit/s signals in a TRAU in the MSC. Consequently there are two different connection options:

- For connection of DECT to GSM the DECT voice data can be reduced to 13 kbit/s using RPE-LTP and transmitted on four submultiplex channels per 64 kbit/s channel. The advantage is a low transmission rate at the A-interface, which substantially increases the transmission capacity. The disadvantage with this procedure is that there is a deterioration in the DECT speech quality at the GSM level.

- GSM is able to switch a connection to a DECT coverage area without using a TRAU because of the ability of the interworking unit to supply the data at a 64 kbit/s PCM rate for the MSC. A transcoder of the interworking unit converts the user data of the DECT PCM reference point into the form of data at the PCM A-reference point of GSM and in reverse [12].

  The advantage is a high level of voice quality from DECT to other digital terminals operating at the same high data rate. The disadvantage is the added traffic load on the 2 Mbit/s transmission path at the A-interface.

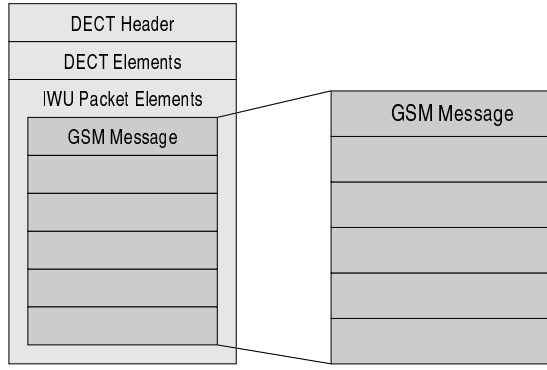


Figure 9.88: Structure of a DECT message that transports a GSM message

The voice data must be converted, because DECT provides ADPCM speech with 32 kbit/s at the reference interface.

9.16.2.3 Alternative Signalling

An alternative form of signalling would be possible if, for signalling with GSM, the dual-mode units used the GSM messages, which are regarded as *external user data* and are transported unchanged when a connection to DECT exists. The DECT system is transparent to this signal transmission. The IWU packet elements (see Figure 9.88) carry out the transport by tunnelling the original messages.

Conversion of the CC and MM protocols between DECT and GSM does not apply to the transparent transmission of signalling data. The DECT system offers a suitable element for this in the form of a `IWU Packet Element`, which allows external user data to be transmitted transparently through DECT between the mobile station and the attached interworking unit. GSM messages at the IWU are declared as external user data and forwarded over an existing DECT connection to the dual-mode unit through the use of `IWU Packet Elements`. These elements can be transported in most of the CC and MM messages of DECT. If at any time there is no message to which this kind of element can be attached, DECT then sends an IWU_Information message that applies only to the transport of these elements. The DECT messages are processed at the destination (mobile station or interworking unit), and the content of the `IWU Packet Elements`, which corresponds to the GSM messages, is forwarded to the GSM part of the dual-mode unit or the interworking unit. Transmission in both directions is in accordance with the same scheme. It should be noted that a DECT CC connection established in the same way as any other normal DECT connection is required for forwarding the `IWU Packet Elements` before the first GSM message is transmitted.

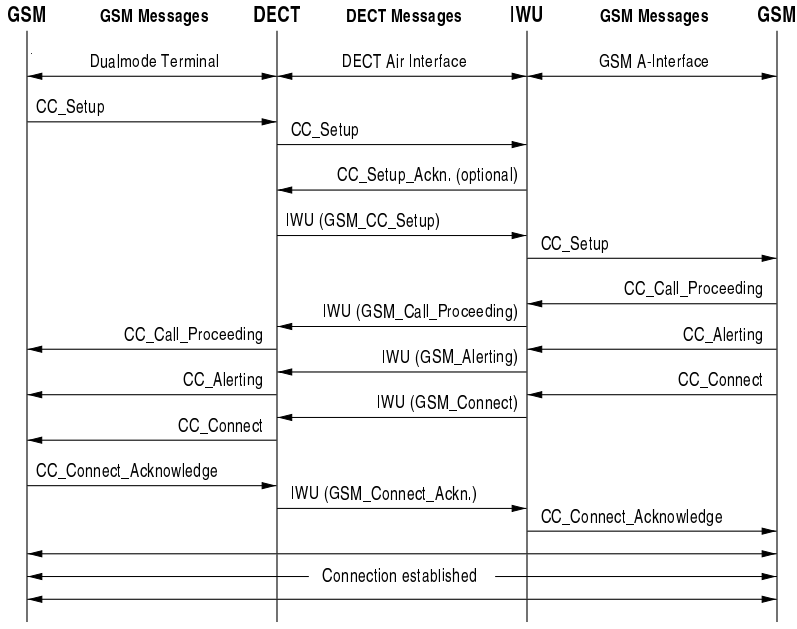


Figure 9.89: Connection setup process

The only aspect that is eliminated from this procedure is the conversion of the signalling data at the IWU. The user data with different coding must continue to be recoded by an IWU. Furthermore, the dual-mode units must first be connected to the DECT system before they can set up a connection to GSM.

The disadvantage of this solution is that more data has to be transported over the DECT radio channel than if the messages had been converted into a DECT format. Moreover, with transparent transmission all signalling data must be transmitted for both DECT and for GSM. With conventional message conversion different messages used for DECT are also used in a similar form for GSM.

Hardware Problems occur because of the volume of transmitted data and because of the security procedures. DECT authentication and coding must be carried out correctly in the dual-mode units. Once a DECT connection exists, GSM authentication is necessary.

Schematic Procedures An example of a message procedure during transparent transmission is illustrated in Figure 9.89. It shows that a Setup message is transmitted twice: once for DECT and once for GSM. The additional signalling along with the added burden on the radio interface account for the extra time required for setting up a connection.

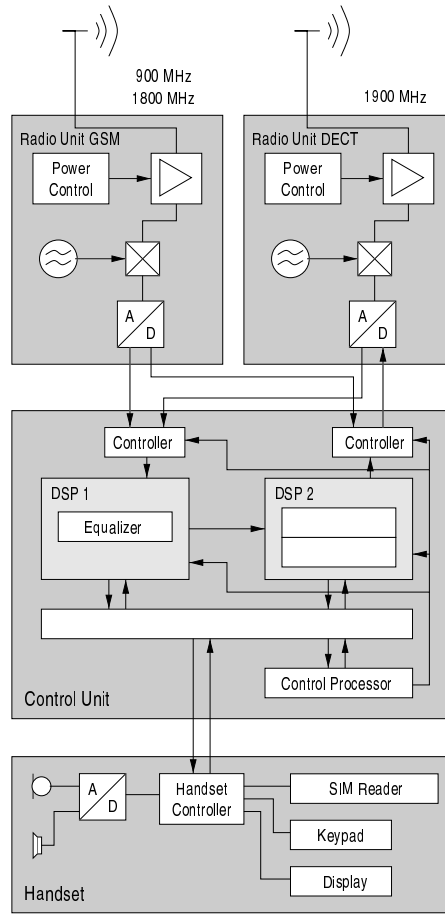


Figure 9.90: Block diagram of a dual-mode unit

9.16.3 Dual-Mode Units for DECT-GSM

A *dual-mode unit* is a subscriber unit that can be used to set up a connection over DECT or over GSM. The selection of the system used by the unit is controlled by an appropriate algorithm that gives priority to connecting over DECT. The advantages lie in the higher availability of DECT radio channels and the cost savings of private DECT branch exchanges. A dual-mode unit also serves as a standard GSM mobile station.

If a dual-mode unit is switched on and in standby mode, it periodically listens in on the GSM and DECT frequencies. GSM base stations send out an identification that allows a mobile station to detect the current location area. If the location area identification (`LAI`) received does not agree with the one stored in the SIM, the GSM location area is updated. If the unit enters

a DECT coverage area, a location update is carried out over DECT and the DECT area is indicated as the new location area to GSM. The DECT system then administers the dual-mode unit as a DECT unit.

Except for the two high-frequency parts and associated control functions, a dual-mode unit is structured the same way as a GSM unit.

Mobile stations can be divided roughly into three functional units: radio unit, control unit, and the handset as the interface between user and mobile station (see Figure 9.90).

Radio unit The radio unit establishes the radio link between mobile and base station and contains all the relevant modules. The digital data from the control unit is converted into analogue signals in a digital–analogue converter, modulated, amplified and transmitted to the antenna.

Signals received from the antenna are amplified, then demodulated, converted into digital signals through an analogue–digital converter, and forwarded to the control unit [1].

A dual-mode unit requires two radio units — one operating on the GSM frequencies (900 MHz) and one on the DECT frequencies (1900 MHz) — and a common control unit. It is possible to combine both radio units into one unit, and these kinds of units are available in the market.

Control unit The control unit contains signal processors for voice and channel coding and for the equalizer. Both signal processors are connected to an *input/output expansion device* (IOX) that has an interface to the handset. All the parts are controlled by a control processor.

A dual-mode unit that communicates over DECT as well as over GSM can use both signal processors to process the signals for both modes of operation, see Figure 9.90. If required, the corresponding software can be loaded from a storage unit. The unit then becomes a so-called *software defined radio*, which is capable of changing its transmit/receive signal processing through software control.

Handset The handset is the interface between user and mobile station. The digital signals that emanate from the handset controller are converted into analogue signals for the speaker by a digital–analogue converter. Correspondingly, the analogue signals coming from the microphone are converted into digital voice data in an analogue–digital converter. The display, keyboard and SIM card reader are controlled over the handset controller.

The handset of a dual-mode unit is the same as that of a GSM or DECT terminal. No modifications are required to the displays, keypad, speakers or microphone. The SIM card reader must also be able to read a DAM. The software must be adapted to dual-mode operation.

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10
Wireless Local Loop Systems

Wireline communications networks are expensive and time-consuming to configure. Growing demand for telecommunications services and the imminent liberalization of the telecommunications market in Europe have greatly increased the interest in radio-based communication (*Wireless Local Loops*, WLL), also referred to as *Radio in the Local Loop* (RLL), see [3].

Future competitors of telecommunications monopoly service providers, such as power suppliers who already have their own wide-area networks, will usually lack access networks to the end users. Message transfer over existing power supply lines is currently being attempted using CDMA transmission, and allows rates of several Mbit/s. The CEN 50 065 standard, which enables the use of carrier frequencies of up to 140 kHz, is available for this purpose. In 1997 Northern Telecom conducted its first field tests in this area in Great Britain. If successful, the power supply network could be used in the future as a user access network for voice and data services, playing a major role in promoting competition and price reductions in telecommunications services.

Meeting the following requirements, wireless technology is very effective at bridging the "last mile" of the path between the fixed network access of private network operators (*Point of Presence*, POP) and the user:

- Fast and economic network configuration
- Economic network operations
- Flexible and expandable network structure
- Possible added-value through (restricted) mobility

For the user, a connection over a WLL system replaces the wireline fixed network connection and should guarantee the same quality of service characteristics—bit-error ratio, delay time and blocking probability—as the public analogue telephone network (*Public Switched Telephone Network*, PSTN) or even the *Integrated Services Digital Network*, ISDN.

Figure 10.1 shows the different levels of mobility for a WLL user. In addition to connections without mobility in which the customer uses a normal telephone connected through a cable to an antenna in the wall of the house, systems with restricted mobility also exist, allowing a user to roam in a restricted area, e.g., in a radio cell defined by an internal base station. Systems in which users enjoy the same kind of mobility as in cellular mobile radio

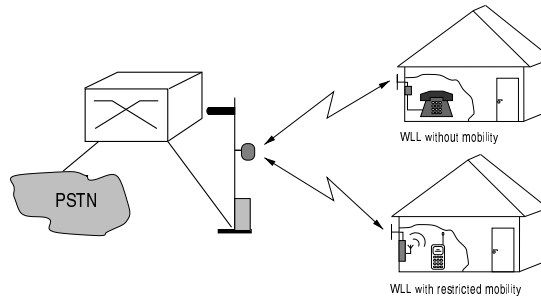


Figure 10.1: WLL without and with restricted mobility

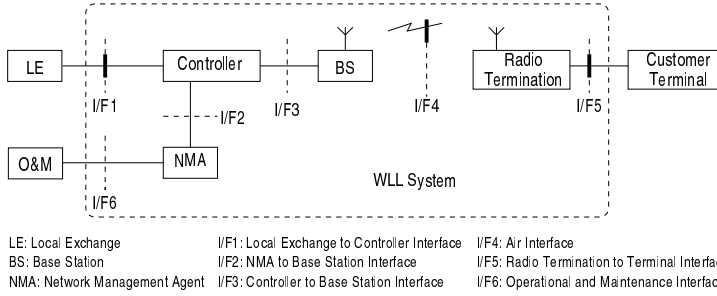


Figure 10.2: ETSI reference model for WLL systems

networks (e.g. GSM) are called *Personal Communications Systems* (PCS); see Section 3.15.

Because of the great interest in WLL technology for the above reasons, the ETSI working party *Radio Equipment and Systems 3* (ETSI RES 3), founded in January 1993, published a report [1] in November 1994 addressing the marketing aspects, different WLL technologies as well as scenarios and characteristics that are of interest from the standpoint of the dimensioning of WLL systems, such as radio range and capacity aspects. The reference model presented in Figure 10.2 originates from this report.

The reference model shows the interfaces of a WLL system and its elements. In addition to antennas, the base station (BS) has facilities for measuring and controlling radio connections with the radio termination of the customer terminal. The controller connects the base station to the local exchange (LE) and controls the base station.

10.1 Technologies for WLL Systems

The following technologies are suitable for WLL systems [2]:

- Analogue cellular mobile radio

- GSM/DCS1800 derivations
- CDMA systems based on US-TIA standard IS-95
- Digital cordless radio networks:
  - DECT
  - *Personal Handyphone System*, PHS (Japan)
  - US *Personal Access Communication System* (PACS)
- Digital *point-to-multipoint* (PMP) radio relay systems

Directional antennas with, e.g., 5–12 dB gain are normally provided for stationary residential lines to enable a low bit-error ratio on the radio channel to be achieved even when base stations have a large coverage radius (typically 2–5 km). However, discussions are also taking place about systems in which the stationary customer line can get by with an indoor antenna, which then has to be small and unobtrusive. The base station then of course has to operate with smaller supply radii.

The suitability of the systems mentioned above depends heavily on the types of user—in other words on the services that are to be offered by the WLL system. A differentiation is made between the following types of user:

- Residential users who require an analogue (PSTN) connection or an ISDN basic rate interface (BRI).
- Small-business customers who operate a small branch exchange for which they require a number of ISDN BRIs (each with $n \cdot 144$ kbit/s) but for whom an ISDN primary rate interface (PRI) would be impractical and too expensive.
- Large-business customers who operate large branch exchanges and require a gateway to data networks such as X.21 and X.25 and frame relay along with normal telephone applications. Their connection capacity needs are on the order of one or more ISDN PRIs ($n \cdot 2048$ kbit/s, $n = 1, 2, \ldots$).

10.1.1 Cellular Mobile Radio Networks

The Ericsson RAS 1000 System, which is based on the NMT standard (*Nordic Mobile Telephony*), was used by Deutsche Telekom AG (DTAG) in 1993 to connect around 13 000 customers in Potsdam, Germany. However, analogue cellular systems are no longer competitive compared with the other systems presented here because of the high cost per connection and the disadvantages compared with digital networks.

Experiments have also been made with digital mobile radio networks. Studies have been conducted with systems such as GSM900/1800 to determine the

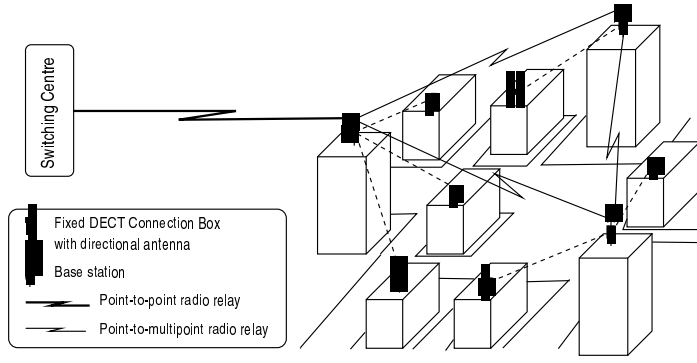


Figure 10.3: Customer coverage by over the roofs supply

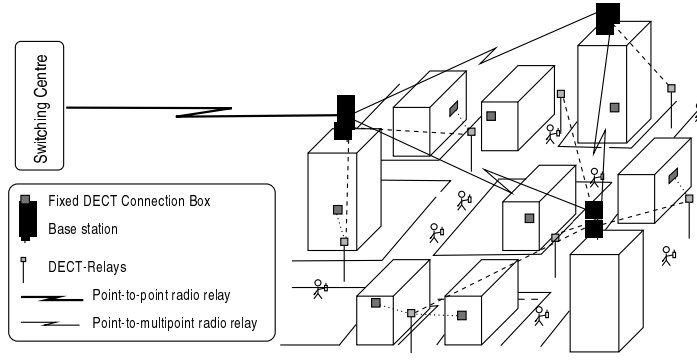


Figure 10.4: Customer coverage by below the roofs supply

feasibility of the possibility shown in the lower part of Figure 10.1. This would allow customers to communicate cost-effectively indoors using their home base stations and to use the same terminal to access the corresponding cellular network. However, mobile radio networks only offer narrowband channels and therefore cannot achieve the quality of service of a PSTN or ISDN/BRI access for wireline users. Nevertheless, all users who just want to place a call have access to the wireless connection to the mobile radio networks.

10.1.2 Digital Cordless Radio Networks

Some of the new network operators are pinning their hopes on WLL systems based on the DECT standard or on CT2 or PHS. At around 300–400 US$, the cost per connection is considerably less than with GSM-based solutions. DECT is able to support coverage *over the roofs* as well as *below the roofs* (see Figures 10.3 and 10.4). *Over the roofs* coverage would not be able to reach ditches, but with low-gain receiving antennas could penetrate through house walls. *Below the roofs* systems operate with base stations (*Radio Fixed Parts*,

Table 10.1: Frequency bands for WLL systems in Europe

Frequency band [GHz]			Applications	Channel grid [MHz]
24.549	...	26.061	PMP radio relay	3.5, 7, 14
17			Radio relay for Hiperlan/4	
3.41	...	3.60	PMP radio relay	3.5
2.5	...	2.67	PMP radio relay	
1.88	...	1.9	DECT-WLL	1.7

RFP) that are installed under the roof, thereby illuminating the ditches and enabling full mobility of the PCS system both indoors and outdoors.

10.1.3 Digital PMP Systems

Point-to-multipoint radio relay systems offer attractive connection options particularly for business customers, because they can provide the customer with channels with high transmission rates. The frequencies for PMP-WLL applications are listed in Table 10.1.

WLL systems based on PMP technology are offered by a large number of suppliers, and represent an interesting connection option in urban and rural areas because of their ability to bridge up to 20 km in borderline cases. Because the transmission capacity needs of stationary customers connected over PMP systems vary, there exist PMP radio relay systems that can dynamically change their (switched) channels in terms of capacity and allocate them to the areas where they are most urgently needed (see Figure 10.5).

PMP systems are frequently also implemented as multihop systems, as shown in Figure 10.6. Several radio links (*Line-of-Sight Radio*, LOS) belonging to point-to-point or point-to-multipoint (PMP) systems are arranged sequentially in order to bridge the path between the fixed network access (*Point of Presence*, POP) and the customer's connection. Another interesting option offered by a number of manufacturers is a combination of PMP systems with DECT systems connected to a PMP end terminal for the connection of residential customers.

10.2 Different WLL Scenarios

The competitive scenarios examined here are based on standard scenarios from ETSI RES-3 [1]. The following scenarios were defined:

1. An existing operator, new area to be developed.
2. Replacement of copper lines with WLL.
3. Reaching the capacity limits of an existing fixed network.

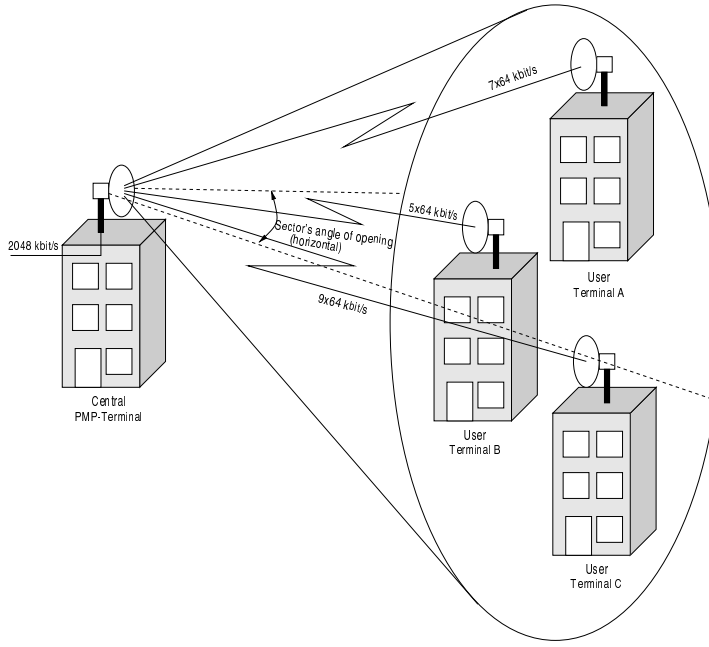


Figure 10.5: PMP radio system with a fixed or dynamic channel allocation controlled by the central terminal

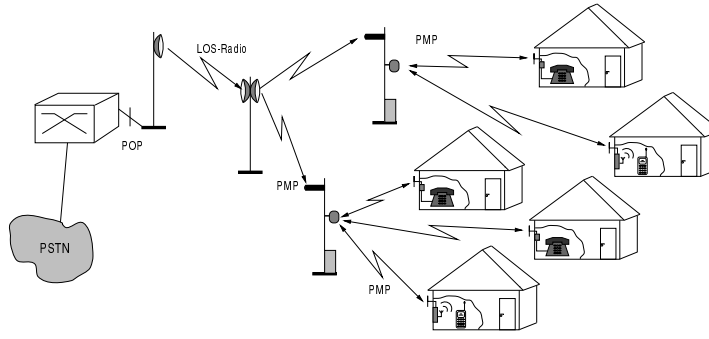


Figure 10.6: Multihop connection-oriented PMP system

4. New operator in competition with existing operator.

ETSI scenario 4 is the one being considered as a model for capacity studies (see Figure 10.7). The assumption is that a provider of telecommunications services is in competition with the national telephone company, and perhaps with other operators. Of the three areas defined in the model scenario, only areas A and B will be considered (see the assumptions in Table 10.2). It

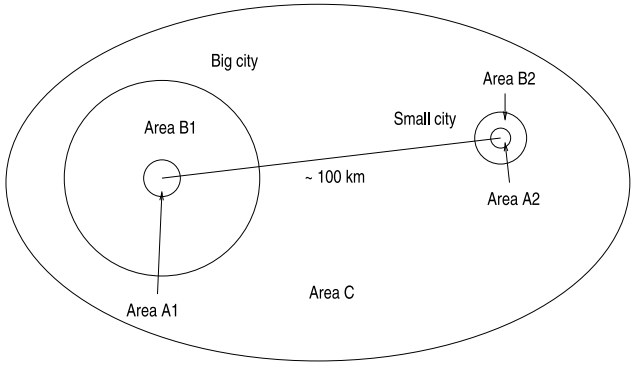


Figure 10.7: ETSI scenario 4

Table 10.2: Definition of areas of ETSI scenario

	Big city	Small city
Number of connections	500 000	50 000
Density of inner area	2000/km^2	1000/km^2
Density of outer area	500/km^2	500/km^2
Radius of inner area	4.5 km	2 km
Radius of outer area	16 km	5 km
Average traffic	70 mErl./user	70 mErl./user
Penetration	1 % increase of users p. a. during the first 10 years	

is assumed that it would not be viable to supply coverage to the sparsely populated area C.

On the assumption of a certain penetration of the respective area, e.g., connection of x % of the users of the WLL system, and with the knowledge of the capacity (in Erl./km^2) of the WLL system under discussion, the calculations in the model show that it is possible to implement voice telephony with any of the WLL technologies mentioned above. However, the initial investment required in developing a connection area is high, and operation of the service especially during the early phases is not economical because of the time involved in gradually enticing customers away from the fixed network operator. In the long run, all WLL technologies would appear to be cost-effective and promising. If the legal conditions allow a competitor direct access to individual subscribers via the wired local loop network by the so-called unbundled access, and if the costs for this use are low (as is the case in Germany at 2.3 Pfg./min), then this will have a considerable effect on the competitiveness of radio-based WLL systems, because they might appear to be too costly.

The most suitable systems for introduction are those that can manage without much infrastructure—in other words, those that can operate in frequency ranges with high diffraction (preferably below 1 GHz) and are therefore able

to illuminate a coverage area with small technical effort. A prerequisite is sufficiently high transmitter power of 2–8 W, because otherwise too many base stations would be required. The more users there are, the more base stations are needed, and the transmitter power decreases—as experienced with cellular systems.

10.3 Direct User Connection in Access Network

In deregulated markets the onus is typically on the previous monopoly owner of the wireline local loop network (*incumbent*) to provide competitors with unbundled user access over their fixed networks. In so doing, the incumbent must allow the competitor, against an appropriate reimbursement of costs, to have direct access to the two-wire lines in the local loop. Non-unbundled access, on the other hand, means that competitors must use (and pay for) the incumbent's multiplex systems every time they want to reach one of their subscribers.

Owing to the changeover from analogue to digital technology (ISDN), old-established telecom companies (*incumbents*) actually have sufficient space in their main distribution frames to allow them to provide direct access within the corresponding rooms. It is not currently clear to what extent new network operators will be taking advantage of direct user access.

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11
Personal Handyphone System (PHS)*

11.1 Development of the Personal Handyphone System in Japan

At the end of the 1980s, two cordless telephone systems, CT2 and DECT, had entered the second step of their development process. At that time, Japan had not yet developed any comparable technology; therefore work began in 1989 on a Japanese cordless standard, which has become known as the Personal Handyphone System (PHS). Disadvantages of conventional cellular telephone systems, such as high costs of infrastructure and cell-planning resulting in high communication fees, had motivated the development of a less expensive system.

The review by the *Telecommunications Technology Council*, a consulting organization for the Japanese Ministry of Posts and Telecommunications (MPT), and the technical study by the *Research and Development Centre for Radio Systems* (RCR) of PHS started at the beginning of 1991 in Japan. The PHS air interface was then standardized through the publication of the RCR STD-28 [1], Version 1, in December 1993. Various field trials in the Sapporo area in October 1993, and in the Tokyo area in April 1994, were then conducted to prove the feasibility of PHS for various demands and services, respectively.

The PHS service was commercially launched in Japan by three operator groups—(NTT (Nippon Telegraph and Telephone) Personal Group, DDI (Daini Denden Inc.) Pocket Telephone Group and the Astel Group)—in July 1995. The technology gained unprecedent popularity, with the number of subscribers reaching the six million mark by the beginning of 1997—just two years after its introduction. As of August 1998, the number of subscribers was about 6.4 million [4]. Attracted by the success in Japan, other Asian countries, including Thailand and Hong Kong, and some South American countries announced plans to establish PHS networks.

*With the collaboration of Matthias Siebert

Table 11.1: Parameters of the Personal Handyphone System

Frequency band	1893.5–1919.6 MHz
Access method	TDMA/TDD
Channel assign. method	DCA (with restrictions for control channels)
Number of RF carriers	87 (incl. 6 control and 4 guard channels)
Carrier spacing	300 kHz
Frame duration	5 ms
Number of slots/frame	8
Modulation	$\pi/4$ DQPSK
Output power (average)	CS: 500 mW or less
	PS: 10 mW or less
Traffic channels/transceiver	3 (resp. 4)
Transmission rate/carrier	384 kbit/s
Net bit rate/channel	32 kbit/s user data
	6.8 or 12.4 kbit/s signalling information

11.2 System Overview

The aims of PHS span those of cordless and cellular systems, encompassing the idea of a low-cost wireless handset that can be used in both indoor and outdoor environments to access fixed network supported services. Like cordless systems (see Chapter 8), PHS provides access to private communication systems for greater flexibility in the office environment or at home. It also forms the basis for public network access by subscribers moving with pedestrian speed. A microcellular structure has been adopted; thus the radio transmit power can be much smaller than that of existing cellular telephone systems. As described for DECT, by means of dynamic channel allocation (DCA), cell station engineering is simplified and a multi-operator environment in the same service area is facilitated. However, some restrictions concerning the fixed control frequencies still require a certain amount of frequency planning in advance.

Various services are supported. Apart from telephone services, the PHS air interface supports voiceband data and facsimile communications at rates up to 9.6 kbit/s. Also, there is the capability of data transmission at rates of 32/64 kbit/s. According to [5] the upper limit of data transmission has been raised to 128 kbit/s, using four channels simultaneously. An overview of the parameters of the Personal Handyphone System is given in Table 11.1.

Like CT2 (see Section 8.1), PHS defines only two network elements: terminal and base station. The nomenclature for a base station is *cell station* (CS) and a PHS terminal is referred to as a *personal station*(PS).

11.2.1 Personal Station (PS)

Similarly to DECT (see Chapter 9), personal stations transmit with an average (peak) RF power of 10 mW (80 mW). As a special feature, comparable to the TETRA system (see Section 6.3), PHS also specifies direct radio communication between a pair of terminals. The direct mode of communication does not need the intervention of cell stations (CSs). As shown in Figure 11.4, ten frequency carriers are designated for such a use. However, since it is only a supplementary service that should not restrict the ordinary use of these frequencies, the direct communication must end within a time limit of three minutes.

As PHS is a system for private and public use, PSs support two modes of operation, namely public and private operation modes. The public operation mode enables the PS to access the public PHS service areas. The private operation mode enables a PS to access private systems like a wireless PBX or the home digital cordless system. The same PS can be used at home and in the office by selecting a relevant private mode within the pre-registered private systems.

Terminal mobility allows the PS a smooth transition between public and private PHS services by selecting the appropriate mode on the PS. This selection can also be done automatically (*Automatic Selection Mode*). When the PS is within the range of both services, it will operate in a predetermined mode. Optionally, an automatic dual-mode operation is possible, where the PS can be paged from both the public system and the private system when it is within the range of both.

11.2.2 Cell Station (CS)

A cell station consists of CS equipment and antennas. These two parts together create a microcell service area with a radius of several hundred metres. Cell stations control most of the tasks concerning the air interface. These include *dynamic channel allocation* (DCA), *superframe establishment*, *diversity support*, and many more. What is unique is the use of diversity, both for the uplink (by means of post-detection selection) and for the downlink (by means of transmitter antenna selection), controlled by the CS. The average (peak) RF power depends on the kind of CS. An outdoor standard-power CS, for example, transmits at an RF power of 20 mW (160 mW).

11.3 PHS Radio Characteristics

11.3.1 Speech Coding

Similarly to DECT, a 64 kbit/s full-rate voice coding is employed first; then the signal is transcoded into 32 kbit/s ADPCM (*Adaptive Differential Pulse Code Modulation*) based on the ITU-T Recommendation G.726.

Half-rate and quarter-rate voice coding methods are not specified at present, but a superframe configuration allows multiple codecs with lower rates, down to 8 kbit/s, to be incorporated in the system when they become available.

The ADPCM compresses speech data without degrading speech quality. Performance of voiceband data transmission via modems is not significantly degraded, and non-speech service can be supported.

11.3.2 Modulation

PHS employs a $\pi/4$-shifted DQPSK modulation with a roll-off factor of 0.5. This modulation scheme permits a variety of demodulation techniques to be used, such as delay detection, coherent detection and frequency discrimination detection. Furthermore, the use of the DQPSK modulation method enables a higher spectrum utilization efficiency compared with GMSK modulation.

11.3.3 Access Method

In common with DECT, the access method applied in PHS is hybrid time-division/frequency-division multiple-access (TDMA/FDMA) with time-division duplexing (TDD). The use of TDD also makes it possible to modify service bit rates. The latest ARIB Standard, RCR STD-28 Version 3, issued in November 1997, lists the specifications for PHS 64 kbit/s digital data transmission. It enables high-speed wireless access to the ISDN network from the PS by assigning a second TCH in parallel.

A TDMA frame has a length of 5 ms and carries 8 slots. The first four slots are downlink; the other four slots are uplink slots (see Table 11.1 and Figure 11.1).

The TDMA/TDD technology allows deviation from the allocation of paired transmit/receive channels, which is usually required in order to accomplish symmetric two-way communication and is able to support asymmetric communication relationships. TDMA/TDD is flexible, because it does not need paired bands and both the lower and upper ends of the spectrum can easily be expanded to accommodate needs as in RCR STD-28 Version 3.

11.3.3.1 Communication Physical Slot Designation Method

Designation of communication physical slots is performed by a signal (*link channel assignment* message) on a Signalling Control Channel (SCCH) (see Section 11.4.1.3), sent from CS. The slot designation position is indicated by the slot number, counted relatively to the first slot starting 2.5 ms after the signal (link channel assignment message) has been received by the PS.

An example is given in Figure 11.1: here the link channel assignment message is transmitted in the first TDMA slot. The PS waits 2.5 ms after the reception, and then starts to count the following slots until the relative value maps with that given in the message. As a bidirectional channel is always

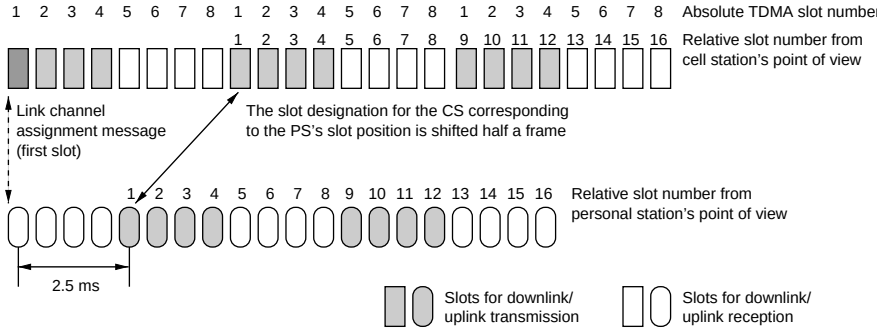


Figure 11.1: Example of relative slot numbers

made up of a pair of slots, the second slot is defined to follow half a frame length (2.5 ms) later, corresponding to the TDD scheme.

Thus the physical time slot number is specified by a combination of absolute and relative slot number by the CS. The TDMA slot number (SN) of a communication carrier is obtained from the following equation:

$$\text{TDMA SN} = \{(\text{absolute SN} + \text{relative SN} - 2)\text{mod}4\} + 1 \qquad (11.1)$$

11.3.3.2 PHS Superframe Structure

The minimum cycle of the *downlink* logical control channel (LCCH) that specifies the slot position of the first repeated LCCH elements is specified as the LCCH superframe. All transmission/reception timing of physical slots for controlling intermittent transmission and SCCH uplink slot designation is generated based on the superframe structure.

Elements (subchannels) of the downlink LCCH (see Section 11.4.1 for a detailed description) are the Broadcast Control Channel (BCCH), Paging Channels (PCH, $P_1 - P_k$: number of paging groups = k), the Signalling Control Channel (SCCH) and an optional User-Specific Control Channel (USCCH).

The BCCH must be transmitted in the first slot of the LCCH superframe whereby the lead position of the superframe is reported. In detail, this is done by means of profile data contained in the *radio channel information broadcasting* message (see Section 11.5.5.1). If necessary, it is possible to temporarily steal LCCH elements, except for BCCH, and send some other LCCH elements.

The downlink logical control channel (LCCH) has the superframe structure shown in Figure 11.2. After each n TDMA frames, the CS intermittently transmits an LCCH slot (as discussed in Section 11.4.1, the CS does not transmit control signals in each TDMA frame). The parameter m defines the number of LCCH elements that have to be transmitted until all kinds of information have been conveyed once.

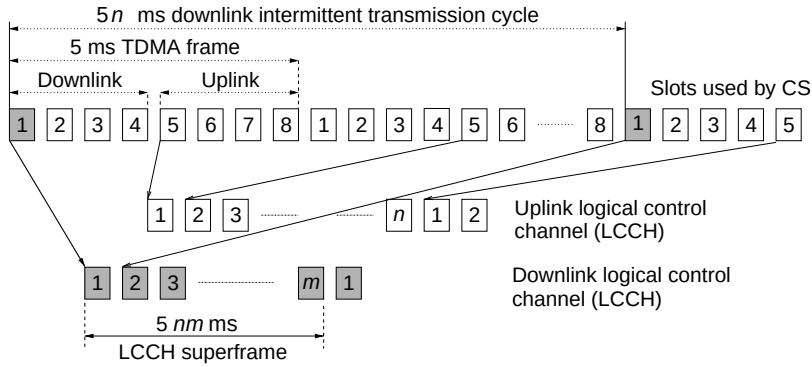


Figure 11.2: PHS superframe

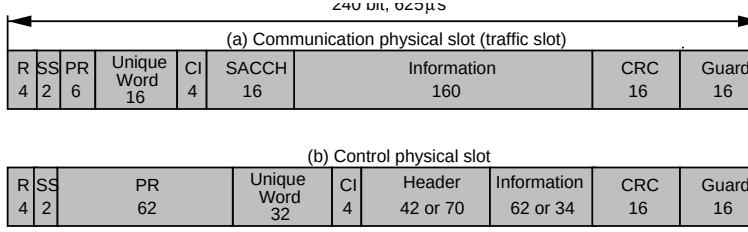


Figure 11.3: PHS time-slot formats

For the uplink, no superframe structure is defined. Personal stations transmit their first signalling message by using the *Slotted Aloha* protocol (see Section 2.8.1), using an uplink SCCH slot, if they want a connection to be established.

11.3.4 Slot Structure

As a frame in PHS lasts 5 ms and each frame consists of 8 slots, each slot has a length of 625 µs in which 240 bits can be arranged. PHS specifies several time-slot formats corresponding to different logical channels. Basically, there are two categories: *control physical slots* used by *common control channels* (CCHs), and *communication physical slots* used by *traffic channels* (TCHs); see Figure 11.3.

In contrast to the DECT standard, where, as a concession to more cost-effective hardware, certain slots (and related traffic channels) might not be used (see Figure 9.50), PHS makes use of all its time slots. Therefore all slot formats start with a 4-bit ramp time in which the PS or CS turns on its transmitter and a 2-bit start symbol for establishing the phase of the remote demodulator. At the end of each slot format, there is another common data field for performing an ITU-T 16-bit CRC. The start symbol is followed

by a *preamble*, a layer-1 signal pattern used to establish bit synchronization. Its number of bits depends on the kind of slot format. Together, start symbol and preamble are repetitions of the pattern 1001. With control physical slots, a preamble of 62 bits is used to allow synchronization for each slot independently; with communication physical slots, a preamble of 6 bits serves to update the *synchronization* established in the previously transmitted slots.

Additionally, all time slots carry a *unique word*, known in advance by the receiver, which is different for the downlink and uplink channels. It also differs for control physical slots and communication physical slots, so that it helps to distinguish between them. Moreover, together with the CRC check, it is used for error detection. As it is known in advance, the receiving party listening for its dedicated slot either detects it or not (unique word detection error).

The *channel identifier* follows right after the unique word. It is similar to the flag bits in GSM bursts. For example, the channel identifier sequence 0000 indicates that the time slot carries user information (TCH), while 0001 indicates that the time slot carries a fast associated control channel (FACCH). Slow associated control channels (SACCH) do not have a special channel identifier since they are part of each communication physical slot, with the exception of an optional user-specific packet channel (USPCH), described in Section 11.4.2.4.

The most important data field in Figure 11.3 (a) is the *information field*, which carries the user data. For telephone services such as voice transmission it consists of 160 bits from an adaptive differential pulse code modulation (*ADPCM*) encoder. Therefore, in common with DECT, the bit rate carried by communication physical slots (with the exception of USPCH) is

$$\frac{160 \text{ bit/frame}}{0.005 \text{ s/frame}} = 32 \text{ kbit/s} \tag{11.2}$$

Control physical slots have headers containing addresses. Broadcasting point-to-multipoint channels such as the *broadcasting control channel* (BCCH) and the *paging channel* (PCH) only need to transmit a 42-bit CS identification code. Thus the information field contains 62 bits. On the other hand, bidirectional point-to-point channels such as the *signalling control channel* (SCCH) need to specify a CS identification code (42 bits) and a PS identification code (28 bits). In these channels the length of the information field is 34 bit. This implies that the information rate for logical control channels is either

$$\frac{62 \text{ bit/frame}}{0.005 \text{ s/frame}} = 12.4 \text{ kbit/s} \quad \text{or} \quad \frac{34 \text{ bit/frame}}{0.005 \text{ s/frame}} = 6.8 \text{ kbit/s} \tag{11.3}$$

Finally, all slot formats have a *guard time* of 41.7 µs, corresponding to 16 bits, i.e., a burst carried in a slot is 224 bits in length.

11.3.5 Radio-Frequency Band

The PHS band spans 26.1 MHz from 1893.5–1919.6 MHz. Originally the radio-frequency band allocated for PHS service spanned 23.1 MHz in the range of 1.895–1918.1 MHz. Because it was working close to capacity in hot-spot areas, an extension became necessary. There are 87 carriers in the PHS band, with a spacing of 300 kHz. Figure 11.4 shows the relationship between the PHS frequency band and the carrier numbers. Control carriers for private use are assigned to 1898.450 MHz and 1900.250 MHz for Japan, and 1903.850 MHz and 1905.650 MHz for other countries. In Japan, four control carriers are reserved for public use. One control channel is assigned to each of the three PHS operators, and the remaining one is set aside as a spare channel. In order to protect the public control channels from adjacent channel interferences, they are enclosed by guard channels. Currently the spectrum for public use is 15 MHz. The spectrum for private use is 11.1 MHz, which can be shared with public use. In addition, the first 10 carriers of private use (1895.15–1897.85 MHz) are also designated for direct communication between personal stations (Transceiver mode, Walkie Talkie).

11.3.6 Frequency Allocation

As Figure 11.4 shows, the PHS frequency band is not separately allocated for each operator, but is shared, with the exception of the channels that are pre-assigned as control channels dynamically by all PHS operators by means of *dynamic channel assignment* (DCA). This is an autonomous decentralized radio channel control technology, which enables efficient and flexible use of frequencies and time slots according to the local interference levels of CS and PS.

In contrast to the DECT system, where it is the mobile's task to select a suitable channel, in PHS this is done by the cell station. With a *link channel establishment request* or a *TCH switching channel request* (see Section 11.5.5.1), the PS asks for the assignment of a channel. The cell station can automatically pick up carriers at random and select an available carrier that has no interference problem. As a result of checking signal strength when a call is established, the CS renews an internal two-dimensional frequency–time matrix with available channels.

If no carrier is available, the CS refuses the request. The PS will then automatically request again. This can be done up to three times; then the PS has to wait a certain time before another try is possible.

11.3.7 Microcellular Architecture

PHS applies a microcellular architecture, which permits efficient spectrum utilization and the use of low-power handsets. Thus long standby/talk times can be realized. In cooperation with PHS, large numbers of cell stations can be deployed without planning. (In fact, this is only true for traffic channels.

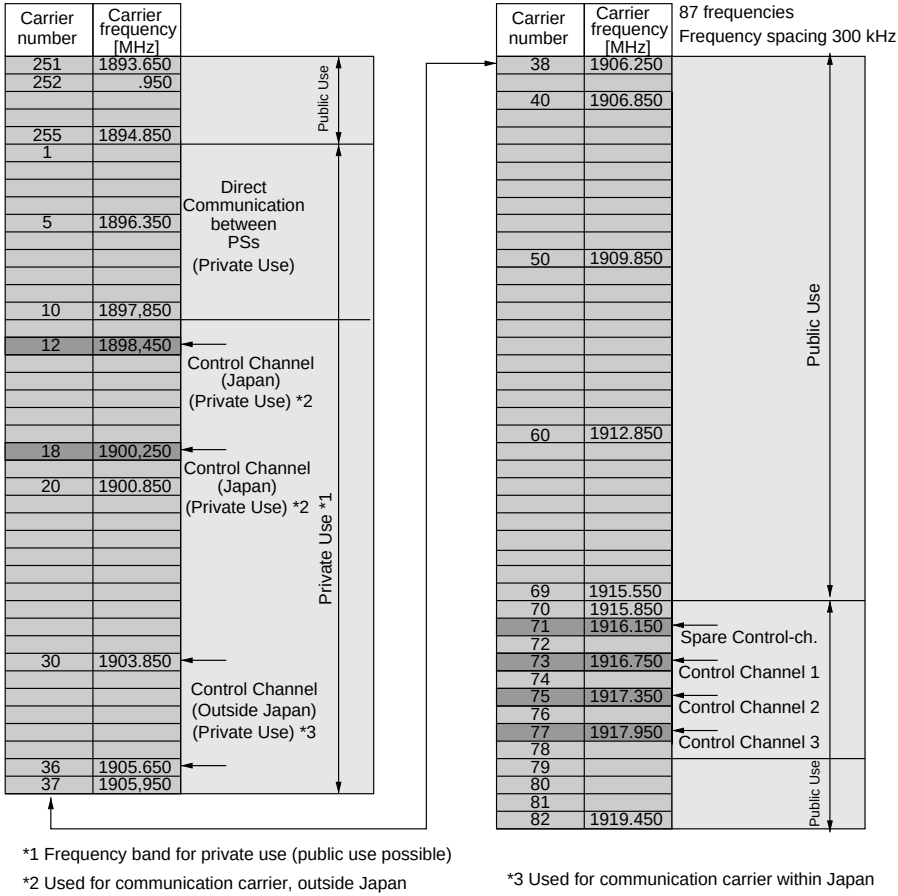


Figure 11.4: PHS frequency table

Concerning the fixed control channels, frequency planning still has to be done). For example, NTT launched a PHS service in July 1995 with some 25 000 standard cell stations in Tokyo, using a conventional small-cell structure, 150–200 m, with antennas mounted on public telephone boxes.

The microcellular technology allows easy addition and removal of CSs in the field, based on factors such as traffic demand and the presence of other operators serving the same area. Cell stations with 10 mW of average power are deployed at about 200 m spacing in downtown areas. Wider separation between cell stations is possible using CSs with higher-power transmitters and extremely low-noise amplifiers.

11.3.8 Handover

In PHS, a handover is called *channel switching*. During a call, both network elements—the cell station and the personal station—observe the channel quality. This is done by evaluating the *radio signal strength indicator* (RSSI) and the *frame-error ratio*. The latter is indicated by detected errors in the cyclic redundancy check (CRC) in every time slot, and also unique word errors if there is a call in progress. In response to deteriorating quality, either network element can initiate a handover.

PHS distinguishes between two types of handover: one is called *recalling-type* and the other *traffic channel switching-type*.

TCH switching-type If a high frame-error ratio is detected but the received signal strength indicates that the terminal is still in the vicinity of the serving cell station, this is mostly due to rising interference. Remedial measures can be taken by switching to another slot or frequency; thus the serving cell station does not have to be changed. To do so, the terminal transmits a *TCH switching request* message on the fast associated control channel (FACCH). The cell station responds with a *TCH switching indication* message, directing the PS to a new physical channel. Then the two stations exchange synchronization bursts on the new channel and, if successful, resume communication. In other systems this function is called intracell handover.

Recalling-type Here, change of the communication channel is done in such a manner that the handset establishes a connection in the same way as a call originating in a new communication cell. This function allows the call to be maintained whilst the traffic channel is switched between cell stations or interfaces as the personal station moves during the call. The stimulus for this kind of handover had been that the received radio signal strength became too weak to maintain communications with the serving cell station. Measuring signal strengths on common control channels received from surrounding cells, the PS picks up a new CS. Therefore it transmits a *link channel establishment request* message to the chosen cell station on a signalling control channel (SCCH). In answer to this, the CS responds with a *link channel assignment* message, also transmitted on SCCH, and both stations start synchronizing their operations on the new channel.

To perform this kind of intercell handover, both PS and CS need access to a signalling control channel. For the uplink, a *Slotted-ALOHA* protocol is applied, and the downlink's access is restricted by the superframe structure. As a result, seamless handovers cannot take place owing to a short time gap.

According to [3], intercell handovers are performed to support connections at motorcar speed. However, this is not guaranteed by the providers. If a new cell station is not available when the handover is

required, the communication channel is switched back to the previous CS. Therefore the communication channel is restored when returning quickly to the previous cell.

11.4 PHS Radio Channel Structures

The Personal Handyphone System supports traffic channels (TCHs) and control channels (CCHs), as shown in Figure 11.5.

11.4.1 Logical Control Channels (LCCH)

The basic structure for all kind of control slots, with exception of the associated control channel (ACCH), is a control slot as shown in Figure 11.6.

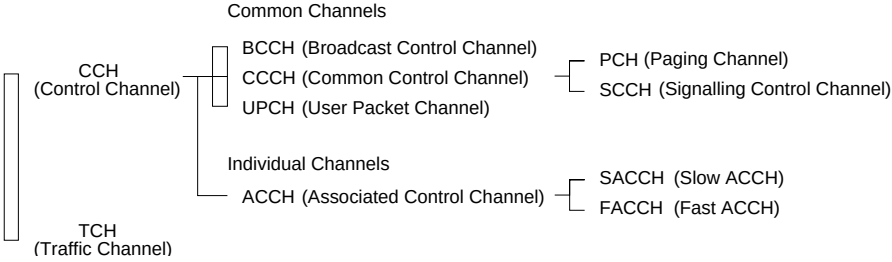


Figure 11.5: Function channel structure

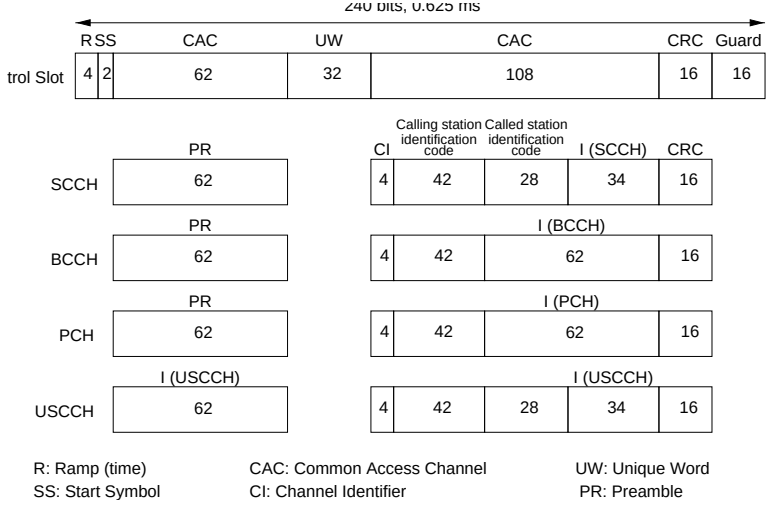


Figure 11.6: Control physical slot structures

11.4.1.1 Broadcast Control Channel (BCCH)

This is a one-way downlink channel for broadcasting control information from the cell station to the personal station. It transmits information related to channel structure, system information and restrictions. The BCCH carries general system information to all terminals in a cell. Cell stations establish a superframe structure (see Figure 11.2) and inform personal stations by means of messages transmitted on the BCCH (see Section 11.5.5.1). The transmission of these paging messages is scheduled in a manner that enables terminals to turn off their receivers for a high proportion of the time. Thus, corresponding to the sleep mode of digital cellular systems, PHS defines a *battery saving* mode of operation for PSs that do not have a call in progress.

11.4.1.2 Paging Channel (PCH)

This is a one-way downlink point-to-multipoint channel that simultaneously transmits identical information to individual cells or a wide area of multiple cells (the paging area) from the CS to PS (for more details see Section 11.5.5.1).

11.4.1.3 Signalling Control Channel (SCCH)

This is a bidirectional point-to-point channel that transmits information needed for call connection between CS and PS by means of messages (for detailed information see Section 11.5.5.1). It transmits independent information to each cell. Personal stations contend for access to the reverse direction randomly, by means of the *Slotted Aloha* protocol.

11.4.1.4 User-Specific Control Channel (USCCH)

A User Packet Channel (UPCH) that can be set up on the control physical slot is defined as the USCCH (see Figure 11.6). If it satisfies the specified items, the usage method is arbitrary. This two-way channel for packet data communication of user information can be used for transmitting short messages to and from personal stations that do not have calls in progress. The uplink channel is random-access.

11.4.2 Service Channels

The basic structure of all kinds of service channels is a traffic slot as shown in Figure 11.7.

11.4.2.1 Synchronization Burst

The synchronization burst is used for ensuring frequency and time synchronization when switching channels and when setting up communication physical slots. It includes a 32-bit unique word.

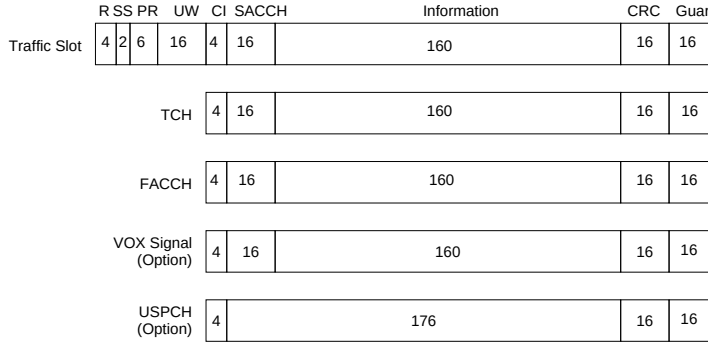


Figure 11.7: Communication physical slot structures

11.4.2.2 Traffic Channel (TCH)

User information is transmitted via the TCH, a point-to-point bidirectional channel.

11.4.2.3 Associated Control Channel (ACCH)

This is a two-way channel that is associated with a TCH. It carries out transmission of control information and user packet data necessary for call control. The channel that is ordinarily auxiliary to TCH is called SACCH (*Slow Associated Control Channel*), and the channel that temporarily steals TCH capacity and carries out high-speed data transmission is called FACCH (*Fast Associated Control Channel*). SACCH and FACCH are mutually independent, and may exist on the same physical slot.

Slow Associated Control Channel (SACCH) The SACCH conveys bidirectional control signals between a cell station and a personal station related to a call in progress.

Fast Associated Control Channel (FACCH) The FACCH can temporarily steal slots from the traffic channel (these are the slots carrying the user information), in order to exchange time-critical control information, e.g., for handover support.

11.4.2.4 User-Specific Packet Channel (USPCH)

The UPCH that can be set up on the communication physical slot is defined as USPCH. If it satisfies the specified items, the usage method is arbitrary. The USPCH uses a service channel assigned to a specific terminal. The timeslot structure of the USPCH (see Figure 11.7) resembles the communication physical slot structure, but with the difference that the USPCH does not contain a SACCH field. Instead, these 16 bits become part of the information field, now holding 176 bits.

11.4.2.5 Voice Activity Channel (VOX)

This is an optional channel that transmits VOX control start and background noise information. It operates in connection with a voice-activity detector that conserves battery power by turning off the PS transmitter when it detects a pause in the speech of the subscriber using the PS. When this happens, the PS transmits a special message in VOX format in one out of every four frames describing the background noise at the PS.

11.5 Network Operations

In a manner similar to DECT and GSM, PHS formally defines three network control protocols:

1. Radio-Frequency Transmission Management (RT)
2. Mobility Management (MM)
3. Call Control (CC)

These layer-3 standards specify procedures for establishing, maintaining, switching, releasing network connections, PS location and authentication at the radio interface of the Personal Handyphone System. These procedures apply to messages exchanged through link channels in the link channel establishment phase and service channels in the other phases. Their aim is to provide the environment, procedures and messages needed for radio control, mobility control and call control in link channels as in service channels.

11.5.1 Radio-Frequency Transmission Management (RT)

RT incorporates radio resources management functions and also signal encryption. The radio frequency transmission entity RT has functions related to management of radio resources. These functions include radio zone selection, radio line setup, maintenance, switching and disconnection functions.

11.5.2 Mobility Management (MM)

The mobility management entity has functions related to the mobility support of a PS. These functions include location registration, authentication functions and handover support.

For each network control protocol (RT, MM, CC), a state transition diagram can be set up. Examples of how this is done are shown in Figure 11.8 and Figure 11.9. Incoming and outgoing messages (the latter are in parentheses), which are described below, stimulate state transitions.

Messages for mobility management (MM), which are transmitted on SACCH/FACCH, are (see Figures 11.8 and 11.9):

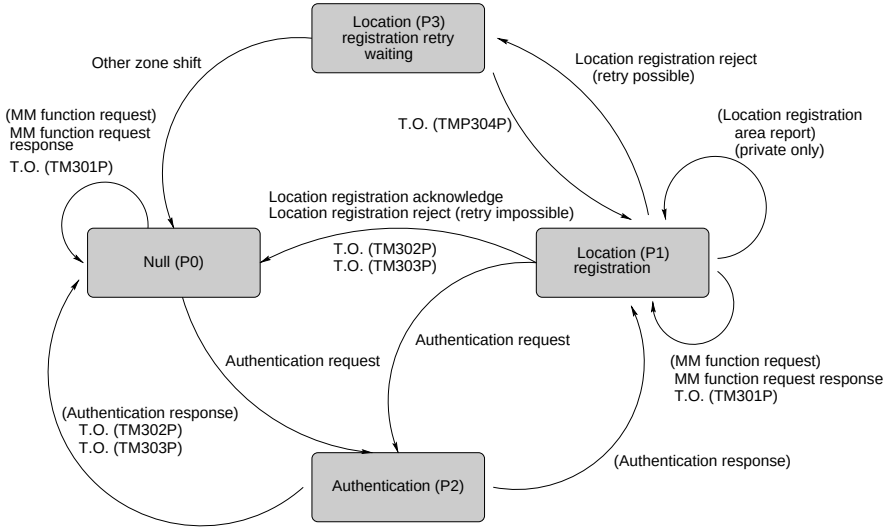


Figure 11.8: MM state transition diagram (PS side)

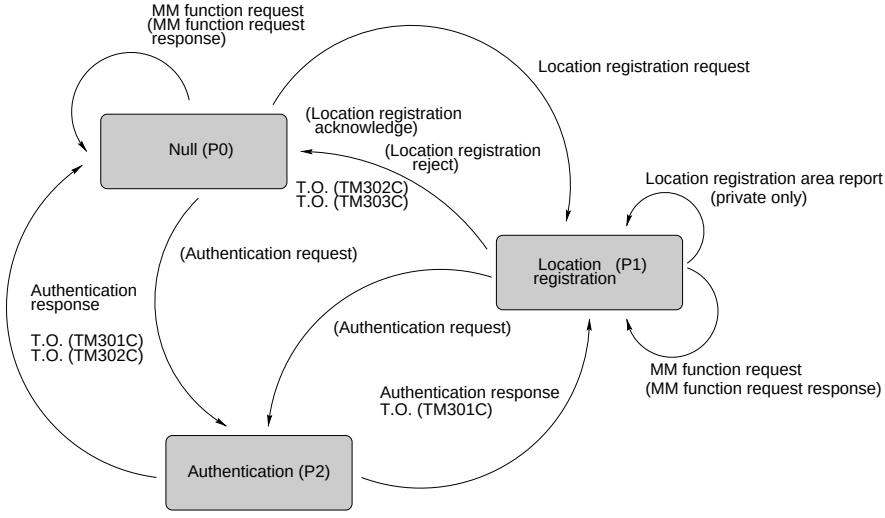


Figure 11.9: MM state transition diagram (CS side)

Authentication Request (CS → PS, DL) This message is for confirming the legitimacy of PS, and it is transmitted from CS to PS.

Authentication Response (PS → CS, UL) This message is for reporting the response to the Authentication Request. It is transmitted from PS to CS.

Function Request (PS → CS, UL) This message is transmitted from PS to CS when PS requests MM functions from CS.

Function Request Response (CS → PS, DL) This message is transmitted from CS to PS for responding to the MM function request from PS.

Location Registration Acknowledge (CS → PS, DL) This message, transmitted from CS to PS, is for reporting that the location registration request was accepted.

Location Registration Area Report (PS → CS, UL) This message is used to transmit from PS to CS the specified CS-ID and reception level from CS in order for PS to specify the paging zone.

Location Registration Request (PS → CS, UL) This messages, transmitted from PS to CS, is for PS requesting a new location.

Location Registration Reject (CS → PS, DL) This message is transmitted from CS to PS in the event of Location Registration Reject.

11.5.3 Call Control (CC)

The call control (CC) entity has functions related to connection call control. These functions include call setup, maintenance and release functions. They are described in the following for calls between PSs.

Messages for call control, transmitted on SACCH/FACCH, are described in the following subsections.

11.5.3.1 Call Establishment Messages

ALERTing (CS ↔ PS, DL, UL) This message is sent from the PS to CS to indicate the fact that the PS is in the process of calling. The CS sends this message to report to the PS that a call is initiated on the destination side.

CALL PROCeeding (CS ↔ PS, DL, UL) This message is to describe that the requested setup is started. It is transferred from the destination PS to CS or from CS to the originating PS. It also shows that setup information cannot be received beyond this point.

CONNect (CS ↔ PS, DL, UL) This message is sent from the destination PS to the CS, or from the CS to the originating PS, to report the fact that the destination PS received a call.

CONNect ACKnowledge (CS ↔ PS, DL, UL) This message is transmitted from the CS to the destination PS to report that the call is established. Also, for symmetry of protocol procedures, it may be transmitted from the originating PS to CS.

PROGress (CS ↔ PS, DL, UL) This message is sent from CS or PS to indicate the progress of a call in cases where interconnection or inband information/pattern is provided.

SETUP (CS ↔ PS, DL, UL) This message is transferred in order to start call setup from the origination PS to CS or from CS to the destination PS.

SETUP ACKnowledge (CS → PS, DL) This optional message is sent from CS to the origination PS to describe that call setup is initiated and to request further information. It can only be used in private systems.

11.5.3.2 Call Clearing Messages

DISConnect (CS ↔ PS, DL, UL) This message is transmitted from PS to request call clearing from CS, or from CS in order to describe that the call was disconnected.

RELease (CS ↔ PS, DL, UL) This message is transmitted in one direction from either PS or CS. It shows that the equipment that transmitted this message has already disconnected the traffic channel. Further it is transmitted to release the traffic channel and call reference.

RELease COMPlete (CS ↔ PS, DL, UL) This message is transmitted in one direction from PS or CS. It shows that the equipment that transmitted this message has already released the traffic channel and call reference.

11.5.3.3 Other Messages

FACility (CS ↔ PS, DL, UL) This message is sent in order to request or check supplementary services.

INFOrmation (CS ↔ PS, DL, UL) This optional message is used when it is desired to transmit information from PS or CS.

NOTIFY (CS → PS, DL) This message is sent from CS to PS to indicate information pertaining to a call, such as user suspended.

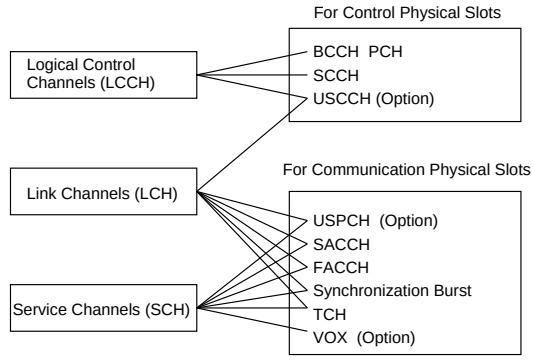


Figure 11.10: Correspondence between protocol phase channels and function channels

STATus (CS ↔ PS, DL, UL) This message is transferred from either PS or CS at any time within the call for reporting error status or as a response to STATus ENQuiry message.

STATus ENQuiry (CS ↔ PS, DL, UL) This optional message is transmitted at any time for requesting a status message from layer 3 to PS or CS. If this message is received, transmission of the STATus message is mandatory as a response.

11.5.4 Protocol Model

In general, communication is made up of three phases: call establishment, communication and call release. As shown in Figure 11.11, the *link channel establishment phase* is followed by the *service channel establishment phase*, and finally by the *communication phase*. Figure 11.10 shows what channels are being used as link, service and logical control channels respectively.

The link channel establishment and the service channel establishment phases use the control protocol stack, while the communication phase uses layers 1 to 2 of the user plane conformant to the ISO-OSI model (see Section 2.5). At the end of the communication phase, the control plane (management) is used to disconnect the call; see Figure 11.11.

11.5.5 Call Establishment

The call connection phase consists of the *link channel establishment phase* for establishing the link with the radio interface, and the *service channel establishment phase* for establishing the radio link for telephone service such as voice transmission and non-telephone service such as ISDN.

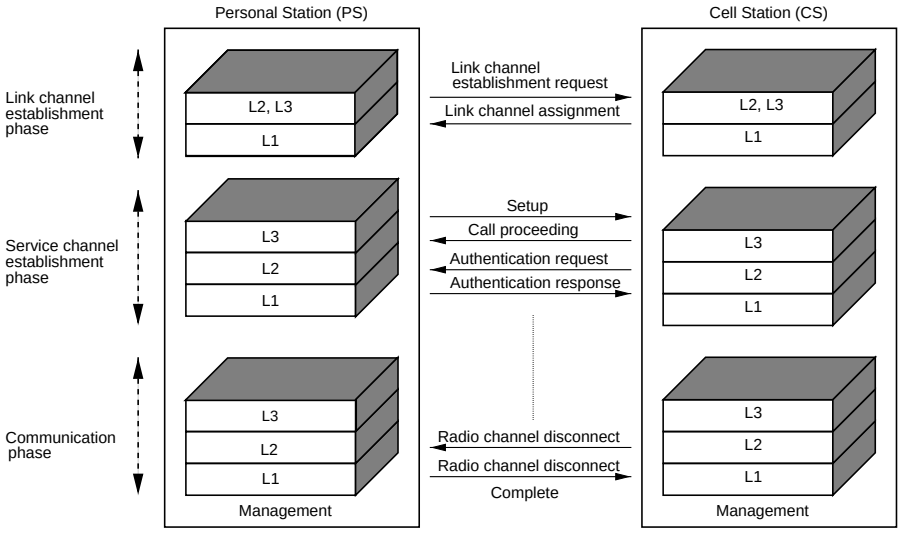


Figure 11.11: Basic structure of signals

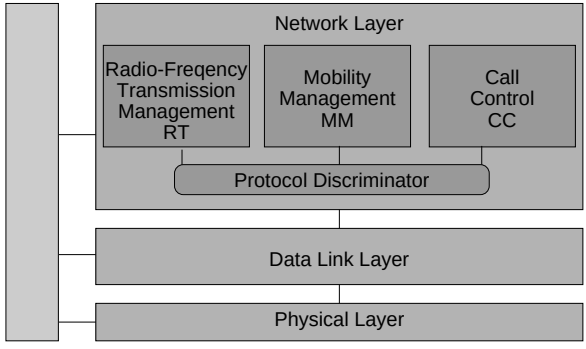


Figure 11.12: Hierarchical structure of service channel establishment phase and communication phase

11.5.5.1 Link Channel Establishment

The link channel establishment phase is defined as the stage of using control channel functions to select a channel (henceforth referred to as *link channel*) with the quality and capacity required for each service's call connection, and to select the protocol type required in the next phase of a call connection. For the purpose of assigning a dedicated channel for communication, personal stations and base stations exchange messages on common control channels, such as BCCH, SCCH and PCH. Figure 11.12 shows the functions of call control involved in this phase.

A possible start of a communication is described below: The PS starts the process by sending a link channel establishment request message using a signalling control channel (SCCH). The stimulus for this request is either a paging message received from the CS, a stimulus to register the location of the PS or a call originated by the mobile subscriber. In contrast to DECT, the CS is responsible for allocating a service channel. After receiving a request from the PS, the CS selects a service channel and transmits the chosen pair of slot and frequency, together with a link channel assignment message, to the personal station, again using the SCCH. The PS then tunes to the assigned channel, and the two stations transmit synchronization bursts to establish communication.

Messages defined in the link channel establishment phase are as follows:

Messages for Channel Setup (Transmitted on SCCH)

Idle (CS → PS, DL) In downlink SCCH timing, only when there is no significant signal to be transmitted.

Link Channel Establishment Request (PS → CS, UL) The PS carries out link set up request to CS using this message. The link set up request is only possible from PS to CS (uplink). PS designates the link channel (LCH) type needed for communication and the call connection protocol type, and transmits the link channel establishment request to CS.

Link Channel Assignment (CS → PS, DL) After there has been a link channel establishment request from PS, CS uses this message to perform channel assignment in response to PS. Furthermore, CS needs to confirm that it is possible to provide the LCH protocol type and LCH type that are requested by PS. The most important pieces of information belonging to this message are the carrier number (see Table 11.4) and the time slot number. The latter is specified by a combination of an absolute and relative slot number (see Section 11.3.3.1).

Link Channel Assignment Reject (CS → PS, DL) CS uses this message, in response to a link channel (re-)request from PS, to inform that channel setup is not possible. The CS can indicate the reject reason in this message, such as that there is no free radio channel, or no free outgoing line on the wired side, or that LCH protocol type or LCH type request cannot be satisfied, etc.

Link Channel Establishment Re-Request (PS → CS, UL) PS can use this message to halt the link channel establishment sequence and re-request CS for link channel establishment after a link channel assignment message has been received from CS.

Messages for Broadcasting (Transmitted on BCCH)

Radio Channel Information Broadcasting (CS → PS, DL) CS must broadcast the radio channel structure to PS using this message. This includes

information about the downlink LCCH superframe structure (see Section 11.2), the battery saving cycle, various paging information and others.

System Information Broadcasting (CS → PS, DL) Using this message, CS can broadcast restriction information such as congestion control to PS. Furthermore, CS can give information about its type, service attributes and other.

2nd-System Information Broadcasting (CS → PS, DL) By using this message, CS can broadcast information such as country code and system type to PS.

3rd-System Information Broadcasting (CS → PS, DL) CS can broadcast 3rd-system information for a public system to PS, using this message.

Option Information Broadcasting (CS → PS, DL) Using this message, CS can broadcast private system option information to PS.

Message for Paging (Transmitted on PCH)

Paging (CS → PS, DL) Using this message, CS reports that PS received a call. When PS responds to the paging from CS, it is necessary to activate the link channel establishment request.

11.5.5.2 Service Channel Establishment

The service channel establishment phase is defined as the stage of using link channel functions obtained in the link channel establishment phase to select a channel (henceforth referred to as the *service channel*) with the capacity required for providing service and to select the protocol type required in the communication phase.

Ordinary call control (CC), higher-level mobility management (MM) and radio-frequency transmission management (RT) functions are performed in this phase. Messages are transmitted by means of an associated control channel (ACCH).

Messages Pertaining to Activation of Communication (Transmitted on SACCH/FACCH)

Definition Information Request (PS → CS, UL) This message is transmitted to CS for requesting notification of definition information.

Definition Information Response (CS → PS, DL) This message is transmitted from CS to PS to report definition information.

Encryption Key Set (PS → CS, UL) This message is used to report the encryption key, and is transmitted from PS to CS as necessary. It is applied

in the service channel establishment phase. If the encryption key was set by this message, the encryption function is effective starting from the first TCH data of the communication phase.

Function Request (PS → CS, UL) This message is transmitted from PS to CS to perform an RT function request to CS.

Function Request Response (CS → PS, DL) This message is transmitted from CS to PS to notify PS of standards and to respond to an RT function request from PS.

Paging Response (PS → CS, UL) This message is transmitted from PS to CS to respond to a paging call from CS.

Zone Information Indication (CS → PS, DL) This message is transmitted from CS to PS to report zone information.

Messages Pertaining to Connection Release (Transmitted on SACCH/FACCH)

PS Release (CS → PS, DL) This message is transmitted from CS to PS in order to unilaterally release the radio channel.

Radio-Channel Disconnect (CS → PS, DL) CS transmits this message to PS to release the radio channel.

Radio-Channel Disconnect Complete (PS → CS, UL) This message from PS to CS is used to indicate the fact that the radio channel was released and as a response to Radio-Channel Disconnect. After PS transmits this message, it enters standby state.

Messages Pertaining to Connection Establishment (Transmitted on SACCH/FACCH)

Condition Inquiry (CS → PS, DL) This message is transmitted from CS to PS for querying about the reception level of the local zone and peripheral zones.

Condition Report (CS ↔ PS, DL, UL) This message is transmitted from CS to PS to autonomously report a condition, and from PS to CS to respond to a condition inquiry.

Encryption Control (CS ↔ PS, DL, UL) This messages is transmitted in either direction to indicate operation or stopping of the encryption function.

Encryption Control Acknowledgement (CS ↔ PS, DL, UL) This message is transmitted in either direction as an acknowledgement of encryption control. Encryption control in the communication phase (start/stop of encryption process) is activated by transmission of this message.

TCH Switching Indication (CS → PS, DL) This message from CS to PS is used to indicate TCH switching. After PS has received this message, it immediately starts the operation of switching to the specified channel.

TCH Switching Request (PS → CS, UL) This message is transmitted from PS to CS to perform a TCH switching request from PS to CS.

TCH Switching Request Reject (CS → PS, DL) This message is transmitted from CS to PS when the TCH Switching Request to CS from PS is rejected.

TCH Switching Re-Request (PS → CS, UL) This message is transmitted from PS to CS to re-request TCH switching from PS to CS. It is used for channel switching during communication.

Transmission Power Control (CS ↔ PS, DL, UL) This message is transmitted in order to specify operation or prohibition of the Transmission Power Control function.

VOX Control (CS ↔ PS, DL, UL) This message is transmitted to indicate operation or prohibition of the VOX function.

PS-ID Notification (CS ↔ PS, DL, UL) This message is transmitted in either direction to report the respective PS identification code (PS-ID) during communication.

11.5.6 Communication Phase

User information such as voice and non-voice data is transmitted in this phase. Layered control signals are transmitted through an associated control channel (ACCH).

When both parties are connected, the call enters the communication phase, in which the principle activity is the exchange of user information. During a communication, either network element can command the other to adjust its transmitted power by transmitting a transmission power control message on either the slow associated control channel (SACCH) or the fast associated control channel (FACCH). This message requests an adjustment between −32 dB and +32 dB relative to the current power. Other messages pertaining to connection establishment and connection release that are used in this phase have been presented in Section 11.5.5.2.

In the communication phase it is possible to employ the optimum channel and the optimum protocol for each service.

11.6 Network Interfaces/Technologies

11.6.1 Private Communication System Application

Home Digital Cordless Telephone The PHS terminal, together with a CS that includes a cordless base unit function, can be used as a home digital cordless telephone, thereby extending the access method of conventional wired service. The air interface between PSs and CSs complies with the previously described PHS air interface standard. To provide security, the standard specifies special features, e.g., scrambling.

Wireless PBX/LAN PHS can also function as a private communication system for office use, such as the digital wireless Private Branch eXchange (PBX) system and the wireless Local Area Network (LAN). The digital wireless PBX system consists of Personal Stations (PSs), Cell Stations (CSs) and a digital PBX, which is responsible for the overall system control. The air interface between PSs and CSs complies with the previously described PHS air interface standard.

One possible future development is the integrated services concept of the fixed and the mobile networks. The PHS MoU [5] group has specified the minimum common technical requirements for PHS/GSM dual-mode terminals, which realize the integration of the GSM service and the PHS home-office cordless telephone service.

11.6.2 Public PHS

This application of PHS technology allows a wide-area public communication service with moderate mobility. It provides coverage for urban and suburban areas, both outdoors and indoors. Public indoor spaces include underground, shopping centres, railway stations, etc.

The basic public PHS network consists of Personal Stations (PSs), public Cell Stations (CSs), a PHS Switching Centre (PSC), and a Service Control Point (SCP) that stores PHS subscriber data and location information (Independent Network Type). However, if an existing intelligent network (IN) already offers PSC and SCP facilities, they can be overtaken in order to realize PHS services (Public Network Utilizing type). In both networks, the core equipment is the PSC, which is a digital switching system with ISDN functions including PHS service software. The two types of public PHS networks are shown in Figure 11.13.

The network interface between a CS and the public PHS PSC conforms to an ISDN interface that is modified to carry out PHS-specific functions such as location registration, authentication and handover.

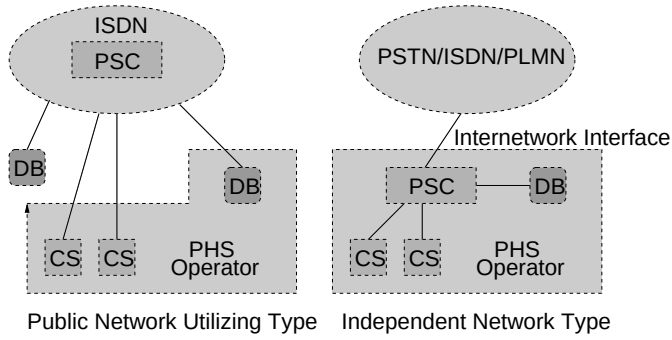


Figure 11.13: Types of public PHS networks

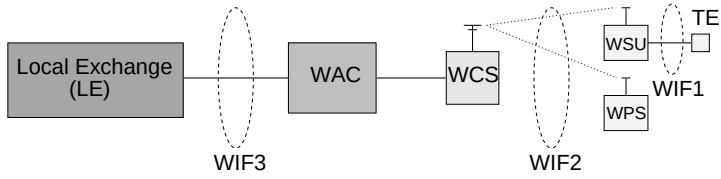


Figure 11.14: PHS Wireless Local Loop network model

11.6.3 Wireless Local Loop (WLL)

The PHS-WLL System is a wireless access system that provides various services, such as plain old telephone service (POTS), high-speed data service, limited mobile service, etc. ISDN services will be supported in the future.

A PHS-WLL network consists of *WLL Access Controllers* (WACs), *WLL Cell Stations* (WCSs), *WLL Subscriber Units* (WSUs) and *WLL Personal Stations* (WPSs). WACs are located between the *Local Exchange* (LE) and the WCSs. The interface between the *WLL Access Controller* (WAC) and an LE, referred to in Figure 11.14 as *WLL Interface 1* (WIF1), can be either an analog or a digital interface, such as an analog two-wire interface, a digital V5-interface, etc. In WLL systems, the PHS-specific functions such as location registration, handover (for WPS) and authentication are managed by the WAC. The *WLL Interface 2* (WIF2) is an air interface based on the standard described in this chapter.

11.7 Standards and References

The PHS technology was developed in Japan and standardized there. The *Association of Radio Industries and Businesses* (ARIB), formerly known as the Research and Development Centre for Radio Systems (RCR), drafted a

standard for the PHS Common Air Interface, which was published as RCR STD-28 [2].

Meanwhile, the *Telecommunication Technology Committee* (TTC) has formulated standards based on ITU-T Recommendations for the PHS User–Network and Network–Network Roaming and Public Cell Station–PHS Service Control Procedures, were published at the end of 1994. The internetworking standards (JT-Q761, 762 and 763) relate to the functions required to connect a PHS call between different PHS carrier networks. The roaming standard (JT-Q1218a), finalized in November 1994, defines the functionality needed to enable a PHS user to make calls via a PHS network with which the user has no contact.

Further extensions to the basic PHS standards are being developed. One such is aimed at Wireless Local Loop (WLL) applications (including support of ISDN services) and another at PHS over cable TV (CATV) networks. PHS has been developed in parallel with the Japanese Personal Digital Cellular (PDC) system, and both have been conceived within an overall evolutionary framework. Just as dual-mode DECT/GSM terminals have been announced in Europe, it is envisaged that dual-mode PHS/PDC terminals may also be developed in Japan.

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12
Wireless Broadband Systems and Wireless ATM*

Broadband systems are generally those systems that provide a high transmission rate besides other features like integration of services. An exact definition of this term can be found in ITU Rec. I.113, which characterizes broadband services as having a higher transmission rate than a primary multiplex connection in ISDN (2048 kbit/s).

A brief overview of the current state of development of wireless broadband systems, particularly in Europe, including the fundamentals of ATM in B-ISDN, is given below. This is followed by important aspects of development of wireless ATM for movable and mobile stations.

12.1 European Research in Broadband Systems

The importance of wireless broadband systems is evident from the number of projects being carried out in this field, e.g., within the European research programme ACTS (Advanced Communication Technologies and Services) [2]:

ACTS/MEDIAN Wireless LAN at 60 GHz transmission of ATM cells;

ACTS/Cobucco Multimedia terminal;

ACTS/FRANS High-bit-rate subscriber connections;

ACTS/MagicWAND Indoor wireless ATM system at 5 GHz;

ACTS/OnTheMove Mobile multimedia value-added services;

ACTS/SAMBA Cellular ATM broadband system at 40 GHz;

ACTS/CABSINET Cellular interactive multimedia communications system for metropolitan areas (at 5, 17, 40 GHz);

ETSI/RES 10 HIPERLAN 1 (wireless LAN with 23 Mbit/s at 5 GHz; see Section 13.2);

*With the collaboration of Andreas Hettich, Arndt Kadelka, Andreas Krämling, Dietmar Petras

ETSI/BRAN Broadband wireless access networks that also support ATM;

ATM Forum TCP over ATM, MPEG over ATM, wireless ATM;

DAVIC/LMDS Digital And Video Council/Local Multipoint Distribution System;

ATMmobil Key development project of the Federal German Minister of Research and Technology: development of wireless ATM systems (at 5, 19, 40, 60 GHz).

A brief discussion of some of the projects follows.

Until 1995, the EU research programmes RACE (Research and Development in Advanced Communications Technologies in Europe) I and RACE II were devoted to the development and testing of prototypes of systems with broadband radio transmission.

From 1992 to 1994 the RACE II programme promoted the development of third-generation mobile radio systems, with the objective of integrating GSM, DECT, paging, mobile satellite radio and trunked mobile radio systems along with their different applications into an *Universal Mobile Telecommunications System* (UMTS) with a multiplex data rate up to 2 Mbit/s at the radio interface. This effort included the development of standardized terminals and an expansion of services with high data rates.

Along with these systems, which were designed to provide a high degree of mobility, the RACE II project MBS (*Mobile Broadband System*) undertook the development and testing of a technology and system concept for a wireless ATM system at 60 GHz that demonstrated the possibility of video transmission with a 16 Mbit/s transmission rate (net) at a 50 km/h speed of movement of the terminal [11, 32].

12.1.1 MBS

The RACE II/MBS project undertook studies of techniques for linking mobile terminals to stationary broadband networks with data rates at the multiplex radio interface of up to 155 Mbit/s. Narrowband services were also to be provided. The MBS system made a particularly important impact, and convinced the professional world of the possibility of providing the services of broadband ISDN to mobile users through wireless ATM transmission [6, 22, 29, 31, 33].

In addition to providing a link-up to the broadband ISDN, the MBS concept also supports a cooperation with other systems such as UMTS. The type of network and level of integration can vary all the way from a privately operated MBS system with a low level of service integration and mobility up to a public MBS system with a high level of integration, extensive mobility and coverage of a wide area [5]. Figure 12.1 shows MBS in relationship to other systems with differing levels of mobility support for their terminals and transmission rates. It can be seen that MBS combines the wide-ranging service spectrum of broadband ISDN with the mobility of mobile radio networks whilst offering the

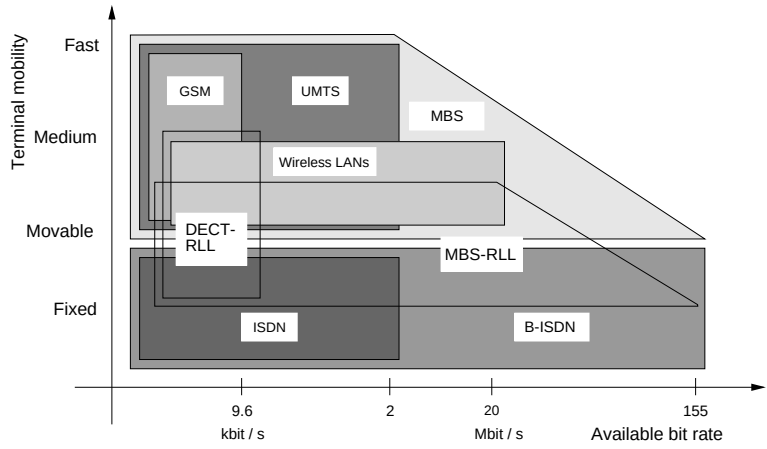


Figure 12.1: MBS and other data networks

services of wideband and narrowband systems such as UMTS, W-LAN, GSM, DECT and their derivatives for Radio in the Local Loop (RLL) applications.

Owing to the flexibility of MBS and the availability of the services of B-ISDN, a variety of different applications are possible. These are indicated (with no claim to completeness) according to the data rates required and the mobility of their users in Figure 12.2.

The author and his research group were responsible for designing the radio and network protocols for MBS, which, although they were not implemented in the demonstrator system, were developed as part of the project and were incorporated into a number of successor projects in the ACTS programme, where they were developed further (see Section 12.1.2). For example, the MBS project as one of the first proposed an ATM-based radio interface for mobile use and specified it as part of the system [29, 30, 36].

12.1.2 Wireless Broadband Communications in the ACTS Programme

As the successor to RACE II, the ACTS research programme [3] of the European Union was conducting field trials and demonstrations to monitor the developed systems for real applications.

Along with the development of UMTS, promising attributes of MBS have been developed further in the following ACTS projects.

12.1.2.1 MEDIAN

MEDIAN (*Wireless Broadband CPN/LAN (*Customer Premises Network*) for Professional and Residential Multimedia Applications*) develops transmission

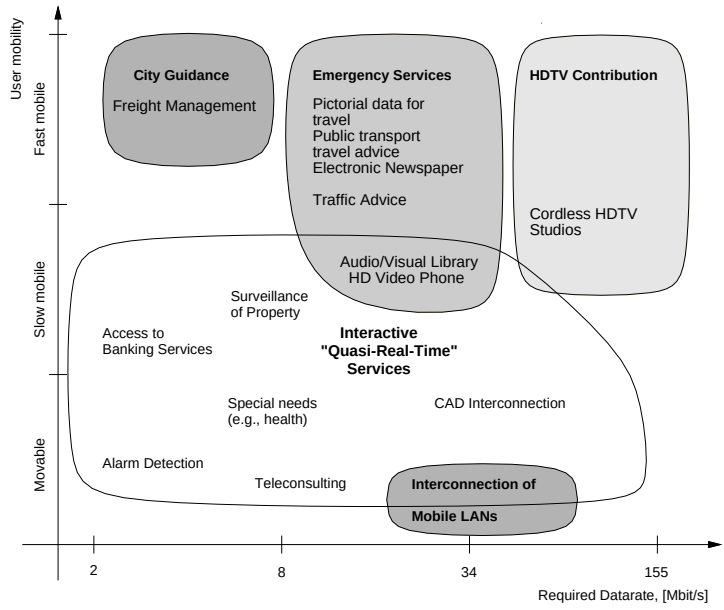


Figure 12.2: MBS applications and services

technology at 60 GHz for wireless networks at data rates of up to 155 Mbit/s for multimedia, voice and video applications.

The goal is to develop a demonstration system for multimedia applications, including research into modulation, channel coding, channel access methods and cooperation with ATM fixed networks at high data rates. MEDIAN provided B-ISDN access to mobile users through transparent transmission of the ATM cells of B-ISDN over the radio interface.

12.1.2.2 Magic WAND

WAND (*Wireless ATM Network Demonstrator*) extended the use of ATM technology to mobile users, and examined realistic user environments. The field of application covered Internet services over ATM in indoor areas with a 20 Mbit/s transmission rate at 5 GHz. The project realized an indoor wireless ATM demonstration network.

The emphasis was on modelling the radio channel and developing channel access protocols, as well as new control and signalling functions to be submitted to ETSI for possible adoption in a later standard for wireless ATM systems.

12.1.2.3 SAMBA

SAMBA (*System for Advanced Multimedia Broadband Applications*) aimed to expand the ATM fixed network using a cellular radio access network to provide mobile users with access to broadband multimedia applications. Mobile ATM terminals have been shown to be able to access services comparable to those used by terminals in the ATM fixed network. Therefore, besides the development of the system elements, the main priorities of SAMBA were integration of ATM fixed network and mobile support. A demonstration system at 40 GHz has been created and demonstrated at the EXPO'98 in Lisbon which provided transparent ATM links with transmission rates of up to 34 Mbit/s for all ATM categories of service.

In contrast to other ACTS broadband projects, the time-critical ATM services CBR (Constant Bit Rat) and VBR (Variable Bit Rate) were also supported and appropriate provisions made for radio protocols so that the radio channel offered a quality of service comparable to a fibre-optic transmission path (within the framework of ATM quality of service requirements).

The SAMBA project also developed a technology, not yet provided by the ATM Forum, for call handover between different ATM fixed network access points. The author and his research group were responsible for the implementation of the protocols of the radio interface and of the ATM network protocols, using their experience from MBS (see Section 12.1.1) [25, 27].

12.1.2.4 AWACS

The AWACS (*ATM Wireless Access Communication System*) project further developed a system based on the NTT/AWA system and developed a demonstrator to support terminals with limited mobility and provide public access to the ATM fixed network. The system operates in the 19 GHz range and provides users with user data rates of up to 34 Mbit/s.

In addition to developing the demonstrator, AWACS carried out extensive research in the areas of channel and source coding, intelligent antennas, optimization of LLC protocols, 40 GHz transmission technology and mobility management.

12.1.2.5 AMUSE

AMUSE (*Advanced Multimedia Services for Residential Users*) specified and developed a demonstrator for advanced multimedia services to link residential customers to an ATM infrastructure. The services were offered under real conditions through the use of different technologies such as HFC (*Hybrid Fibre Coax*), ADSL (*Asymmetrical Digital Subscriber Line*), FTTC/FTTB (*Fibre to the Curb/Building*) and WLL (*Wireless Local Loop*).

A possibility has been explored for setting up end-to-end links to different access networks. Moreover, the individual local field tests have been linked over the European ATM network.

The project was set up in two phases: services such as *Video on Demand* (VoD), *News on Demand* (NoD) and high-speed Internet access were developed in the first phase; other services were offered in the second phase.

12.1.3 ATMmobil

This 1996–2000 programme of the German Ministry of Research and Technology is involved in the development of concepts and the corresponding demonstrators for four forms of wireless ATM systems.

The concept ATM-RLL (*Radio in the Local Loop*) promotes using ATM point-to-multipoint line-of-sight radio at 26/40 GHz to bridge the last few miles in a local loop area.

The second concept (W-ATM LAN) is examining the wireless connection of mobile computers to support multimedia applications at 5 and 19 GHz [8].

The third concept (cellular W-ATM) links mobile terminals with an ATM radio interface over a cellular network at 5 GHz to an ATM broadband network access point [4].

The fourth concept (*Integrated Broadband Mobile System*, IBMS) involves the development of wireless transmission technology for indoors and outdoors. Along with infrared as the medium for indoor purposes, millimetre waves at 5, 17, 40 and 60 GHz are being used. Adaptive antennas, single-carrier transmission technology, radio interface, radio resources and mobility management are the focus of research [12].

Similarly to UMTS for mobile radio systems with multiplex transmission rates of up to 2 Mbit/s at the radio interface, the integrated individual concepts mentioned above are being pursued within the framework of ATMmobil. The author propoded and headed the ATMmobil project, and members of his research group participated in the implementation work of the first, second and third concept mentioned above and were also involved in the ETSI-BRAN standardization (see Section 12.1.5).

12.1.4 The Role of the ATM Forum in the Standardization of Wireless ATM Systems

Although the ATM Forum is not an official standards body, it is playing an important role in the quasi-standardization of certain forms of the ATM fixed network through its association with industry and its products. In June 1996 the ATM Forum became involved in WLAN standardization. The WLAN group originally wanted to focus its attention on mobility support by ATM fixed networks, a project that was supposed to run until the first quarter of 1999 [28]. There are now indications that the radio interface will also be addressed. Based on the experience of 1998, the anticipated market for wireless ATM systems is already so large that the standardization of the radio interface for worldwide use will not be left up to Europe (ETSI) only.

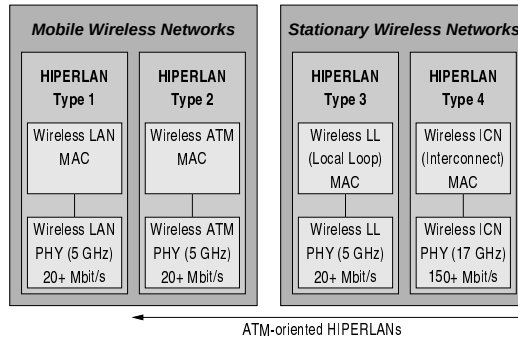


Figure 12.3: The four different types of HIPERLAN

12.1.5 The ETSI Contribution to W-ATM Standardization

The standardization group ETSI RES 10 (*Radio Equipment and Systems*, RES), now ETSI BRAN (*Broadband Radio Access Networks*), is currently developing a family of standards referred to as HIPERLAN (*High-Performance Radio Local Area Network*) for wireless broadband communication at 5 and 17 GHz. There are four different types of HIPERLAN:

HIPERLAN Type 1 is a standard for wireless communication between computer systems at 5 GHz in close proximity to one another (see Chapter 13).

HIPERLAN Type 2 refers to a wireless access system at 5 GHz to ATM fixed networks with a multiplex bit rate of 25 Mbit/s for W-ATM LANs.

HIPERLAN Type 3 is also referred to as HIPERACCESS. It is an application in HIPERLAN Type 2 technology at 5 GHz for outdoor distances of up to 1 km (W-ATM RLL).

HIPERLAN Type 4 at 17 GHz is also referred to as HIPERLINK. It will offer rates of up to 155 Mbit/s for short distances for the connection of W-ATM systems.

The ETSI BRAN group is standardizing the radio interface [10] (see Figure 12.3). The status as of September 1998 can be found in [17].

Table 12.1 presents a comparison of the key features of all four systems.

12.2 Services in Broadband ISDN

Multiplex data rates of up to 155 Mbit/s on the wireless user connection are necessary for the integration of wireless broadband applications in B-ISDN. These kinds of applications require services for continuous interactive data

Table 12.1: Parameters of the ETSI BRAN HIPERLANs

HIPERLAN Wireless	Type 1 LAN	Type 2 ATM	Type 3 Local loop	Type 4 Point-to-point
Carrier freq. [GHz]	5	5	5	17
Network top.	Decentral	Central	PTP	PTP
Antennas	Omni	Omni	Lobe	Directional
Cell type	Pico	Pico	Cigar Like	Radio relay
Area of applic.	Ind./Outd.	Ind./Outd.	Outd.	Ind./Outd.
Operator	Private	Private/Public	Private/Public	Private
Mobility	Port./Move.	Port./Move.	Stationary	Stationary
Backbone	LAN	B-ISDN, ATM	ATM network	B-ISDN
Data rate [MBit/s]	20	24	48	155
Comm. range [m]	50–100	50–100	5000	50–500
Product [Year]	1998	2000	After 2000	After 2000

as well as for bursty-type interactive data. Along with voice transmission, applications with continuous bit streams include video conferencing in which real-time requirements must also be strictly maintained. Interactive services are characterized by a wide fluctuation in the requirements for bit rates. Thus a short information request to a database can result in a very long response (counted in bit time) requiring a high transmission rate. A distinction is made between the following:

- Interactive services
  - Telefony
  - Video telephony
  - Broadband video conferencing
- Inquiry services
  - Access to databases
  - Radio, TV, HDTV, Video-on-Demand
  - Electronic newspaper
  - Videopost
- Data communications
  - LAN links
  - File transfer
  - CAM links
  - High-resolution video transmission

Synchronous transmission methods (*Synchronous Time-Division Multiplexing*, STDM) have difficulty coping with the varying requirements of broadband services. Although an oversizing of the transmission capacity of synchronous channels reduces waiting times, it results in a poor utilization of capacity in the transmission medium. ATM technology (*Asynchronous Transfer Mode*) is better suited to dealing with the demands of broadband services.

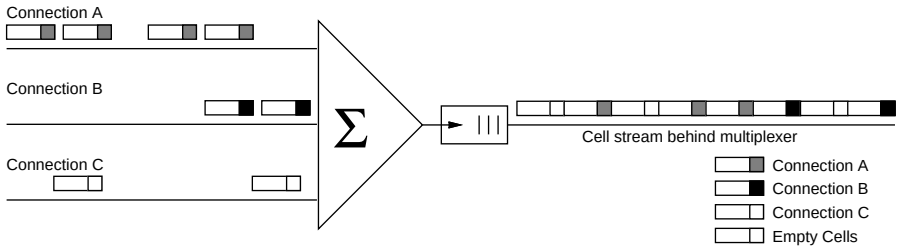


Figure 12.4: Statistical multiplexing of cells to a medium

12.2.1 ATM as a Transmission Technology in B-ISDN

Asynchronous Transfer Mode (ATM) is the connection-oriented packet-switching method used in B-ISDN. ATM combines the advantages of connection and packet-oriented switching—specifically the statistical multiplexing of data from different connections to one medium and the message switching of packets in the network nodes between the communicating terminals. The data streams to be transmitted are divided into short blocks of a fixed length, referred to as ATM cells. The cells of different connections are transmitted with time interleaving over a physical channel. Depending on their data rates, the connections are dynamically allocated varying amounts of transmission capacity, with some of them transmitting a large number of cells per time unit and others only very few. The cells in each connection are transmitted in the order of their arrival.

The ATM multiplexer adds empty cells to the multiplex data stream if none of the connections requires transmission capacity and a synchronous transmission method is being used (see Figure 12.4).

12.2.2 Structure of an ATM Cell

An ATM cell comprises 53 bytes, consisting of a 5-byte long header field and a 48-byte long information field containing data of the higher ATM protocol-stack layers and user data. The switching of the cells is connection-oriented. All cells in a virtual connection take the same transmission path, which was established when virtual channels were set up on different switching sections in the network during call setup. The cells are controlled through the network on the basis of the routing information stored in the cell header (see Figure 12.5).

`VCI` **(Virtual Channel Identifier), 2 bytes** An identification of the virtual channel differentiates between the different concurrent logical channels and their cells. The virtual channel number is always only assigned to one switching section.

`VPI` **(Virtual Path Identifier), 8 or 12 bits** A channel group is identified by the parameter `VPI`. A differentiation can be made between a large num-

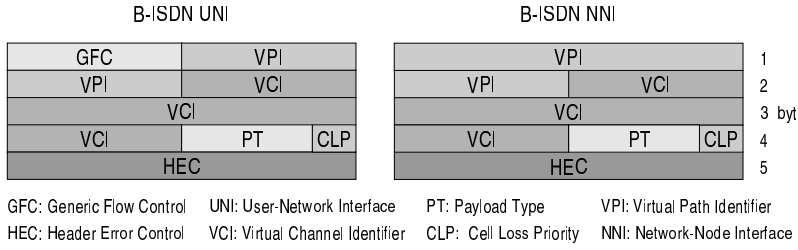


Figure 12.5: Header of an ATM cell

ber of groups in the same direction, each containing several virtual channels. The cells of channels in the same group can be processed especially quickly by the switch-fabric of an ATM switch and then forwarded; *cross-connects* are used for this purpose.

`PT` **(Payload Type), 3 bits** This parameter describes the type of information field and provides a differentiation between user and signalling information. The signalling information is required, for example, to update the routing tables managed in the switching centres. The switching centre carries out updates by evaluating the header field as well as the information field of the ATM cell. User data being transmitted in the information field of an ATM cell is not taken into account in the switching.

`HEC` **(Header Error Control), 1 byte** Because the header of an ATM cell contains data that is vital to the transport of the cells, it is protected by a checksum. This permits the detection of transmission errors.

`CLP` **(Cell Loss Priority), 1 bit** This parameter identifies cells with a lower priority, which are discarded in the ATM switching centre when there is an overflow in the queue. The bit is also used by multiplex and switching nodes for flow control and traffic shaping.

12.2.3 ATM Switching Technology

As with other packet-switched methods, ATM cells are switched on the basis of the routing information contained in the cell header. The complete originating and target addresses are sent only during connection setup, so that the routing information can be kept as short as possible, thus increasing throughput. Identification of the logical channels is assigned for the different sections of a connection (`VCI`, `VPI`). During connection setup, the ATM switching centres enter relationships between input and output identification (line + logical channel identification) into their routing tables using incoming control information.

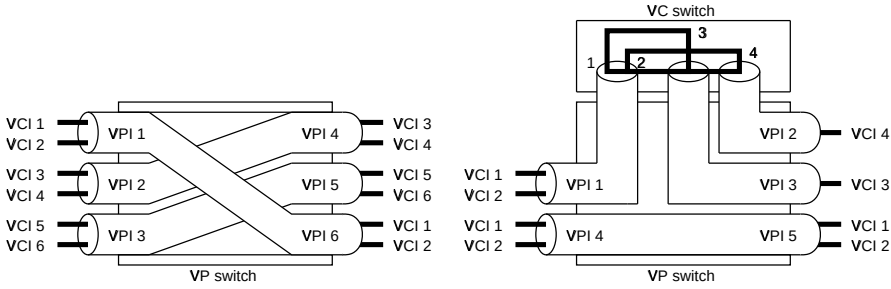


Figure 12.6: Virtual path switching und virtual channel switching; VCI=Virtual Channel Identifier, VPI=Virtual Path Identifier

The switching of the cells is based on the information in these tables and takes place as follows. Switching control derives the logical channel identification (`VCI`, `VPI`) from the incoming cells. The identification `VPI`, `VCI` of the next connection section is obtained through the relationship of the incoming and outgoing pairs (`VCI`, `VPI`) in its routing table, and entered into the cell header, and the cell is then switched through to the appropriate exit of the switching network.

Figure 12.6 shows the switching elements used in the switching centres. A distinction is made between *VP switches* and *VC switches*. The VP switch only evaluates the `VPI` values of the cells, so that the cells can be switched quickly. The entries of the `VCI` fields remain unchanged. The `VCI` identities are changed only if transmission has occurred through a VC switch.

12.2.4 ATM Reference Model

According to the recommendations of the OSI reference model, an ATM reference model with four layers can also be specified (see Figure 12.7). These layers include the *physical layer*, the *ATM layer*, the *ATM adaptation layer* (AAL) and a layer that represents the functions of the *higher layers*. Since the physical layer corresponds to layer 1 of the ISO/OSI reference model and the ATM layer corresponds to layer 3 (network), the data link layer has not been taken into account for the ATM reference model. This is because ATM networks typically use fibre links that guarantee a very low bit-error ratio (BER) and therefore do not need any error control in layer 2.

Furthermore, three different planes are defined: the *user plane*, the *control plane* and the *management plane*. The management plane comprises the functions for *plane management* as well as the functions for *layer management*. Plane management is responsible for management of the entire system, whereas layer management controls the individual layers.

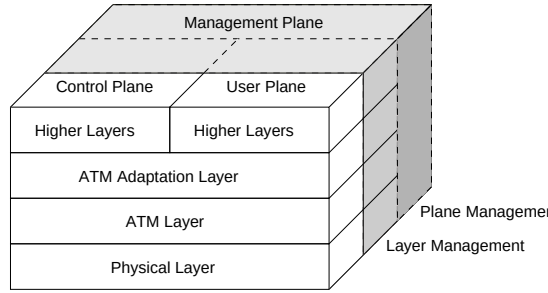


Figure 12.7: The ATM reference model

The services, protocols and interfaces of the physical layer depend on the transmission medium used, and contain all the functions required for the bit-by-bit transmission of information. The tasks of the ATM layer include:

- Multiplexing and demultiplexing cells of different connections
- Control of `VCI` and `VPI`-oriented functions
- Generation or evaluation of cell header information
- Generic flow control at the user-to-network (UNI) interface
- Establishment, routing, operation and release of connections

The ATM adaptation layer adapts the higher ranking layers to the ATM layer. It carries out the necessary segmentation of data streams into cells on the transmitting side and their reassembly into messages on the receiving side, and ensures safe transmission. The AAL layer is divided into two sublayers:

1. The *Segmentation-and-reassembly* (SAR) sublayer for mapping protocol data units of the higher layers to the ATM cells and the reverse.
2. The *Convergence sublayer* (CS), which compensates for the undesirable effects caused by the different cell transit times of different services.

For example, the digital sample values used for voice transmission are consolidated over a certain period of time to enable the complete utilization of the capacity of an ATM cell. The receiver generates a continuous data stream again from the arriving ATM cells.

The different cell transit times through the network are offset by the ATM adaptation layer in the receiver through the addition of a constant time delay before the output.

Table 12.2: Classes of AAL Services

Class	1	2	3	4
Time rel.	Continuous	Continuous	Non-cont.	Non-cont.
Bit rate	Constant	Variable	Variable	Variable
Connect.	Conn.-orient.	Conn.-orient.	Conn.-orient.	Conn.less
Examples	Emulation of sync. circuit switching (voice)	Variable bit rate (video)	Conn.-orient. data trans.	Conn.less data trans.

12.2.5 ATM Classes of Service

The number of protocols required for the ATM adaptation layer has been kept to a minimum through the classification of services into four different groups according to the following parameters: time relationship between source and sink, bit rate and type of connection (see Table 12.2). Time-continuous services with constant or variable bit rates are differentiated according to those that are connection-oriented and those that are connectionless.

In accordance with the class of service, control data is also added to the user information. This is used to allow the restoration of user information that is divided among several cells. The control data is generated by the SAR sublayer when the data is divided into cells. In the receiver the corresponding SAR sublayer must join the data together in the correct sequence according to the control data.

For identification of the sequence of the individual ATM cells, each message contained in a cell is assigned a sequence number to enable the receiver to detect the loss of any cells. Whereas the sequence number is available with all the classes of service, additional backup data and different segment types are being planned only for some of the classes. Owing to the varying control data part, the user data part is 44–48 bytes.

In its specification *Traffic Management (V 4.0)* [13] the ATM Forum differentiates between different classes of service that represent different applications. The specified quality of service parameters for the following classes of service are listed in Table 12.3:

Unspecified bit rate No quality-of-service parameters are specified for the UBR class of service. No guarantees are provided for this class, which means that it only benefits from a *best-effort service*.

Available bit rate The ABR class of service is particularly suitable for applications that are not real-time-oriented and also have no requirements in terms of transmission rate. Only the cell-loss rate is defined as a parameter for the quality of service.

Constant bit rate The CBR class of service is planned for real-time-oriented services with constant bit rates that have a stringent requirement for

Table 12.3: Classes of service and their quality of service parameters

	ATM Layer Service Classes				
Attribute	CBR	VBR(RT)	VBR(NRT)	ABR	UBR
CLR	×	×	×	×	—
CTD	×	×	×	—	—
CDV	×	×	—	—	—

CBR	Constant Bit Rate	VBR	Variable Bit Rate	UBR	Unspecified Bit Rate
RT	Real-Time	NRT	Non-Real-Time	ABR	Available Bit Rate
×	Specified	—	Unspecified		

cell-loss ratio (CLR), *cell-transfer delay* (CTD) and *cell-delay variance* (CDV).

Non-real-time variable bit rate The VBR class of service is a compromise between the VBR and ABR classes of service. This is used for applications that require more quality-of-service guarantees than the ABR class but attach no importance to the assurance of a specific variance in cell transmission times (CDV).

Real-time variable bit rate Real-time-oriented applications that place a stringent demand on delays and their variance as well as on cell loss ratios require a real-time VBR service.

12.2.6 Functions and Protocols of the AAL

Because transmission errors cannot be completely prevented even with fibre-optic technology, an end-to-end error correction procedure dependent on type of service is provided in the AAL layer.

The AAL protocols type 1 and type 2 are used for the real-time-oriented CBR and VBR services. They supply their *protocol data units* (PDU) with sequence numbers and check sums to aid in the detection of lost or incorrectly inserted ATM cells. An FEC procedure can be used as an option to correct bit errors [19]. If the bit-error ratio in the ATM layer exceeds the correction capabilities of the code used—something that may particularly occur from time to time on a radio transmission path—the quality of service requested by the user cannot be guaranteed by this procedure.

An ARQ protocol based on the functions for detection of bit errors and cell loss in the lower AAL sublayers (*Common Part Convergence Sublayer*, CPCS and *Segmentation and Reassembly*, SAR) is provided in the highest sublayer of the AAL (Service-Specific Convergence Sublayer, SSCS) in AAL protocols type 3/4 and type 5 [14]. According to [15], these ARQ protocols can be executed efficiently with a packet-loss probability of 10^{-3}. If the packet-loss probability for packets 1 kbyte in length is to be maintained then the bit-error probability 10^{-7} cannot be exceeded. The bit-error probability of a

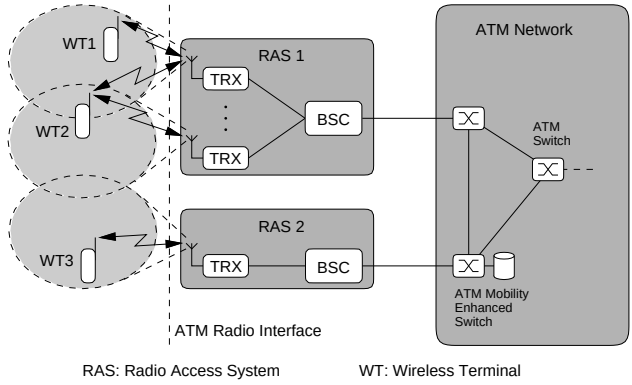


Figure 12.8: Architecture of a cellular ATM mobile radio network

radio transmission path protected by an FEC procedure usually lies above that limit, and is therefore too high for an ARQ procedure to be carried out efficiently in the AAL.

12.3 Architecture of the ATM Radio Interface

Figure 12.8 illustrates the schematic structure of a cellular ATM mobile radio system [9, 32]. The access points to the ATM fixed network are located between the *Radio Access System* (RAS) and the ATM fixed network. Each access network contains one or more transmit and receive facilities (*transceiver*, TRX) as well as a *base station controller* (BSC) that carries out the protocols of the base station. The UNI interface is usually provided between the RAS and the ATM network.

This radio networks offer wireless ATM access from moving or mobile *wireless terminals* (WT) in selected areas, e.g., in buildings, outdoors or in the vicinity of buildings.

12.3.1 The Radio Access System as a Distributed ATM Multiplexer

Wireless local area networks (W-LAN) are a typical application for cellular ATM mobile radio networks. With limited operating time (because of battery-based power supply) and reduced data rates (because of radio transmission), it is desirable to provide wireless terminals in W-LANs access to the same services available to ATM terminals linked to a fixed network. In particular, it should be possible for all ATM applications to be used without any need for modification, i.e., in wireless as well as in wired terminals based on the same AAL services.

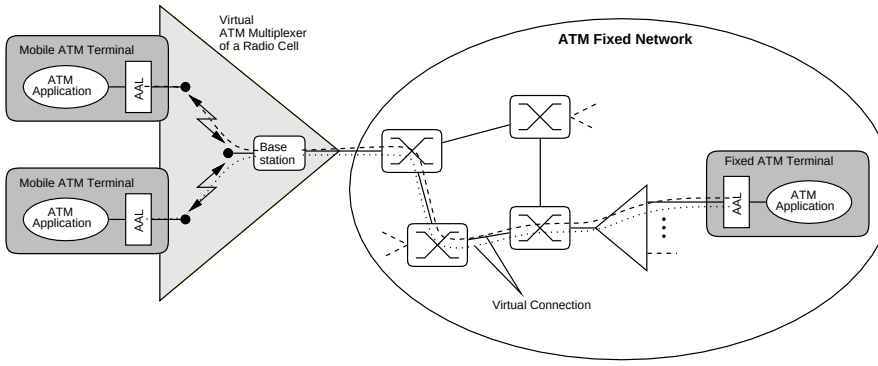


Figure 12.9: Connection of a cellular ATM radio network to an ATM fixed network

Figure 12.9 clarifies how AAL protocols are end-to-end transport protocols because they are operated between terminals only and do not occur in the network nodes. Transmission over the ATM radio interface takes place within the ATM layer with the help of individual ATM cells, with the influence of the radio interface remaining hidden from the service users of the ATM layer (the entities of the AAL). This is referred to below as transparent transmission of ATM cells. From the standpoint of the user, the terminals of a radio cell that operate virtual connections over the RAS behave as if they were connected over a cable to an ATM multiplexer (see Figure 12.9).

12.3.2 Frequencies and Frequency Etiquette for W-ATM Systems

The 5.15–5.25 GHz frequency band is currently being provided for W-ATM systems in Europe as well as in the USA, although CEPT has allocated this band to HIPERLAN 1. The FCC in the USA makes reference to *High-Speed Multimedia Unlicensed Spectrum* for its *National Information Infrastructure* (NII) wireless broadband systems projects. WRC is still studying the problems involved. There are national problems with the 5.25–5.3 GHz expanded HIPERLAN band due to the original usage and in the entire 5.25–5.35 GHz band in the USA. WRC99 will probably be allocating the entire 5.15–5.87 GHz band for use by broadband local radio networks. Bands already being used will have to be reallocated for use by broadband radio networks; this is called *refarming* a band. Because of the large number of different systems (old and new), anticipated systems must ensure that they have the ability to coexist.

12.3.2.1 Frequency Etiquette

The assumption is that different standards will have to coexist with one another in the same band. Frequency-sharing rules or etiquettes are therefore

being discussed, and these would apply to the use of a band at a particular location by systems built on the basis of the same or different standards.

In general, an etiquette is a set of rules that is agreed by the parties concerned so that a group of individuals is able to cooperate with one another without having to communicate whenever a problem arises.

Instead of an exclusive frequency allocation for a particular system, users more likely expect an operation similar to what is usually found in *Industrial, Scientific and Medical* (ISM) bands but with additional restrictions, e.g., parameters for the times when a band can be used.

It was proposed that shared bands should be used in such a way that systems reserving the band (e.g., circuit or period-oriented systems) are separated in the band from systems transmitting in packet-oriented mode and that there be an overlap area that can be used by both types of system if necessary [16]. For the time coordination of the transmission of different systems in the same band, a technique is favoured that uses time containers rotating cyclically between the base station transceivers (TRX) of coexisting base stations and should be capable of enabling neighbouring (interfering) systems to transmit on an alternating basis. A periodic container used by a given base station is considered a TDMA channel with a capacity defined by the container size in bits, and defined by the length of the periodic frame (see Section 12.3.8). It has been proposed [26] to introduce a dynamic channel selection scheme as used by the DECT system (see Section 9.5) to enable base stations to decentrally access and acquire transmit capacity in terms of a number of containers from a given frequency band.

Since the capacity of a frequency channel must be controlled in the area defined by the interference range of a base station (which is much larger than the service range) in order to be able to guarantee quality of service for ATM connections according to the standard, but interference ranges of stations are partly overlapping, there is still work needed to define appropriate algorithms.

12.3.2.2 Transmission Procedures for W-ATM Systems

Since a high multiplex transmission rate of approximately 25 Mbit/s at the wireless interface is being sought, DS-CDMA (*Direct Sequence Spread Spectrum Code Division Multiple Access*) procedures will not be appropriate because of the bandwidth required. Newer demonstration systems use *Orthogonal Frequency-Division Multiplexing* (OFDM) with symbol-by-symbol parallel signal transmission over 16 or 64 FDM carriers on the downlink and with some systems also on the uplink. In addition, 64 QAM and hybrid modulation methods are used. Capacity is usually distributed in the frequency and time domains (FDM, TDM, FDMA, TDMA, FDD, TDD). Recent experiments have involved the use of adaptive antennas for the prevention of interference and improvement to quality of service. *Space-Division Multiple Access* (SDMA) can also be used for this purpose [34].

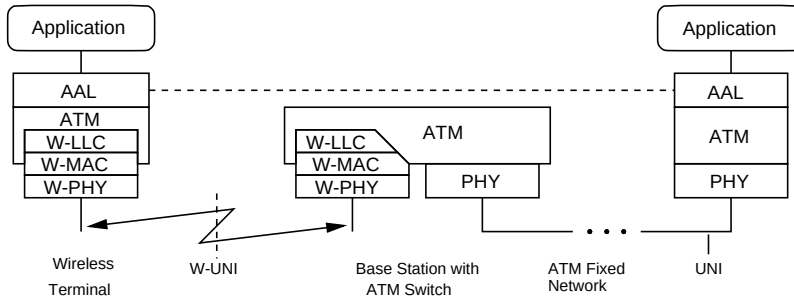


Figure 12.10: Protocol stack for the ATM radio interface (user plane)

12.3.3 Protocol Stack for the ATM Radio Interface

Compared with the fixed network, the radio interface as a distributed ATM multiplexer requires that certain radio-specific aspects also be considered:

Radio propagation For example, diffraction, shadowing, reflection and multipath propagation.

Channel access Coordination of access to shared-use radio channels for implementing the transmission sequence of the ATM cells specified by the scheduler function of the RAS.

Error protection Unreliable transmission conditions on radio channels necessitate the use of error protection schemes to fulfil the quality of service requirements of the individual virtual connections according to the different categories of service.

The quality of service required on a radio path is ensured whereby, along with AAL error protection measures, a data link control protocol (ARQ protocol; see Section 2.7.4) is used directly at the radio interface in the logical link control (LLC) sublayer to guarantee transparency in relation to the AAL. An ARQ protocol specific to the class of service is used, which permits error protection in the LLC layer to be adapted to the individual requirements of each separate virtual connection over a separate access point per ATM service class.

Figure 12.10 shows the resulting protocol stack at the radio interface *Wireless UNI*, which consists of a physical layer that considers the characteristics of the radio transmission (*Wireless Physical Layer*, W-PHY) and a *Data Link Control layer* (DLC). The DLC layer consists of a sublayer for the coordination of channel access (*Medium-Access Control*, MAC) and a sublayer that controls the logical channels and contains the error protection functions (*Logical Link Control*, LLC).

The protocol stack shown in Figure 12.11 is currently being discussed by ETSI BRAN. Along with the W-ATM terminal, a W-ATM access point and

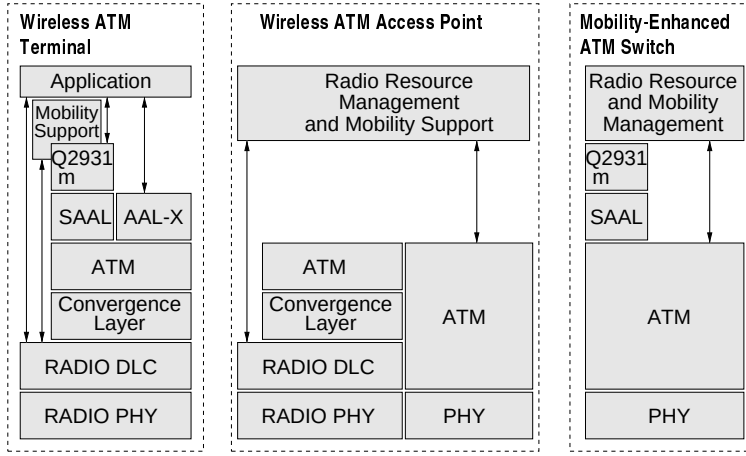


Figure 12.11: HIPERLAN/2 layers and architecture

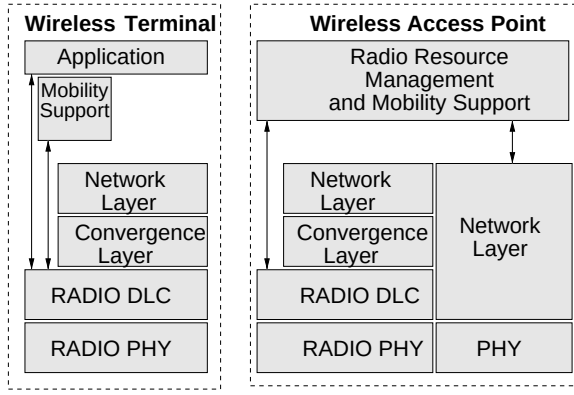


Figure 12.12: HIPERLAN/2 layers and architecture

an ATM switch that has been expanded to include mobility support are shown. The ITU-T signalling based on Q.2931 is located above the *signalling ATM adaptation layer (SAAL)* (SAAL), whereas applications are based on the AAL-X (X = 1, 2, ..., 5). Two layers at the radio interface (R)—the data link control layer and the physical layer—are responsible for safe transmission with error recovery.

Figure 12.12 shows an alternative design of the ETSI/BRAN protocol stack under discussion that is heavily supported by manufacturers aiming to provide wideband and broadband Internet services to their customers. Instead of a W-ATM terminal, any class of wireless broadband terminals can be supported by this protocol stack. This design favours the straightforward support of Internet applications by using the *Internet Protocol* (IP) in the network layer instead of ATM signalling. A convergence sublayer is provided in both pro-

tocol stacks to enable use of the air interface provided by HIPERLAN/2 by any network service and related protocols.

12.3.4 Channel Access

In wireless ATM networks several users have to share the same physical channel. Access to the medium must accordingly be coordinated. In the projects described in Section 12.1 and in other company-specific work groups (worldwide) different channel access protocols have been developed and proposed for consideration in the ETSI/BRAN standardization. The approach incorporated in these protocols is discussed below.

Through the use of TDMA, the physical channel is divided into time slots of the same or of varying length. Within these time slots, signalling information and ATM cells with user data are exchanged between a mobile terminal and the base station controller. The allocation of transmission capacity is carried out dynamically in accordance with the capacity requirements of the terminals.

Most approaches consist of access to the shared physical channel being coordinated by a central entity: the base station. It carries out the dynamic allocation of transmission capacity to the terminals in the form of time slots, introducing a system of priority in which connections with high quality-of-service requirements are given priority. The base station therefore requires information on the number and the class of service of the ATM cells waiting for transmission in the terminals.

At the radio interface the base station produces a distributed ATM multiplexer, which is created by a buffer and a sequence controller (*scheduler*) in the multiplexer: the terminals buffer the cells waiting to be transmitted; the base station controls the sequence of the cells being transmitted over the radio channel, and also considers whether the cells received with errors should be retransmitted.

Sending a signalling message on the downlink, the base station identifies which future time slots following the signalling message on the uplink are available exclusively to a terminal for transmission. Likewise, other time slots for downlink transmission are declared at certain terminals, thereby producing a periodic structure (see Figure 12.13). All information that occurs in the upcoming period is stipulated within a signalling message, followed by the signalling message for the next period.

A period always consists of a downlink and an uplink transmission phase. In the downlink data transmission phase, information is transmitted from the base station to the terminals within a reserved time slot; in the uplink data transmission phase, the reverse occurs. The length of a period can be variable on a constant or a dynamic basis. The ratio of the duration from the uplink to the downlink data phase can be variable or constant. A variable ratio makes it easier to take asymmetrical traffic relationships into account, thereby resulting in better utilization of channel capacity of the radio medium.

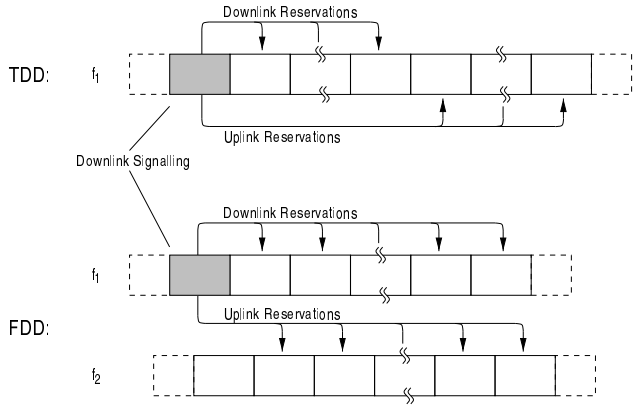


Figure 12.13: MAC signalling using Time- and Frequency-Division Duplexing (TDD and FDD) respectively

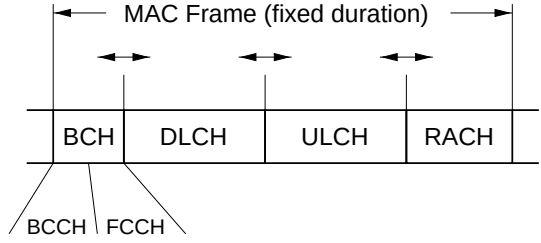


Figure 12.14: Signalling period according to ETSI/BRAN

Either TDD or FDD can be used for the directional separation of transmission (see Figure 12.13).

Requests for capacity (e.g., number of waiting cells) can be transmitted piggyback with ATM cells during the uplink data phase or through signalling time slots on the uplink that can be reserved for the different terminals or used in random access. Since a signalling time slot is short compared with an ATM cell because of the message length, partial time slots can be used. Restricted because of the radio range, a W-ATM base station usually serves only a small number of active terminals at any one time. Therefore a reservation-based *polling* method can be suitable for the transmission of capacity requests from terminals to the base station [18].

According to ETSI/BRAN, the structure of a period is shown as a MAC frame in Figure 12.14.

The frame contains four elements:

- Broadcast Channel (BCH) with two subchannels:
  - Broadcast Control Channel (BCCH)

  - Frame Control Channel (FCCH)

  The BCCH carries cell-specific information. The FCCH allocates time slots of the current period to terminals and acknowledges received data by the base station.

- Downlink Channel (DLCH) with time slots allocated by the FCCH for transmission of data to well-defined terminals.
- Uplink Channel (ULCH) with time slots reserved by the FCCH for uplink transmission by well-defined terminals.
- Random Access Channel (RACH) with short time slots dedicated to specific terminals and/or available for random access by terminals to signal their transmission capacity needs to the base station.

12.3.5 The LLC Layer

Fibre-optic cable is used as the transmission medium in the fixed network. In addition to providing a high transmission capacity, the medium is characterized by a very low packet-error ratio in the range of 10^{-9}. Protection against transmission error in, for example, real-time-oriented services is therefore not necessary. Data transfer services that require even lower error ratios are protected against transmission error in the ATM adaptation layer (see Section 12.2.6).

A radio channel, on the other hand, is a transmission medium with a high level of packet error ratio on the scale of 10^{-3}. Since even real-time-oriented services may have higher requirements than that, all services may need protection against transmission error. There is a choice of three different methods:

- FEC (*Forward Error Correction*): error protection through error detection and correction at the receiver side;
- ARQ (*Automatic Repeat Request*): error recovery by packet retransmission when an error is detected;
- error recovery in combination with FEC.

The third method, with the physical layer of the protocol stack of the radio interface taking over the function of *forward error correction*, is the most commonly used. An ARQ protocol that protects connections from transmission error is provided in the *Logical Link Control* (LLC) layer.

12.3.5.1 Cell Discarding

Conventional ARQ protocols are suitable for services that are not time-critical. With time-critical services the maximum transmission delay of data packets should also be taken into account, because there is no point in transmitting

an excessively delayed packet that is taking transmission capacity away from other waiting packets. The sender should transmit all data packets successfully before exceeding the allowable transmission delay. When it is seen that this will not be possible with some of the packets, these are no longer required to be transmitted and are discarded. A distinction is made between two different situations:

1. A packet was discarded before it was transmitted.
2. A packet is to be retransmitted because it was requested again.

In the first case, there is no need for the sender to inform the receiver about discarding, because the receiver is not aware that a sequence number has been assigned and therefore will not be able to know that a discarded packet is missing. In the second case the data packet has already been assigned a sequence number that is known to the receiver. The receiver must therefore be informed of which packet has been discarded, or else it will block the transmission and not forward all the subsequent packets to the next-higher layer, since one of the packets in the sequence is missing.

12.3.6 Dynamic Capacity Allocation with Packet-Oriented Radio Interfaces

A particularly efficient form of media access control is required for packet-oriented radio interfaces, because this process is constantly needed for each packet and not only once per connection for setting up a channel as is usual with many second-generation mobile radio systems.

It is very easy for a central base station to reserve transmission capacity on the uplink; on the downlink it determines how it is used anyway. A base station either is connected directly to the fixed network or uses a wireless connection in order to reach a neighbouring base station on the route to the fixed network access point. One simple way of linking base stations (node) together for packet transmission is under the control of a central node (see Figure 12.15).

Current proposals for controlling access rights to the radio medium are therefore based on the use of central facilities [24, 29]. In simpler versions transmission capacity has to be requested packet-by-packet by means of the S-Aloha protocol, which is a very inefficient procedure. The best procedure entails using a transmitted packet to transmit a reservation request for the subsequent packet if the latter is already in the queue.

Uplink use, centrally coordinated by a base station, will not be subject to any interference as long as no other base stations are active in its interference range. Otherwise the mobile stations of different base stations would be transmitting at the same time or overlapping and possibly colliding.

Packets received from a mobile station's own base station could also be affected by interference from other base stations.

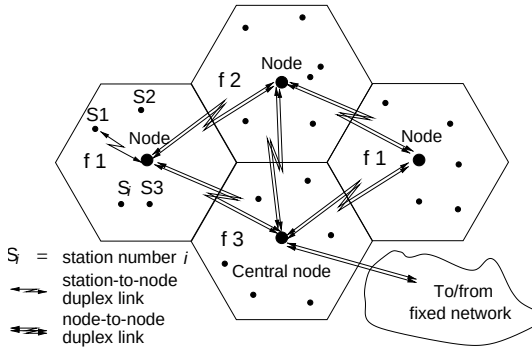


Figure 12.15: Cellular wireless ATM network with central control of stations by the node of the cell, where nodes of frequencies f_i in a 3-cluster and wireless communication between nodes is controlled by a central node.

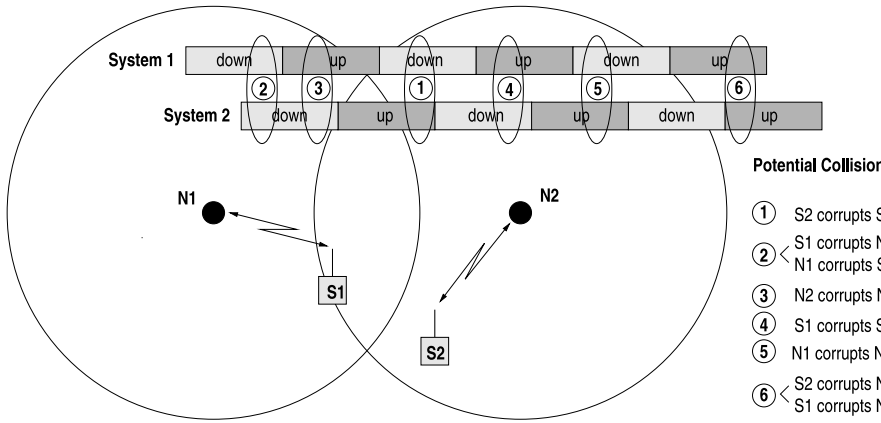


Figure 12.16: Unlike in cellular TDMA/FDD networks, with TDMA/TDD and non-synchronized nodes all the cells interfere with each other at random times.

Cell clusters in which each cell has a fixed allocation of frequencies are used to avoid reciprocal interference from the base station supply areas (see Figure 12.15). Frequency etiquettes for coordinating the times when frequencies are used by a base station under a decentral control are also being discussed; see the container technique mentioned in Section 12.3.2.1.

If clustering is not considered then a number of possibilities are available for reciprocal interference to neighbouring base and mobile stations (see Figure 12.16). In the illustration the coverage range is shown as a circle around each base station, and the assumption is that the interference range is clearly larger and that the two centrally controlled systems are not synchronized with base stations N1 and N2. The possibility is obvious that any combination of base and mobile stations is capable of disrupting each other. If this disruption

is to be avoided through the use of static frequency planning (with clusterization) then the cluster size must be clearly larger than is the case with synchronized base stations.

Frequency etiquettes with fixed channel allocation aid in the prevention of reciprocal interference, but tend to reduce spectral efficiency compared with systems with dynamic channel allocation. This is because capacity is allocated to stations and systems at fixed times and cannot be changed dynamically, e.g., when a system does not need the current capacity but a neighbouring system is experiencing a bottleneck.

The prerequisite for dynamic flexible channel allocation under co-channel interference is a criteria that allows a projection of future usage based on current occupation of the medium.

12.3.7 Channel Concept for a Packet-Oriented Radio Interface

The following statements generally apply [7]:

1. A radio channel can be divided among several users if they require an already-known constant bit rate (CBR).
2. During a sufficiently short period of time T_c, each communications relationship possesses a CBR characteristic.
3. The time needed for setting up a connection can be regarded as the practical lower limit for T_c.
4. If a terminal is multiplexing several services on a connection then the time needed is greater than T_c.
5. Multimedia terminals require protocols that support centrally controlled and ad hoc networks.

Channel-oriented reservation-based medium allocation is an interesting option if the following characteristics are required:

- Simple allocation of transmission capacity
- Support of terminals with highly variable bit rates
- Mechanisms for avoiding collisions
- Transmission over radio by terminals using different modulation techniques
- Transparent support for layer-3 protocols and applications based on them
- Compatibility with centrally and decentrally-controlled (ad-hoc) protocols

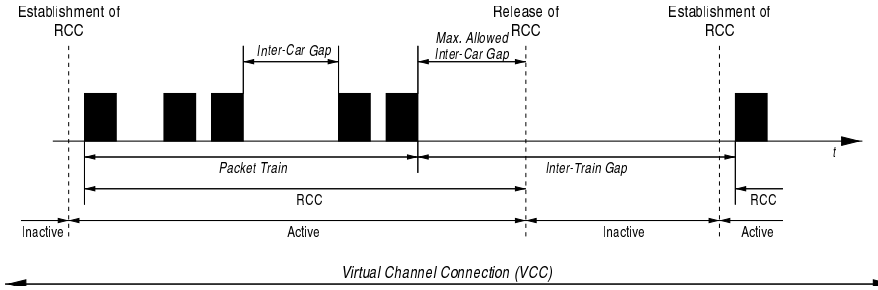


Figure 12.17: The packet train model (RCC = Real Channel Connection)

12.3.7.1 Bursty Sources of Information and Packet Trains

Bursty traffic sources can be described using the *packet train model* [21], which is a finite state machine model with at least two states in which "cars" (packets) of the "train" (close sequence of packets) are produced after a random short period of time in one state and a longer random period of time in the other (see Figure 12.17). The longer intervals correspond to the inter-train times; the shorter intervals to the inter-car times.

If the inter-car gap T_i is regarded as a parameter then different train lengths can be defined for the same traffic source, thereby influencing the frequency of the trains [35]. Obviously T_i should correspond to T_c, the time needed to establish a channel, with T_i increasing with T_c.

According to Figure 12.17, an existing virtual connection is only supported with the transmission capacity of the radio channel (*real channel connection*, RCC) for the duration of a train, and the capacity is withdrawn once the train has come to an end, i.e. the inter-car gap has exceeded.

The receiver uses a reverse channel with the correct transmission rate to communicate with the sender in order to acknowledge individual *cars* or groups of cars of the train, in other words to transmit its own data and send acknowledgements piggybacked. In rigid TDD systems with fixed time intervals for transmission facilities and the same transmission rates per direction, the occupancy of a TDD channel also leads to the occupancy of the one in the opposite direction. With asymmetrical allocation of channel capacity each receiver must indicate the receive time channel in its transmit channel so that other stations can be made aware of the state of occupancy.

This allows a bidirectional physical channel to be recognized as occupied and respected by all stations currently in the detection radius of the respective receiver during the duration of a train, thereby avoiding spontaneous interference by accessing stations, which are hidden stations as far as the receiver is concerned.

What remains to be determined is the amount of capacity to be allocated per physical channel for the duration of a train and how connection setup takes place (see the recommendations in [7, 35, 37]). Suggestions for this

are also given in the DECT standard providing a multilink protocol; see Section 9.4.2.6.

12.3.7.2 Connection Admission Control for ATM Radio Interfaces

If the packet train model is followed and physical connections are set up only for the duration of need then the connection admission controller must ensure that for each accepted virtual connection each new train is guaranteed to receive a physical channel with only a short delay. Therefore the mechanisms familiar from ATM networks appear to be directly transferrable.

If the decentral organization of an ad hoc network is selected then the connection admission control entity is implemented on a distributed basis. A relevant proposal can be found in [35].

In the W-ATM applications being anticipated, VBR services requiring real-time support (see Table 12.3) will only make a relatively small demand on capacity utilization. If the plan is to interrupt the physical channels reserved for ABR services at any time despite the packet trains being run then it can be expected that the quality of service of these variable-bit-rate (VBR) services will be guaranteed.

12.3.8 Dynamic Channel Selection for W-LANs

Since a number of base stations will have to share a frequency channel and since the transmit capacity of a base station might change substantially over time, a dynamic channel selection procedure appears to be appropriate to provide a base station with a guaranteed capacity to multiplex the packet traffic of its terminals at the air interface as explained in Section 12.3.4.

For that purpose, a periodic frame is defined for frequency channels where frames contain a number of containers; see Figure 12.18. In the same way as described in Section 9.5 for the DECT system, a WATM base station is allowed to access and reserve one or a number of periodic containers (a so-called container channel) with a capacity according to the current needs [26] and to release the TDMA container channel when not needed further. The occupancy of a container channel might last for seconds or even minutes, and a base station is allowed to operate multiple container channels in parallel according to its current needs.

The (signalling) periods described in Section 12.3.4 are mapped by a base station to its container channels; see Figure 12.19.

This technique apparently fulfils the conditions described in Section 12.3.7 and makes an optimum use of a TDMA container channel possible. The only requirement is that base stations respect, i.e., not access, container channels observed by their own measurements or by measurements of their mobile terminals to be currently in use.

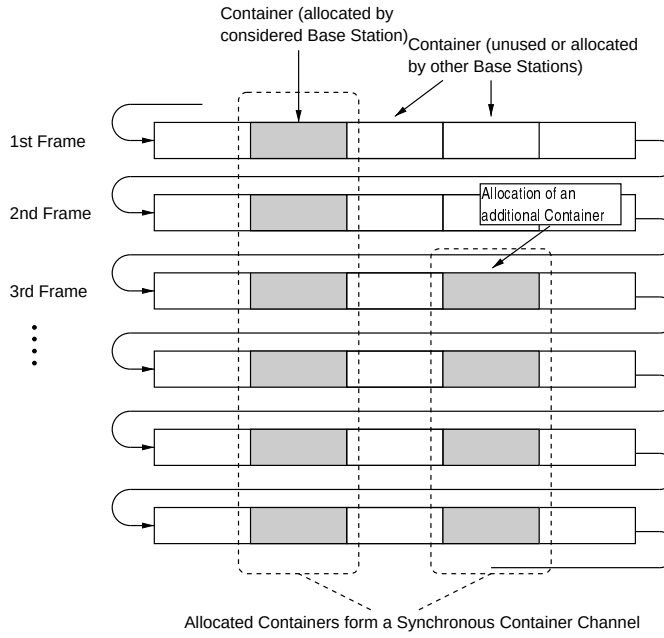


Figure 12.18: Container Channel

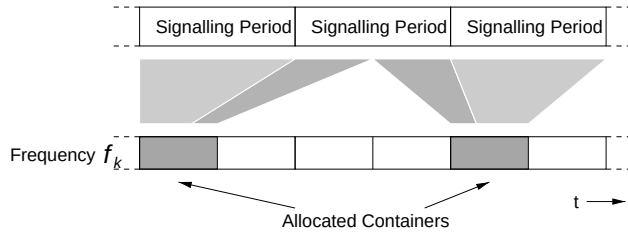


Figure 12.19: Signalling periods and container channels

12.4 Mobility Support for W-ATM Systems

The architecture of a broadband mobile radio system uses an ATM network as the transport platform (see Figure 5.1). An ATM network contains no mobility support functions such as location management of mobile users or handover support. Figure 12.20 presents a typical W-ATM protocol stack for the control plane [23]. Appropriately for an ATM fixed network, the standard protocol for user network signalling (UNIsig, [20]) is supported for call control (CC) in the *fixed control plane* (see shadowed areas). The radio access system is transparent for these protocols. Protocols for radio resource management (RM) and mobility management (MM) are supported in the *wireless control*

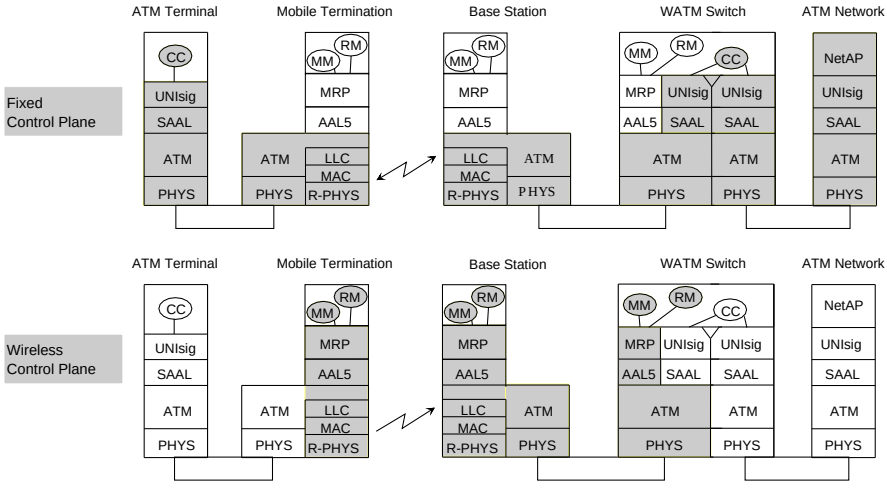


Figure 12.20: Protocol stack in wireless ATM

plane within the radio access system (*Mobility and Resource Management Protocol*, MRP).

12.4.1 Radio Handover

Radio handover takes place between two transceivers (BST) of the same base station; see Figure 12.21. The virtual connections are switched over within the *base station controller* (BSC), independently of the ATM fixed network.

A handover can be divided into three stages:

1. **Measurement of radio resources** A handover is normally executed because of poor transmission conditions and the subsequent unsatisfactory quality of service of the radio channel. The conditions can be assessed only if measurements are conducted frequently enough. It is also necessary for the effects of radio propagation to be recognized and taken into account. This necessitates measurements not only of the actual connection but also those of the alternative radio channels.

2. **Handover decision** This is the stage when it is decided whether a handover should be excecuted. The relevant algorithm uses as its criteria the measurement values provided of the radio conditions and the capacity utilization of the radio cells.

3. **Handover execution** The signalling protocol is executed at this stage in a handover to enable the relevant MAC connection to be switched from one radio cell to another.

Wireless ATM systems have the advantage that terminals and base stations are able to initiate handovers independently of one other. A terminal can

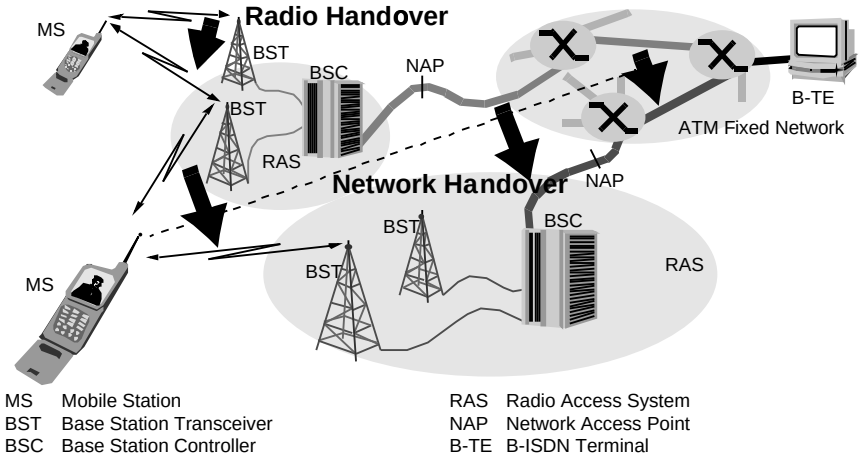


Figure 12.21: Radio and network handover in wireless ATM

initiate a handover because of receiving conditions; a base station can force a terminal to execute a handover (*forced handover*) if the receiving conditions or cell load require it.

There are two types of handover initiated by a terminal. If the handover is carried out during the actual connection, it is called a *backward handover*. If the connection has already been interrupted or is very poor, the handover signalling must be carried out over an alternative channel of the medium; this is called *forward handover*.

12.4.2 Network Handover

During each network handover, a change in route is necessary for any virtual channel connection in the fixed network that is not being supported by current ATM networks; see Figure 12.21. Different network handover concepts, such as those being discussed by the ATM Forum, are introduced below.

The simplest protocol for switching between network access points in the ATM fixed network consists of terminating the old end-to-end connection and setting up a new one. This procedure would cause a complete disruption to service, which users are hardly willing to accept, and consequently it is not discussed further.

The time overlap of an old and a new connection during network handover would confront the corresponding terminal (in the fixed network) with two simultaneous connections. At times the user data would be divided between both connections in a way that would be confusing to the corresponding terminal, and it would not be possible to receive the user data without some modification to the terminal's protocols. Because of the distance involved and the subsequent difficulty in synchronizing the data streams of both con-

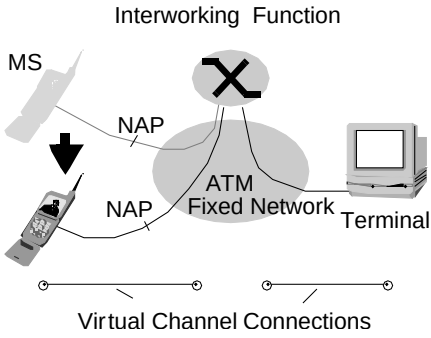


Figure 12.22: Network handover support in the AAL layer

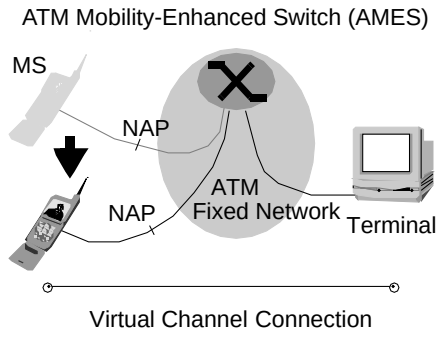


Figure 12.23: Network handover support in the ATM layer

nections, a dynamic modification of the protocols of terminals would result in a reduced quality of service. Furthermore, this kind of modification would affect all ATM terminals, and therefore is not workable.

Current mobile radio systems, such as GSM, have mobile services switching centres (MSC) that supply the functions necessary for network handover between old and new connections without any loss of information. A similar solution is also possible with ATM networks. This involves breaking down a virtual end-to-end connection into two virtual channel connections, one each from the mobile switching centre to the mobile station and to the corresponding terminal (see Figure 12.22). If during the network handover a virtual channel connection is being set up between a mobile station and the mobile radio switching centre from the new *network access point* (NAP) then this can under certain conditions through the use of a bridging function produce a *seamless* handover between the old and the new virtual channel connection at the mobile switching centre.

Virtual channel connections are set up between entities of the ATM adaptation layer (AAL); see Figure 12.10. A separation of the end-to-end connection into two virtual channel connections as shown in Figure 12.22 requires a transition function between the terminating ATM adaptation layers of the two virtual connections in an interworking function. A switching function, as well as the bridging function as one of the main components of a network handover, is then carried out in the ATM adaptation layer (AAL).

A possible alternative is leaving the switching function to the ATM layer. The actual switching is then carried out by an enhanced ATM switching centre with mobility functions which is therefore also referred to as an *ATM Mobility-Enhanced Switch* (AMES). In this case the virtual end-to-end channel connection remains intact (see Figure 12.23). Consequently only an end-to-end connection is available for call control in the network, and no interworking functions or adaptation functions are required between subnets. The role of the bridging function is more difficult to accomplish in the AMES, because

the standard stipulates asynchronous transmission and the ATM cells are not numbered. Therefore with each network handover there is the danger of ATM cells being lost.

A network handover in the ATM layer requires the rerouting of the virtual end-to-end channel connection in the VP or the VC switch within the ATM layer of the ATM fixed network.

Two typical concepts for executing a network handover are presented below. They are based on the rerouting of a virtual channel connection.

12.4.2.1 Virtual Tree

There are different suggestions for executing a network handover. For example, a distinction must be made between backward and forward handover protocols when the handover is initiated by a mobile station (see Section 12.4.1). The suitability of the network handover protocol depends mainly on the choice of the radio handover protocol.

With *backward handover* protocols, in a network handover the mobile station deregisters from the old radio access system (RAS), which consists of a BSC and several BSTs, before it changes to the new RAS. Therefore in preparation a virtual channel connection is set up to form a new radio access system over the AMES that is currently carrying out the connection. The end-to-end connection will then be switched to that system during the network handover.

With *forward handover* protocols, it is not possible for the AMES to be informed in advance. The mobile station registers over the radio interface in the new radio access system, which must then set up a new virtual channel connection to the AMES. The amount of time needed for setting up a new ATM network connection is in the region of 100 ms per ATM network element. Therefore, depending on the network topology, values of the order of seconds can accumulate. Connection setup times of this magnitude during network handover cannot be offset by the network through intermediate storage and accelerated transmission, and instead lead to an interruption of service. Depending on the handover frequency, connection setup and termination place an additional burden on an ATM fixed network because of the signalling traffic and switching involved.

A service without interruption can be guaranteed through pre-reserved virtual channel connections between all RASs and the AMES. They can then be used immediately during network handover, and produce the respective connection setup. However, each RAS must then constantly maintain reserved virtual channel connections to the AMES. This produces a so-called virtual connection tree (see Figure 12.24) for each connection from a mobile station to AMES. Every virtual branch of this tree consists of two virtual channel connections (one for each transmission direction), which in turn consists of a chain of virtual channels between ATM network elements. In Figure 12.24 the virtual branches are abstracted as a direct line.

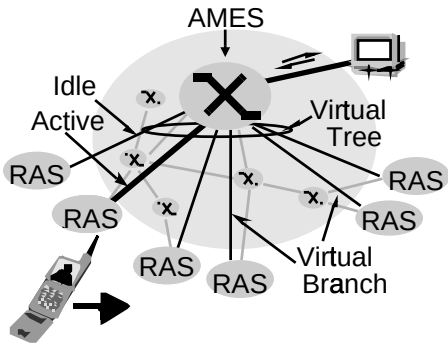


Figure 12.24: Virtual connection tree

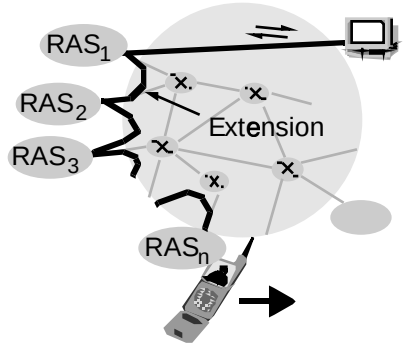


Figure 12.25: Extension of virtual channel connections

A virtual tree is set up for each call and continues to exist for the duration of an entire call. Only one virtual branch of a tree carries user data, whereas all the other virtual branches are seen as preparatory branches and remain unused until replacing a branch formerly being used. Although they are set up, i.e., all entries in the routing tables of the ATM network elements are available, they do not transmit ATM cells. In a network handover the virtual branch of the new radio access system is used, whereas the old virtual branch is idle but continues to exist.

In AMES this changeover must be supported by a switching function, which is carried out differently depending on transmission direction. There are various concepts for this [1, 36]. Virtual trees tie up additional capacity in a fixed network, because all branches must provide the quality of service required for an RAS.

12.4.2.2 Extension of Virtual Channel Connections between Radio Access Systems

Instead of the routing between different virtual branches being switched over, a virtual channel connection can be extended through the execution of a network handover that includes the extension of the virtual channel connection from the old to the new radio access system. The old radio access system then takes over the function of the AMES.

When a connection is being set up, the location area of the mobile station establishes the first radio access system as the anchor for the entire connection. If a network handover occurs, the virtual channel connection is extended bidirectionally from the anchor to the new radio access system. In each subsequent network handover another extension is added between each old RAS and the new one, thereby producing a chain of extensions such as that shown in Figure 12.25. The route can be shortened again in a network handover when the mobile station changes back to the previous RAS.

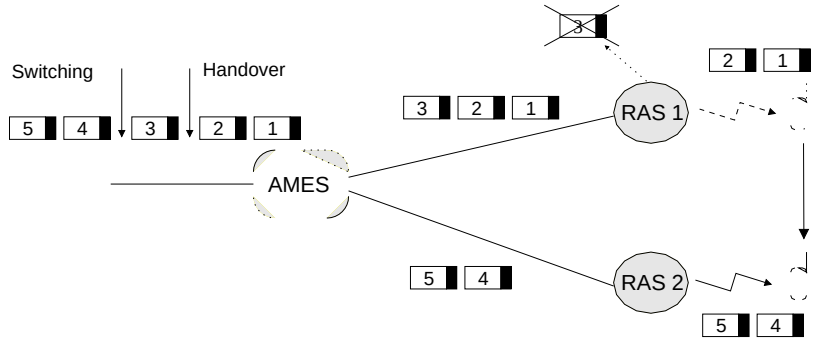


Figure 12.26: Cell loss during handover in a simple virtual tree

Here too a distinction has to be made between backward and forward handovers. In the first case the extension can be set up before the change in radio access system. In the second case the extension of the bidirectional virtual channel connection to the old radio access system must be set up after the radio handover by the new radio access system.

As discussed in Section 12.4.2.1, this kind of connection setup can result in an interruption to service, which explains why it makes sense to use reserved virtual channels between neighbouring radio access systems. Similarly to the virtual tree, as a precaution for each communicating mobile station a bidirectional virtual channel connection should be set up to each neighbouring RAS.

12.4.2.3 Guaranteed Quality of Service

For network handovers in wireless ATM systems it is essential that the quality of service for the respective ATM connection be maintained. The procedures given as examples in the previous subsections allow the execution of an accelerated network handover through the use of preset channels. These concepts on their own, however, cannot guarantee that the quality of service as agreed at connection setup is also maintainable during a network handover.

It is clear from Figure 12.26 that ATM cells in a virtual tree can be lost during a network handover. In a forward handover the AMES is notified by the new radio access system (RAS 2) that a handover has taken place. The AMES thereupon switches over the virtual channel connection to the new branch of the tree. ATM cells remaining on the old branch after a handover will be lost (cell 3 in this example). It is obvious that the number of lost cells is dependent on the data rate of the service and on the duration of the handover signalling.

Protocols that have a minimal effect on quality of service during handover are currently at the specification stage. It is generally not possible for the

quality of service of the ATM fixed network to be guaranteed in all situations in a mobility-supporting wireless ATM system.

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13
Wireless Local Area Networks*

Since the introduction of lightweight portable computers (laptops, notebooks), a great deal of attention has been focused on the development of wireless computer networks (*Wireless Local Area Network*, WLAN).

Thanks to standardization in the field of local area networks, it is comparatively easy to find systems that will still be upgradable even in a few years' time. Around 70 % of all computers connected to networks are compliant with the IEEE 802.3 (*Ethernet*) and IEEE 802.5 (*Token Ring*) standards. Connection is normally over a permanent wireline link. The problems that can occur are the surfacing of mechanical defects (corrosion) after a few years and violations of rules on radiated interference. It is difficult to adapt these networks to cope with changing office conditions. Mobile network nodes are not possible.

The obvious approach is to leave out the cable entirely. This idea is almost as old as the concept of the so-called ALOHA system, which used radio to connect terminals to their processing computers. The newer wireless LANs work with the most up-to-date radio technology. Data is encrypted and extensive error-protection mechanisms are available. Integrity of data is also guaranteed.

Just like wireline LANs, wireless LANs can be divided into different architectures and performance categories. Many companies offer products for wireless point-to-point connections, but only very few build LANs for multipoint communication. Today wireless networks use spread-spectrum, narrowband microwave or infrared signals for transmission (see Table 13.1). Because of legal regulations, networks using spread-spectrum and narrowband microwave cannot be operated in most countries unless special authorization has been given.

Until now, wireless LANs have only had a very small share of the market. This is partly due to the higher costs per network node, but no doubt also because of the late standardization in this area. In spite of this, suppliers are projecting a growth in wireless networks over the next few years. Standards such as IEEE 802.11 or HIPERLAN/1 discussed below will help to increase user acceptance of wireless LANs.

*With the collaboration of Christian Plenge and Andreas Hettich

Table 13.1: Characteristics of different transmission techniques

	Spread spectrum	Microwave	Infrared
Frequency [GHz]	1–6	18.825–19.205	30 000
Range [m]	30–250	10–50	25
Power [W]	<1	0.025	—

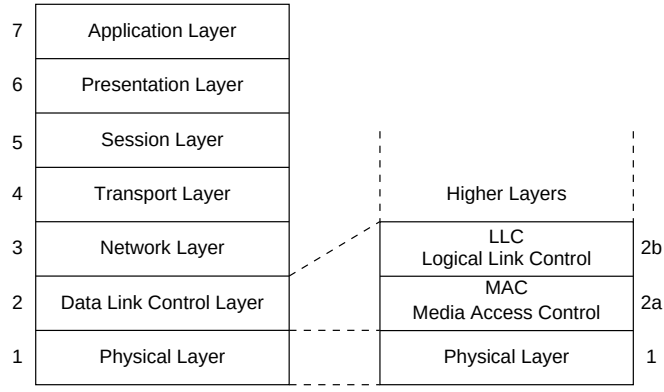


Figure 13.1: The IEEE 802/ISO 8802 standard

13.1 Standards

The diversity of LAN systems in terms of cabling, transmission techniques, transmission speeds, access procedures and variations thereof necessitated standardization in order to facilitate their acceptance and enable different LANs to work together.

Committee 802 of the *Institute of Electrical and Electronics Engineers* (IEEE) in the early 1980s had developed a standard for *Local Area Networks* (LANs) with speeds of up to 20 Mbit/s that offers security on the communications side for manufacturers as well as for users and has largely been accepted.

The standard mainly restricts itself to the lower two layers of the ISO/OSI reference model (see Figure 13.1). A separation is made between *Logical Link Control* (LLC) and *Medium-Access Control* (MAC). The LLC layer upwardly offers all systems a standard interface for establishing logical connections. The MAC sublayer supports protocols such as token ring, token bus, CSMA/CD (*Ethernet*).

In Western Europe the standards for wireless radio LANs are specified by ETSI. The technical group RES 10 (*Radio Equipment and Systems*) at ETSI has developed HIPERLAN/1, the European standard for wireless LANs [3].

Table 13.2: Mid-frequencies of HIPERLAN/1

Channel No.	Mid-frequency [MHz]	Channel No.	Mid-frequency [MHz]
0	5 176.4680	3	5 247.0562
1	5 199.9974	4	5 270.5856
2	5 223.5268		

The frequency bands at 5.2 GHz and 17.1 GHz are being reserved throughout Europe for this WLAN.

HIPERLAN Type 1 (HIPERLAN/1) describes a wireless LAN for computer–computer/terminal communication. ETSI BRAN (Broadband Radio Access Network, combining RES 10 and TM 4) also has workgroups that are developing specifications for wireless ATM systems (*Asynchronous Transfer Mode*) under the designation of HIPERLAN Type 2 (*Wireless-ATM LAN*), HIPERACCESS (*Wireless-ATM Remote Access*) and HIPERLINK (*Wireless-ATM Interconnect*). Results of their efforts are expected in 1998/99 (see Section 12.1.5).

In parallel to the European HIPERLAN/1, the IEEE in the USA has specified 802.11 as an additional standard for WLANs [5].

Both standards are equipped with an IEEE 802.2 or ISO 8802 compatible interface [6], allowing them to replace the wired transmission systems described above. Because of the restriction of the radio medium (e.g., radio range), both standards must contain functions for the management and maintenance of the radio network that far exceed the normal tasks of the MAC sublayer. These are described along with the technical characteristics of HIPERLAN/1 and IEEE 802.11 in the following sections.

13.2 Technical Characteristics of HIPERLAN/1

HIPERLAN/1 can be used as a universally accepted broadband and flexible ad hoc LAN (see Section 13.3.1) and as such can be connected to other LANs. Up to five frequency channels in the 5.15–5.30 GHz range are being provided for HIPERLAN/1; see Table 13.2.

One channel provides a bit rate of 23.5294 Mbit/s for user and control data. The data rate available to the user is reduced to 10–20 Mbit/s because of the *overhead* added by the protocols of the different sublayers and the channel-access procedure. At maximum 1 W transmitter power the range of a HIPERLAN/1 node should be around 50 m indoors.

Three transmit and receive classes with varying power and sensitivity are specified in the standard. A GMSK modulation method with a bandwidth time product of 0.3 is used on the radio channel.

Synchronous applications as well as those with real-time requirements are supported. The transmission time is not a critical factor with asynchronous traffic, e.g., with electronic mail or file transfer.

The standard contains mechanisms for encryption of sensitive data before it is sent.

HIPERLAN/1 terminals have to be small so they can be used in portable computers. Plans are underway to make them available the size of a PCMCIA card (*Personal Computer Memory Card Interface Association*) with the dimensions 85×54×10.5 mm (excluding antenna system).

Since HIPERLAN/1 systems support applications for battery-operated systems, they must offer low power consumption of a few hundred mW. HIPERLAN/1 offers an energy-saving mode.

HIPERLAN/1 networks should support the mobility of terminals. HIPERLAN/1 stations are therefore designed to be able to exchange information with other stations at up to a speed of 10 m/s, which corresponds to 36 km/h, or up to a rotational speed of 360˚/s.

13.3 Network Environments for HIPERLAN/1

The ETR 069 technical report [4] produced by the ETSI RES 10 committee defined the services and possibilities planned for HIPERLAN/1. Some of the applications that will benefit from new solutions and an overview of the HIPERLAN/1 network topologies are presented in the following [2]. HIPERLAN/1 appears without the "/1" below.

13.3.1 HIPERLAN Applications

Wireless offices A WLAN is a better option than a fixed network in listed buildings or in environments where constructional changes are so frequent that cabling cannot be installed, e.g., film and photographic studios. Furthermore, it should be possible, for example, to use portable computers in different locations and connect them easily to a network.

Ad hoc networks Ad hoc networks are radio networks without any kind of permanent communications infrastructure. A group of users can form its own closed complex. This means that at conferences, conventions, and large functions, or in the event of accidents or catastrophes, computers can communicate with each other without having been cabled together beforehand. Each user carries his part of the network with him in the form of his computer with a radio-LAN connection.

Medicine Within a radio-LAN, doctors would be able to have direct and interactive access to remote data such as X-rays when visiting their patients. This would make the work of doctors easier and produce better and faster diagnosis for patients.

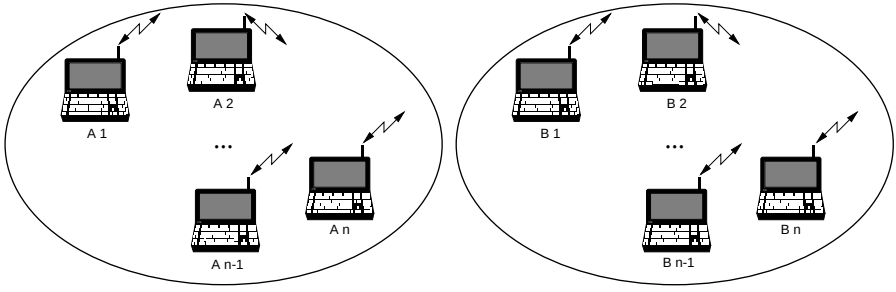


Figure 13.2: Independent HIPERLANs

Industrial applications More and more work in industry is being automated. In many cases the controlling computers are in a central facility managing a large number of machines that are restricted to use in their locations. A wireless connection to the network would allow these machines (e.g., industrial robots or unmanned vehicles) more freedom of movement and the possibility of being used with more flexibility. Maintenance personnel would use laptops to retrieve the data needed for diagnostic purposes.

13.3.2 Network Topologies

A HIPERLAN is not organized centrally, and instead has a completely distributed architecture with a dynamic allocation of network and network node identifiers. Each station (node) is differentiated from the other stations by a unique *node identifier* (`NID`). A number of stations are combined into a network with a shared HIPERLAN identifier (`HID`). This network forms a HIPERLAN.

In contrast to a wireline network, HIPERLANs using the same radio channel cannot be separated from one another. Overlapping can occur. Another problem with radio channels is the range restriction. Mobile HIPERLAN nodes and unfavourable propagation characteristics can cause the fragmentation of a network.

The following network topologies exist as a result of the channel characteristics and the applications presented:

Independent HIPERLANs Two HIPERLANs, A and B, are considered to be independent of each other if no member from HIPERLAN A is located in the transmission range of a member from network B (see Figure 13.2). Even if the same frequencies are used for transmission in both networks, it is assumed that subnetworks A and B do not share the communication medium and therefore also do not cause any interference in each other's network. HIPERLAN A could be an ad hoc network set up for

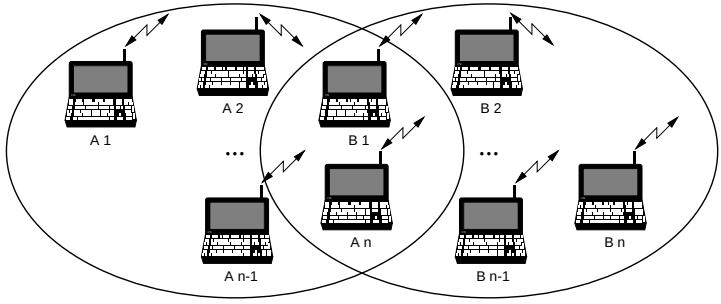


Figure 13.3: Overlapping HIPERLANs

a conference of company X, and HIPERLAN B an LAN used in the factory of company Y.

Overlapping HIPERLANs If the radio range of some of the stations in network A should overlap with some of the stations in network B then these members share the communications medium and its transmission capacity in the area where the overlapping occurs (see Figure 13.3).

An overlapping of networks produces two effects:

- The senders in the different HIPERLANs use the same frequency band, thereby increasing the occurrences of interference. As a result, optimal use of the frequency band is no longer possible because not all the stations are able to receive from each other (*hidden stations*) and therefore can cause interference.
- A station receives data packets from several HIPERLANs with different HIDs. All received data packets are evaluated, and only those with their own HIDs are accepted. As a result, there is a decrease in the maximum possible data transmission capacity and consequently also the data transmission rate in this area.

These effects can be reduced if several frequency channels are introduced. With HIPERLAN, up to five frequency channels in the 5.15–5.30 GHz range are provided.

Multihop networks In addition to their original role as transmitting and receiving stations for their own terminals, some stations in a multihop network also perform the function of relay stations. This allows data to be transmitted over larger distances despite the restricted reachable range of the radio medium. In Figure 13.4 the relay stations (*forwarders*) 2, 4 and 6 are forwarding the traffic of node 1 to destination 7.

Interworking Since most HIPERLAN applications already exist, it must be possible for HIPERLANs to be connected to the usual fixed networks

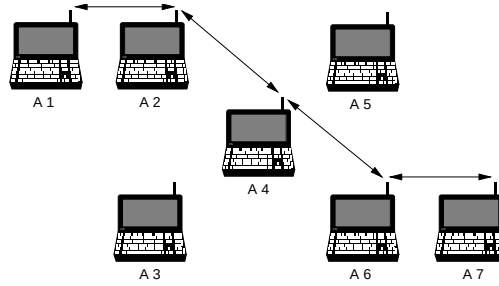


Figure 13.4: Communication in a multihop radio network

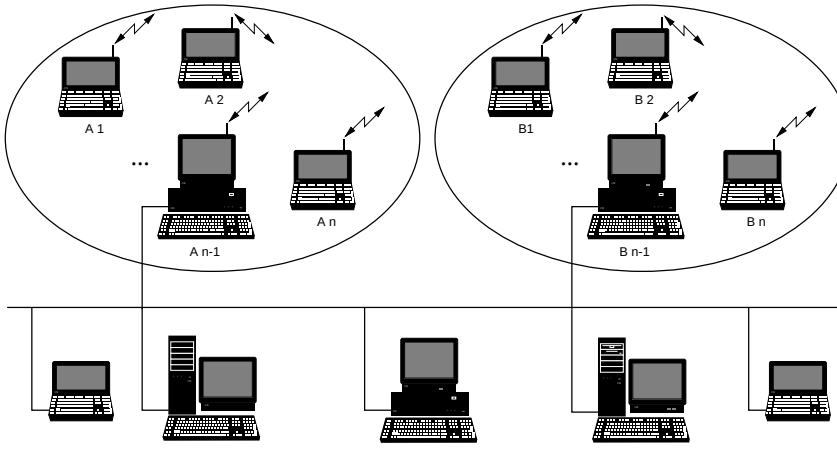


Figure 13.5: Connection to fixed network

(see Figure 13.5). This affects the network layer, and is not part of the HIPERLAN/1 standard.

13.4 HIPERLAN Reference Model

The HIPERLAN reference model defines the components needed to install a private radio subnetwork. It is based on the ISO/OSI reference model and consists of the *Medium-Access Control* (MAC) sublayer, the *Channel-Access Control* (CAC) sublayer and the physical layer; see Figure 13.6.

The organizational part of access is described in the MAC sublayer, access to the radio channel takes place in the CAC sublayer. HIPERLAN provides an ISO-8802 standard interface.

In accordance with the OSI service user/service provider model, each layer provides services for the layer above it. These services are offered at the

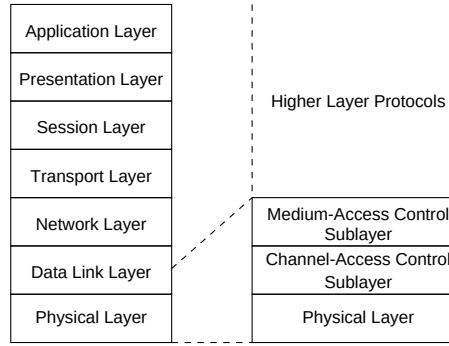


Figure 13.6: ISO/OSI and HIPERLAN reference models

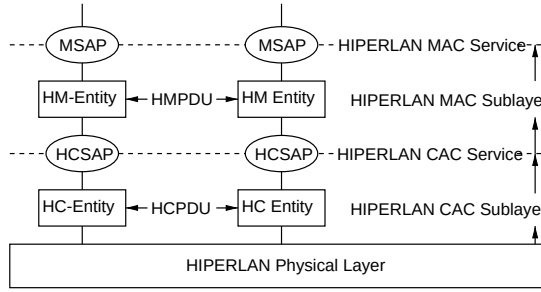


Figure 13.7: Service model for MAC and CAC sublayers

service access points between the layers, and are controlled by exchange of *service primitives* (SP).

The individual sublayers are described in more detail in the following sections. Figure 13.7 shows the HIPERLAN service model.

13.5 HIPERLAN-MAC Sublayer

13.5.1 Tasks of the MAC Sublayer

The MAC sublayer provides the following functions to ensure smooth and reliable HIPERLAN operation.

13.5.1.1 MAC Address Mapping

Because HIPERLAN stations share the radio channel, an overlapping of neighbouring HIPERLANs can occur that causes the radio ranges of a number of wireless networks to overlap in the same radio channel (see Figure 13.3).

A restricted radio range, the mobility of the stations and unfavourable propagation conditions can cause the fragmentation of a HIPERLAN (see Figure 13.2), although A and B are separate subnetworks of the same HIPERLAN.

The standard defines internal address structures. A HIPERLAN address consists of a HIPERLAN name (`HID`) and a station identification (`NID`). The `HID` is used by the MAC protocol to differentiate between the MAC communication of the individual HIPERLANs. If there is a dynamic allocation of the `HID` because of the possible overlapping of cells, it is less likely that mixed communication will occur. The MAC protocol reserves the use of special `HIDs` for communication between the stations of neighbouring HIPERLANs.

It is easier for a user to identify his network using a name rather than the numerical `HID`. The name is also used by the lookup function for establishing which HIPERLANs are operating in the area.

Since no clear guidelines and no administrative coordination exist for HIPERLAN identification, it is possible for a MAC communication to take place from non-distinguishable LANs. This situation is very unlikely, because of the MAC identification scheme used.

The address mapping function converts IEEE-MAC addressing into HIPERLAN addressing.

13.5.1.2 Communication Security

A radio channel can be listened for in a neighbouring area. An encryption algorithm with the appropriate cryptographic management is therefore provided in the MAC sublayer. This protects confidential data from unauthorized eavesdropping, and also guarantees communication security for the radio network.

The HIPERLAN encryption scheme provides for a common set of keys, one of which is used in the encryption operation. Each key has a number that is transmitted with the encrypted data to the receiver. In addition, a common initialization vector for encryption and decryption is required and if necessary transmitted.

The level of transmission security increases with the frequency of the change of key and initialization vectors.

13.5.1.3 Addressing of Service Access Points

For compatibility with the ISO-MAC service definition the MAC service (HM service) uses 48-bit LAN-MAC addresses for identification of MAC service access points (MSAP) (see Figure 13.7). The standard recognizes separate addresses for individual MAC service access points and group addresses for contacting several MSAPs.

An HM entity is linked to a single MSAP, through which it offers MAC services. It is also connected to a single *HIPERLAN-CAC Service Access Point* (HCSAP), over which it uses the HIPERLAN-CAC services.

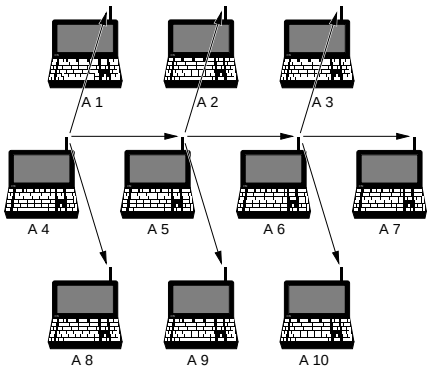


Figure 13.8: Broadcast transmission

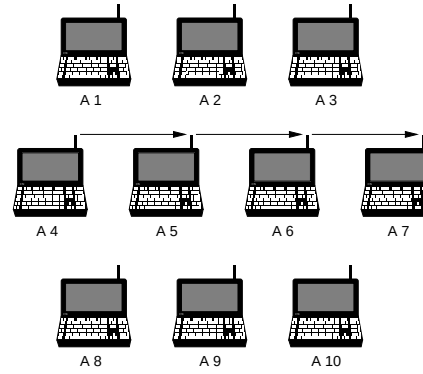


Figure 13.9: Unicast transmission

A single 48-bit LAN-MAC address is used as an MSAP address for addressing the service access point, the HM entity and the users of HIPERLAN-MAC services. An address of the same length is used to identify a group of MSAPs and the users associated with it.

13.5.1.4 Forwarding

The MAC protocol incorporates a *multihop relaying* facility that enables the transmission of data beyond the boundaries of a station's sending area—in other words over several stations. An HM entity is either a forwarder or a non-forwarder. Only forwarders carry out the forwarding of MSDUs when required. Point-to-point (*unicast*) as well as broadcast transmission (*broadcast, multicast*) is possible for the transmission of packets.

Broadcast relaying is used to relay information to all HM entities or when the transmission route is not known. Each station that recognizes a route to the destination station forwards the data packet accordingly. To avoid data packets from being routed by several stations at the same time, the protocol ensures that only a limited number of stations can forward data. Figure 13.8 shows a *broadcast* transmission being sent by station 4.

It is more efficient to forward packets that are addressed to a particular receiver if a *unicast* transmission is used (see Figure 13.9). A packet is then routed to its destination through successive *hops* in accordance with an (optimal) route. *Broadcast relaying* must be used if the route is not known, as explained above.

Each HM entity collects and manages routing information in its *Routing Information Base* (RIB). This information is continuously updated. The data in the RIB becomes obsolete and is discarded once it has passed its validity period. Because of this constant updating of routing information, it is even possible to specify quasi-optimal paths for the forwarding of packets in continuously changing HIPERLANs.

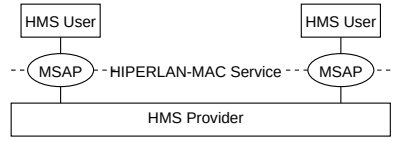


Figure 13.10: HIPERLAN-MAC service model

13.5.1.5 Power Conservation Mechanisms

The *power conservation* function of the MAC protocol provides a power-saving mechanism ensuring that battery-operated systems use a minimal amount of power. The power-saving function is an optional function. It recognizes two types of stations:

- The power-saving terminal (*P-saver*) defines time periods when it is activated. These are the only times it sends and receives data.
- The *P-supporter* only transmits data to its power-saving neighbour when the latter is activated.

13.5.2 MAC Services

As a service provider (HMS provider), the MAC sublayer provides its services (*HIPERLAN-MAC services*, HMS) to the LLC sublayer above it at the MSAP. This makes the LLC layer a HIPERLAN-MAC service user (*HMS user*) of the services offered by the MAC sublayer (see Figure 13.10).

The following is defined for the transmission of *MAC service data units* (MSDU):

- An MSDU is transmitted from the source MSAP to an individual destination MSAP or to a group of destination service access points in a single request to the MSDU transfer service. A connectionless service is provided, i.e., the transfer takes place without the need for a connection to be established explicitly and then later cleared.
- Each MSDU transmission is independent of all others.
- MSDUs are transmitted without any restriction to their content, format or coding; MSDUs are never interpreted from the standpoint of their structure or their content.
- An HMS user can indicate the quality of service desired (*HIPERLAN-MAC Quality of Service*, HMQoS) using user priority and the MSDU lifetime. Other means for influencing the HMS service provider are not planned.
- The HMS provider can carry out the following actions:

Table 13.3: Range of values for quality of service parameters

Parameter	Allowable range of values			Default value
User priority	0	...	1	1
MSDU lifetime	0	...	32767	500
Residual lifetime	0	...	32767	—

- Discard MSDUs
- Duplicate MSDUs
- Change sequence of MSDUs

The HMS service provider notifies the HMS user through an HMQoS error report that it was not possible to meet the desired quality of service for one of the previous invocations of the MSDU transfer function.

HMS users can use the HIPERLAN lookup service to determine which HIPERLANs are in their area.

13.5.2.1 MSAP Addresses

According to Section 13.5.1.3, there are two types of MSAP addresses:

- Individual MSAP addresses for the identification of an individual MSAP.
- Group MSAP addresses that address a group of MAC service access points.

Individual MSAP addresses are allowed to be used as source and destination address, but a group MSAP address can only be used to address a destination.

13.5.2.2 Quality of Service Parameters

When an MSDU is being transferred to *User Data Acceptance* (see Section 13.5.3.4), a MAC service user can indicate the quality of service parameters (see Table 13.3):

User priority (UP) The UP indicates the importance of an MDSU. The user priority controls the quality of service of time-critical services.

Lifetime of data This specifies the maximum lifetime of an MSDU that is allowed to elapse between transmission and reception.

Residual lifetime This indicates how much time is still remaining from the lifetime.

Table 13.4: HMS primitives and their parameters

Task	Service	Primitive	Parameter
Data transfer	MSDU transfer	HM_Unitdata_req	Source Address, Destination Address, MSDU, User Priority, MSDU Lifetime
		HM_Unitdata_ind	Source Address, Destination Address, MSDU, User Priority, MSDU Lifetime, Residual MSDU Lifetime
Control	HMQoS failure report	HM_Qosfailure_ind	Destination Address, User Priority, MSDU Lifetime
	HIPERLAN lookup	HM_Lookup_req	—
		HM_Lookup_conf	HIPERLAN Information

13.5.2.3 MAC Service Primitives

The HIPERLAN-MAC service primitives derived from the ISO/OSI standard represent the possible interactions between HMS user and HMS service provider. The service primitives at the HIPERLAN-MAC service access point and their parameters are listed in Table 13.4.

In addition to the quality of service parameters, the following parameters are also used:

Source Address (SA) The individual sending address of the MSAP.

Destination Address (DA) Address of an individual MSAP or a group of MSAPs to which a packet is directed.

MAC Service Data Unit (MSDU) Data that is transported without modification by the HM service provider between HMS users. The size of the packets is between zero and a value specified by the HMS provider.

HIPERLAN Information Information about other HIPERLANs in the vicinity of an HMS user.

13.5.2.4 Description of Service Primitives in MAC Sublayer

MSDU Transmission If a data packet is ready for transmission, the MAC sublayer is notified through an HM_Unitdata_Request message from the LLC sublayer. All the information required for transmission to an MSAP or to a group of MSAPs is supplied at the same time. The packet is regarded as being independent of the previous or the following packets, and is transmitted in connectionless mode. The receiver is not able to control the rate at which the sender sends the packets. Figure 13.11 shows the primitives for a successful transfer.

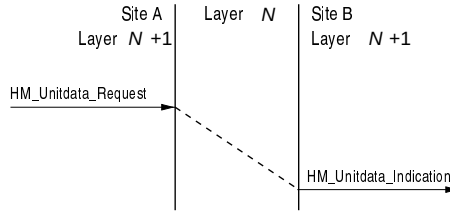


Figure 13.11: Sequence of primitives in successful data transfer

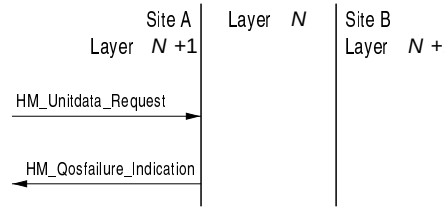


Figure 13.12: Sequence of primitives in a detected failed data transfer

The LLC sublayer does not receive an acknowledgement indicating whether the data packet has been successfully transmitted within the lifetime of the MSDU.

HMQoS Error Report If an MSDU cannot be transmitted with the quality of service prescribed by the user until the lifetime of the data is eliminated, the MAC notifies the LLC sublayer through sending an HM_Qosfailure_Indication that the transmission has failed. The MAC sublayer then discards the MSDU.

The error report gives the HMS user the chance to repeat the transmission with adjusted quality of service parameters; see Figure 13.12.

13.5.3 HIPERLAN-MAC Protocol

Figure 13.13 presents an overview of the elements of the MAC sublayer, which will be discussed in detail below.

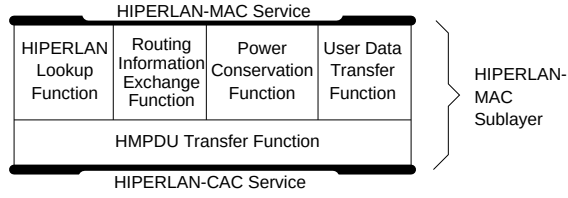


Figure 13.13: Structure of HIPERLAN-MAC sublayer

The MAC protocol, which is a prerequisite for the services offered by the *HIPERLAN Channel-Access Control* sublayer and introduced in Section 13.6.2, is used in the MAC sublayer for communication between HM entities. For the communication, *HIPERLAN-MAC protocol data units* (HMPDU) transmitted over an HC connection are exchanged between the HM entities. For detection of the PDUs already received, each packet is assigned a sequence number from the range 0–65 535.

Initiated by the HMS user—in other words spontaneously (for management)—HMPDUs are created within the MAC sublayer. All HMPDUs are collected in a queue, where they remain until they are transmitted or, because of the expiration of their lifetime, are discarded.

The HIPERLAN-MAC protocol provides for *service data units* (HMSDU) with maximum size 2385-byte length.

13.5.3.1 HIPERLAN Lookup Function

The lookup function is used to establish the communications environment of a HIPERLAN. The following subfunctions are supported for this purpose:

Find A HIPERLAN known by its name can be found if the HIPERLAN identifier (`HID`) associated with this name is established.

Create For the HIPERLAN name indicated, an `HID` that is not being used in the current communications environment is selected for creating a new HIPERLAN.

Destroy A HIPERLAN is implicitly destroyed if there is no user using its `HID`.

Join A user implicitly joins a HIPERLAN by calling up its `HID` and the *encryption* and *decryption* key used and sending his subsequent data packets using this `HID`.

Leave A user implicitly leaves a HIPERLAN when he no longer uses its `HID` in his data packets and evaluates packets with this `HID`.

The lookup function is invoked after receipt of an HM_Lookup_Request service primitive. It generates an LR-HMPDU addressed to all HIPERLAN neighbours. A timer ($\mathtt{T_{lrr}}$) is started, and all responses from neighbouring stations must arrive before it runs out (see Table 13.16).

All receivers of the LR-HMPDU create an LC-HMPDU addressed to the sender of the LR-HMPDU. The HIPERLAN name and the `HID` of the HIPERLAN to which the station belongs are entered in this packet.

The information of all LC-HMPDUs which arrive at the HM entity before the timer runs out is filed in a table. An HM_Lookup_Confirm service primitive is used for forwarding this data to the HMS user making the inquiry. LC-HMPDUs that arrive after the $\mathtt{T_{lrr}}$ timer runs out are discarded.

13.5.3.2 Routing Information Exchange Function

The *Routing Information Exchange Function* enables HM entities to establish a *routing information base* (RIB). The RIB contains lists in which neighbouring relationships between two HM entities and information about individual HM entities are administered. A differentiation is made between the following data records:

- A *neighbour tuple* (N-tuple) with the MSAP address of a neighbour and the neighbouring relationship to this station. All N-tuples together form a *neighbour table* (N-table). N-tuples that contain neighbouring forwarders are combined into a *multipoint relay table* (MRT). Whenever there is a change to this list, the allocated sequence number (MRSC-SN) is increased.
- *Source multipoint relay tuples* (SMR tuples) contain the MSAP address and the MRSC-SN of a neighbour that is using this station as a forwarder. All SMR tuples are combined into a *source multipoint relay table* (SMR table).
- A *topology tuple* (T-tuple) contains the MSAP address of an HM entity in the network, the corresponding MRSC-SN and the MSAP address of the last hop on the route to this station. The corresponding list is the *topology table* (T-table).
- The *routing tuple* (R-tuple) indicates to an HM entity the MSAP address of the next station on the route to the HM entity and the total number of hops required. All R-tuples together form a *routing table* (R-table).

The forwarder administers all the lists mentioned above for all known stations in the HIPERLAN; a non-forwarder only needs the SMR table.

The *Routing Information Exchange Function* allows an exchange of information that enables communication beyond the boundaries of the radio area. The procedures presented in the following sections enable stations to ascertain their immediate neighbourhood and the configuration of the HIPERLAN.

Hello Procedure The Hello procedure is called up at regular intervals ($\mathtt{T_{hr}}$) by the MAC sublayer for creating the *neighbour table*, the *multipoint relay table* and the *source multipoint relay table*; see Table 13.16. In this connection all N-tuples are entered in an H-MPDU, which is then sent to all neighbours. The sequence number of the user's own *multipoint relay table* along with the station type (forwarder/non-forwarder) is also transmitted.

HM entities that receive an H-HMPDU adapt their MRT and N and SMR tables according to the data contained in the information field. If an N-tuple does not yet exist for the sender, one is produced. If the sender uses this HM entity as a forwarder then an SMR tuple is created. A timer with $\mathtt{T_{hr}}$ is started for all data records that have been produced or changed, and when it runs out, the expired N-tuples or SMR tuples are discarded (*aging*).

Topology Control The *topology control* procedure allows the exchange of information for the topology of the HIPERLAN. A TC-HMPDU is created by an HM entity at fixed times (every $\mathtt{T_{tcr}}$) and whenever there has been a change to the SMR table. It contains a list of the SMR tuples and corresponding sequence number. The packet is sent to all HM entities, and should also be forwarded by them. A timer with $\mathtt{T_{tcr}}$ is started for all the list elements and the appropriate tuple is destroyed after the timer has run out.

The receiver of a TC-HMPDU creates a T-tuple for each of the SMR tuples contained in it or restores T-tuples that already exist. Because of the existing T-tuples, the routing table is recalculated after any changes to the T-table. If the maximum number of hops has not yet been reached and an R-tuple, in other words a path, exists for the creator of the TC-HMPDU, the TC-HMPDU is then forwarded with an increased *hop* timer.

Multipoint Relay Determination The MRT, in other words the subset of the N-table that contains the forwarders important to this station is recalculated at regular intervals (every $\mathtt{T_{hr}}$). If the MRT has changed, the corresponding sequence number is increased and the *topology control* procedure is invoked.

13.5.3.3 Power Conservation Function

To save power, an HM entity (*P-saver*) is allowed to participate in the communication process only at set times. The *Power Conservation Function* provides the means for notifying other stations of the active and inactive periods. Certain stations serve as *P-supporters* whereby they delay the messages for particular HM entities (unicast traffic). The supported stations must be entered on the P-saver list administered by the P-supporter.

Wake Pattern Declaration Before a station initiates its power-saving operations, it sends a *Wake Pattern Declaration* packet (WPD-HMPDU). On the packet it enters the time of the last wake state, the time interval between two wake states and the duration of the state. This packet is transmitted to all neighbouring stations.

After a P-supporter has received the packet, it adds the station with the necessary parameters to the list of supported P-savers; in other words, it adjusts the values. Each entry only has the lifetime $\mathtt{T_{wp}}$, and is then discarded.

Deferred Multicast Pattern Declaration (DMPD) To allow the deferred reception also of broadcast traffic, the P-supporters define time intervals by which they delay all received broadcast messages, thereby also allowing the deferred reception of broadcast traffic. P-savers can participate through an appropriate selection of their wake states.

For coordination of the broadcast traffic the P-supporter transmits a ZDMPD-HMPDU at fixed intervals and before every deferred transmission.

It contains the time since the last transmission window, the time interval between two areas, and the duration of the state. This packet is sent to all neighbouring stations.

After receipt of the DMPD packet, *P-savers* who are using the transmitting times of the sender in order to receive multicast traffic adjust the duration and the time for their next active state according to the values supplied.

A P-saver can also define additional wake states in order to participate in the broadcast traffic. This involves a new entry in the corresponding list. The validity of the entry expires after the time T_{dmp}.

13.5.3.4 User Data Transfer Function

The *User Data Transfer Function* undertakes the transmission of data between two HMS users on the basis of the specified quality of service parameters and in accordance with the existing routing information.

The MSDU is normally transferred from the HMS user to the HM entity. Packed in a DT-HMPDU, the packet is passed from HM entity to HM entity until it has reached its destination or the preset parameters (HMQoS) can no longer be maintained. The destination HM entity forwards the MSDU to its HMS users.

User Data Acceptance After an HM_Unitdata_Request has been made by an HMS user, the *User Data Acceptance* function must decide whether it can carry out the transmission with the quality of service requested. It either accepts the MSDU or returns an error message. The criteria for the decision could be average channel delays or other parameters which are not established in the standard.

A DT-HMPDU is produced from the accepted MSDU and the quality-of-service parameters (lifetime and user priority). The type of the DT-HMPDU is determined through the destination address using the *alias resolution* procedure. Four different types are possible, DT0 (default type) to DT3, which differ in the number of address fields they contain. The additional fields contain alias addresses for MSAPs which are required for routing the DT-HMPDUs (see also *alias resolution*). Parameters and timers for the PDUs can be found in Section 13.8.

The routing function is called to forward the next HM entity on the route to its destination.

HMQoS Error Report After the lifetime of one of these HMS users has expired, the corresponding packet is discarded. The HMS user then receives the appropriate error report.

User Data Delivery This function delivers received packets that are directed to this HMS user. The MSDU is obtained from the DT-HMPDU and delivered

Table 13.5: Selection of DT-HMPDU on basis of alias addresses

Alias address known for		HMPDU type	Number and type
SA	DA	generated	of address fields
—	—	DT0	2 (SA, DA)
×	—	DT1	3 (SA, ASA, DA)
—	×	DT2	3 (SA, DA, ADA)
×	×	DT3	4 (SA, ASA, DA, ADA)

with details about the quality of service achieved to the HMS user in an HM_Unitdata_Indication primitive over the MAC service access point.

Alias Resolution The *alias resolution* function resolves the mapping that may occur between alias addresses and MSAP addresses. This function is necessary for routing DT-HMPDUs, generates DT-HMPDUs of the corresponding type (DT0 to DT3) and sets the corresponding address fields.

Alias resolution is initiated by the user data acceptance function and the forwarding function before the routing procedure determines the path of a packet.

The alias table is searched for alias addresses for the source (SA) and destination (DA) addresses. The *alias source address* (ASA) and the *alias destination address* (ADA) are used to create new DT-HMPDUs with appropriately set address fields. Table 13.5 shows which DT-HMPDU is selected depending on the alias address.

The alias learn function is invoked after each receipt of a DT-HMPDU of the type DT1, DT2 or DT3 (these are the only ones that contain alias addresses) for compiling and updating the alias tables. This function enters alias combinations that are not yet known into its tables. All elements of the alias table are subject to aging. They are discarded after expiration of the time T_{ah}.

Routing Based on the routing tuples, the routing function establishes the address of the next HCSAP in order to reach the next HM entity as well as the number of hops remaining on the path to the destination HM entity. This function is invoked by the user data acceptance function after an HM_Unitdata_Request and by the forwarding function. If no routing information is available for the destination, the address of a neighbouring forwarder is selected. If this also is not known, the reserved address for all adjacent stations is used.

Forwarding If a DT-HMPDU that is not addressed to this HM entity is received, the packet must be forwarded by a forwarder. The forwarding function only accepts packets from a unicast transmission when the number of hops

Table 13.6: Mapping quality of service parameters to channel-access priority

	Channel-access priority	
NRML	High UP	Normal UP
NRML < 10 ms	0	1
10 ms ≤ NRML < 20 ms	1	2
20 ms ≤ NRML < 40 ms	2	3
40 ms ≤ NRML < 80 ms	3	4
80 ms ≤ NRML	4	4

does not exceed the maximum value and the last sender is entered in the SMR table. The routing function is used to calculate the next hop and the remaining number of hops for the packet.

13.5.3.5 HMPDU Transfer Function

The HMPDU transfer function sends MAC protocol data units between HM entities. At the start of transmission, the data packet with the highest channel-access priority is determined by the HMPDU transfer function to enable the respective service to be provided with the quality of service requested.

The *normalized residual MSDU lifetime*, NRML is calculated from the *residual lifetime* of the HMPDUs and the number of hops still required to reach the receiver.

$$\texttt{NRML} = \frac{residual\ lifetime}{number\ of\ hops} \tag{13.1}$$

Channel-access priority is then generated from the NRML and the user priority UP as per Table 13.6 and is updated during each transmitting cycle.

HMPDU Selection This function selects the next packet to be transmitted from the HMPDU queue according to the following rules:

- Select HMPDU(s) with the highest channel-access priority.
- From these select the one(s) with the lowest NRML.
- And then select any one of them.

Data where the residual lifetime has expired is not transmitted.

DT-HMPDU Encryption The content of the DT-HMPDU can be encrypted through the selection of a key from the existing set for the coding of data before transmission by the HMPDU transmission function while an HCSDU (HIPERLAN-CAM service data unit) is being generated. The number of the key used and the initialization vector are entered in the PDU.

Taking the key number and the initialization vector, the receiver of a coded DT-HMPDU can then decrypt the packet.

Checksums All PDUs are provided with a CRC32 checksum for the detection of transmission errors.

HMPDU Transmission An HM entity can forward an HMPDU to the CAC service provider for transmission whenever an HC_Sync_Indication primitive exists or anytime after an HC_Free_Indication primitive (see Section 13.6.2.3). This involves selecting the packet with the greatest penetration, per the procedure described above. Through an HC_Unitdata_Request primitive, the HMPDU is transferred with the necessary parameters to the CAC sublayer at the *HIPERLAN-CAC Service Access Point* (HCSAP). Without any changes, the HMPDU becomes the HCSDU. The end of the transmission is indicated by HC_Status_Indication.

Upon receipt of a HC_Data_Indication primitive, the receiving HM entity must accept the HCSDU received as an HMPDU and process it according to its type or its destination.

13.6 HIPERLAN-CAC Sublayer

13.6.1 Tasks of CAC Sublayer

Channel-access takes place in the *Channel-Access Control* (CAC) sublayer through a *Channel-Access Mechanism* (CAM). It is here that it is decided during different phases which station has access to the radio channel to enable it to send asynchronous or time-critical traffic over the physical layer to the receiver. The CAC sublayer contains the functions described below.

13.6.1.1 Selection of HIPERLAN Radio Channel

Although the CAC protocol for multiple radio channel operation is defined, channel selection is not specified in the standard.

13.6.1.2 Channel-Access Procedure NPMA

The MAC protocol expects the CAC sublayer with the CAM to provide a hierarchical independent channel-access mechanism for the support of time-critical traffic.

An expanded *Non-Preemptive Priority Multiple-Access* procedure (NPMA) is used for HIPERLAN. This ensures that traffic with high priority is not interrupted by traffic with a lower priority.

The NPMA procedure works in *channel-access cycles* on the radio channel. An access cycle follows the preceding cycle or runs any time a channel is considered to be available. Each transmission attempt is carried out according

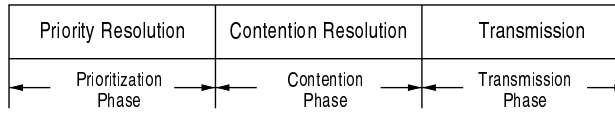


Figure 13.14: Schematic sequence of an NPMA channel-access cycle

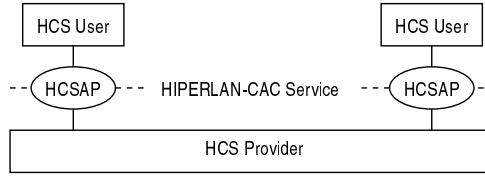


Figure 13.15: HIPERLAN-CAC service model

to access priority, and is synchronized with the access cycles currently running in the channel.

NPMA channel-access takes place in three phases as shown in Figure 13.14: *priority phase*, *contention phase* and *transmission phase*.

During the priority phase, all stations that do not currently have the highest priority are singled out. The access procedure is not interruptible, i.e, only those stations that are ready at the beginning of the cycle are allowed to attempt transmission. Otherwise they have to wait for the next cycle.

The priority phase is followed by the contention phase, during which only those stations with sufficiently high priority participate. The procedure ensures that each station statistically has the same chance for a channel. HIPERLAN uses a two-tier contention phase.

All stations that were not eliminated then send their data packets. A collision will occur if too many stations start to send at the same time.

13.6.2 CAC Services

As a HIPERLAN-CAC service provider (HCS provider) the CAC sublayer provides its services (*HIPERLAN-CAC services* HCS) to the MAC sublayer above it at the CAC service access point. In this case the MAC sublayer is a HIPERLAN-CAC user (*HCS user*) of the services offered by the CAC sublayer. A model illustrating this relationship is shown in Figure 13.15.

The following is defined for the transmission of HIPERLAN-CAC service data units (HCSDUs):

- An HCSDU is transmitted from the source HCSAP to an individual destination HCSAP or to a group of destination service access points in a single request of the HCSDU transfer service. The MAC service user can identify himself and submit one or more destination HCSAPs. The

Table 13.7: Possible values for channel-access priority

Parameter	Range of values	Default value
Channel-access priority	0 . . . 4	4

service is a connectionless one, i.e., the transfer does not require that an explicit connection be set up and then terminated again later.

- Each HCSDU transmission is independent of all the other ones.
- Activation of the HCSDU transfer service is controlled by the HC provider according to the NPMA channel-access procedure.
- Each activation of the HCSDU transfer service is followed by an indication of whether or not the transmission was successful.
- An HCS user can provide the desired channel-access priority for the transmission. Aside from these parameters the user has no influence on the means used by the HCS service provider for the transmission.
- HCSDUs are transmitted without any restrictions in terms of content, format or coding; HCSDUs are never interpreted from the standpoint of their structure or content.

13.6.2.1 HCSAP Addresses

A HIPERLAN-CAC entity (HC entity) offers its services to a single HCS user over a HIPERLAN-CAC service access point (HCSAP).

The result is that each HC entity is linked exactly to an HM entity. For practical reasons the same individual 48-bit LAN-MAC address as for the MSAP is used in addressing the HCSAP, the HC entity and the HCS user.

Similarly, the MSAP group address is used in the identification of a group of HCSAPs and their service users.

13.6.2.2 HC Quality-of-Service Parameters

The only quality-of-service parameter known to the CAC service is channel-access priority. This priority, which is defined in the NPMA, indicates the relative importance of the CAC service data unit during access to a shared channel (see Table 13.7). Numerically smaller values indicate a higher channel-access priority.

13.6.2.3 HIPERLAN-CAC Service Primitives

Table 13.8 presents an overview of the service primitives that are possible at the CAC service access point. The parameters required for transmission are listed in the last column for each primitive.

Table 13.8: HCS primitives and their parameters

Function	Service	Primitive	Parameter
Data transfer	HCSDU transfer	HC_Unitdata_Request	(Source Address, Destination Address HCSDU, HIPERLAN Identifier, Channel Access Priority
		HC_Unitdata_Indication	(Source Address, Destination Address HCSDU, HIPERLAN Identifier)
Control	Transfer control	HC_Sync_Indication	—
		HC_Free_Indication	—
		HC_Status_Indication	(Transfer Status)

Table 13.9: Transmission status

Parameter	Value	Significance
Transfer status	*Transfer Successful*	The HCSDU transmission was successful.
	Transfer Unsuccessful	The HCSDU transmission was unsuccessful. This status is also used when the HCS provider is no longer prepared to carry out an HCSDU transmission.

The following parameters are used in addition to channel-access priority:

Source address Individual address of the HCSAP sender.

Destination address Address of an individual HCSAP or a group of HCSAPs to whom a packet is directed.

CAC service data unit Data not requiring modification by the HC service provider that is transported between HCS users.

HIPERLAN identifier HID of destination HIPERLAN.

Transfer status Table 13.9 shows the possible status for the acknowledgement of transmission by the HC_Status_Indication primitive.

13.6.2.4 Description of CAC Sublayer Service Primitives

Activation of the HCSDU transfer function by an HCS user is basically coordinated by the *transfer control* function of the sublayer. This function is used to notify the MAC sublayer when the CAC sublayer is ready to accept an HCSDU to commence a channel-access cycle.

The HC service provider differentiates between two types of HCSDU transfer. In a synchronized HCSDU transfer the transmissions directly follow each

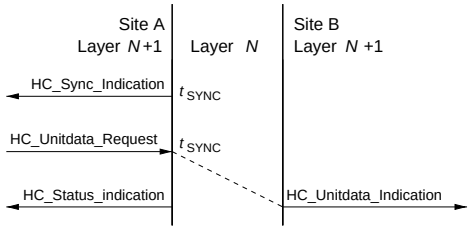


Figure 13.16: Sequence of primitives with successful transmission over a synchronized channel

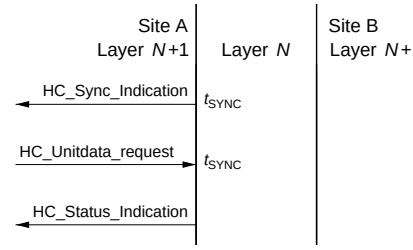


Figure 13.17: Sequence of primitives with unsuccessful transmission over a synchronized channel

other. In this situation channel-access may only commence directly after transmission of the current packet has ended. This is necessary to ensure that the NPMA channel-access procedure is functioning correctly. If a channel has not been used for a longer period of time, transmission can begin immediately after the HCSDU request. There is no synchronization with other transfers.

Every successful or unsuccessful transmission is acknowledged by the CAC sublayer.

Synchronized HCSDU Transfer If there is only synchronized traffic allowed, the CAC sublayer notifies the MAC sublayer accordingly through an HC_Sync_Indication when the new channel-access cycle (t_{SYNC}) begins. This signals to the HCS user that the CAC sublayer can accept data for the following access cycle on the radio channel.

If the MAC sublayer has HCSDUs available for transmission, it must, as soon as it receives the HC_Unitdata_Indication message, immediately create an HC_Unitdata_Request primitive with the packet being sent to initiate the HCSDU transfer. The HCSDU transfer function is not allowed to be invoked at any other time. Figure 13.16 illustrates the sequence of primitives for a successful transmission for synchronized access of the channel.

Owing to the channel-access mechanism, it is possible that an HCSDU could not be transmitted because another station seized the channel or because a collision occurred. This results in a negative acknowledgement of the transfer request (see Figure 13.17).

HCSDU Transfer with Idle Channel Channel usage is not always so high that the CAC sublayer has to synchronize itself. If the radio channel is `Idle` for a longer period of time, the HCS provider notifies the MAC sublayer accordingly through HC_Free_Indication. The CAC sublayer thereby indicates that the HCS user can invoke the HCSDU transfer function at any time until further notice. If there is a change to the channel status, the HCS user is notified through an HC_Status_Indication.

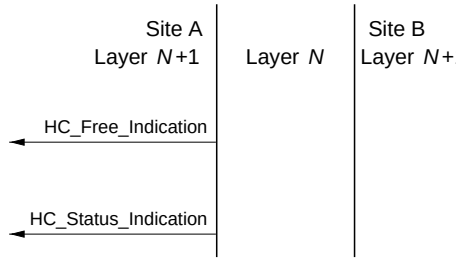


Figure 13.18: Sequence of primitives for ready-to-transmit state when channel is idle

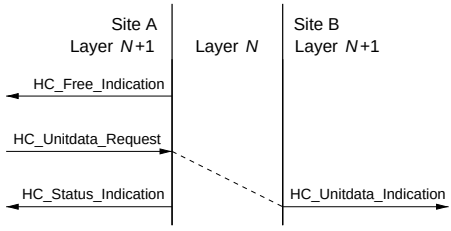


Figure 13.19: Sequence of primitives for successful transmission with an idle channel

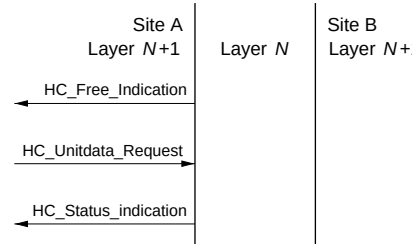


Figure 13.20: Sequence of primitives for unsuccessful transmission with an idle channel

Figure 13.18 illustrates a case when there are no HCSDUs waiting for transmission in the MAC sublayer while a channel is idle. A request for the HCSDU transfer function is not allowed once an HC_Status_Indication primitive has been received.

If the MAC sublayer receives or generates an HCSDU during the time in which the channel is regarded as idle, it can send it for transmission to the CAC sublayer by invoking the HCSDU transfer function. On the other hand, the transfer function can fail. The sequence of service primitives for successful and unsuccessful HCSDU transfer is presented in Figures 13.19 and 13.20.

In each case the HCS user receives an acknowledgement of the HC_Unitdata_Request call through an HC_Status_Indication primitive.

13.6.3 HIPERLAN-CAC Protocol

The CAC protocol is used for communication between two HC entities. It supports the services offered to the HCS user and is described in Section 13.6.2. The HC protocol provides for the transport of HCSDUs with a maximum length of 2422 bytes. HC (HIPERLAN-CAC) protocol data units (HCPDUs) are exchanged between the HC entities and transmitted over a connection in the physical layer (see Table 13.10).

Table 13.10: HIPERLAN-CAC protocol data units

PDU Name	PDU Type	Significance
AK-HCPDU	LBR-HCPDU	Acknowledgement
CP-HCPDU	LBR-HBR-HCPDU	Channel release
DT-HCPDU	LBR-HBR-HCPDU	Data transmission

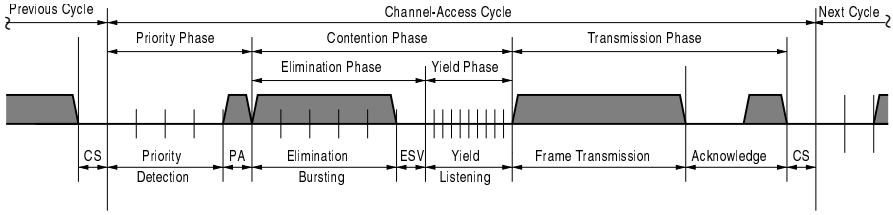


Figure 13.21: Channel activities with EY-NPMA procedure

The structure of the HCPDU can be in two parts: one that is transmitted at a low bit rate (*low-bit-rate part*, LBR part) and one that is transmitted at a high bit rate (*high-bit-rate part*, HBR part). The HBR part always consists of 1–47 blocks, each with 52 bytes. Because of their structure, two types of HCPDUs are possible:

- LBR-HCPDU, which only contains an LBR part
- LBR-HBR-HCPDU, which has an LBR and an HBR part

Transmission of an HCPDU over a channel is described in Section 13.7. The procedures required for the protocol are explained below.

13.6.3.1 EY-NPMA Channel-Access Procedure

ETSI RES10 specified *Elimination Yield–Non-Preemptive Priority Multiple Access* (EY-NPMA) as the channel access procedure for the HIPERLAN standard. With this procedure the collision rate is largely unaffected by the number of stations competing for a channel. It is based on the NPMA procedure discussed in Section 13.6.1.2. As an extension of NPMA, a combination of an *elimination phase* and a *yield phase* is used for the contention phase.

During the elimination phase, as many stations as possible should lose their right to channel-access—but not all of them. The remaining stations compete one more time during the yield phase.

Figure 13.21 shows the phases and channel activities for an EY-NPMA channel-access mechanism. The following sections describe the separate phases of EY-NPMA channel-access; the constants for the parameters defined in the HIPERLAN standard can be found in Section 13.8.

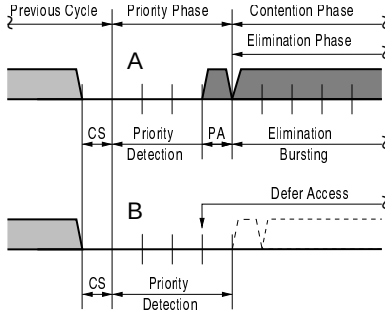


Figure 13.22: Priority phase with EY-NPMA procedure

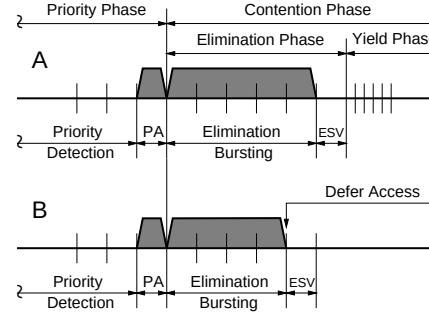


Figure 13.23: Elimination phase with EY-NPMA procedure

Priority Phase All stations ready to transmit wait for the start of the next cycle that follows the channel-access synchronization (CS) of the previous cycle, or they wait until the channel is free and then commence with the priority phase. The *priority phase* is $1 - m_{\mathrm{CAP}}$ *prioritization slots* long. If the packet to be transmitted has channel-access priority n, the station detects the channel for the duration of $n - 1$ priority slots (*priority detection*). If the radio channel is free at this time, the station immediately sends a burst (*priority assertion*, PA). However, if a burst has been sent by another station (A) before the $n - 1$ *prioritization slots* have finished, station (B) ends the access cycle immediately and waits for the next cycle (see Figure 13.22). The priority phase ends with the transmitted burst PA.

The priority phase is followed by a contention phase during which a selection of stations takes place if a number of stations want to transmit data packets with the same priority. The decision is made about which stations are allowed to transmit their packets over the radio channel. The contention phase consists of an elimination and a yield phase.

Elimination Phase The *elimination phase* (EP) is between 0 and m_{ES} *elimination slots* (ES) long. Each station sends an *elimination burst* (EB) of length $0 - m_{\mathrm{ES}}$ slots. The probability that the burst will continue into the next slot is p_E. The probability that a station will send its burst for the duration of n ES is derived from the geometric distribution:

$$p_E(n) = \begin{cases} (p_E)^n(1-p_E), & 0 \le n < m_{ES} \\ (p_E)^{m_{ES}}, & n = m_{ES} \end{cases} \tag{13.2}$$

At the end of its last burst, the station observes the channel for the duration of an *elimination survival verification interval* (ESV). If the channel is idle during this period, the station (A) then continues with the yield phase. If, however, EBs of other stations are still being transmitted on the channel, it terminates (B) the cycle as shown in Figure 13.23.

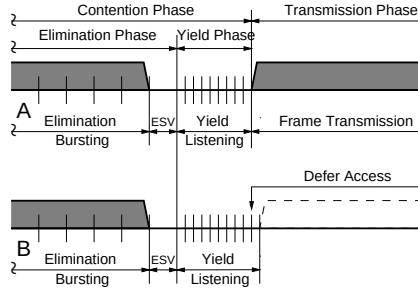


Figure 13.24: Yield phase with EY-NPMA procedure

Yield Phase The *yield phase* is between 0 and m_{YS} *yield slots* (YS) long. Each station listens in on the radio channel for the duration of $0 - m_{YS}$ slots, with p_Y indicating the probability that a station will also hear a slot sequence in the next slot. The probability of a station observing the channel for the duration of n *yield slots* is geometrically distributed:

$$p_Y(n) = \begin{cases} (p_Y)^n(1-p_Y), & 0 \leq n < m_{YS} \\ (p_Y)^{m_{YS}}, & n = m_{YS} \end{cases} \tag{13.3}$$

If no other transmission was detected during the n yield slots of a station, the channel is interpreted as being free. This means that the station has come through the contention phase and immediately begins to transmit its data.

Stations that detect transmission by another station during the yield phase are eliminated. They are not allowed to attempt another access to the channel until the next cycle.

Figure 13.24 shows two stations A and B trying to occupy a channel with A succeeding in obtaining the channel and B having to delay its transmission.

Transmission Phase The following types of transmissions can occur between HC entities during the *transmission phase*:

- CP-HCPDU transmission
- Multicast-DT-HMPDU transmission
- A unicast-DT-HMPDU transmission, followed by an AK-HCPDU transmission as an acknowledgement. The acknowledgement must commence at the latest after the duration i_{ACKW} at the end of the DT transmission (see Table 13.15).

Conditions for Channel-Access Channel-access is allowed to take place under the following two conditions:

- Channel-access when there is a free channel. In this case the channel must be detected as being idle for a minimum duration of i_{IDLE}.

- Synchronized channel-access. If a channel is not free, synchronization is carried out at the end of the current transmission. A full EY-NPMA channel-access cycle commences after a pause of the duration i_{CS} (cycle synchronization).

13.6.3.2 LBR-Part Checksum Computation

Different parts of the LBR part of an HCPDU are provided with a 4-bit checksum during the generation of a packet. An error detected in the checksum test in an LBR-HCPDU when the packet is received will cause the entire packet to be discarded. In an LBR-HBR-HCPDU the information is also contained in the HBR part, so the packet is not invalid.

13.6.3.3 HBR-Part Checksum Computation

Like the LBR part, the HBR part is also protected through error detection by the means of checksums. A 32-bit checksum is generated over the entire HBR part when an HCPDU is created. When received, the packet is discarded if an error has occurred in the checksum.

13.6.3.4 Channel Release Function

The HIPERLAN standard stipulates the five frequencies listed in Table 13.2. Channels 0, 1 and 2 are the normal channels. The use of the remaining channels is specified by national authorities. If use of channels 3 and 4 is permitted then they are released over a CP-HCPDU.

The CP-HCPDU is produced by authorized stations to notify neighbouring stations of the release of channels 3 and 4. The receiving station evaluates the packet and, depending on the entries, either releases the channels or blocks them again. The validity of this release is checked regularly on a timed basis. Channels 3 and 4 are excluded from use once the timer runs out.

13.6.3.5 User Data Transfer

The *User Data Transfer Function* is responsible for transporting data between two HCS users in accordance with the CAC service definition. HCSDUs are transferred by the HCS user (of the MAC sublayer) with channel-access priority for transmission to an HCS user or group of users. The transfer function should only be invoked if the preceding request has been acknowledged. A multicast transmission is never acknowledged, and therefore its transfer is considered successful.

The procedures of the transfer function are described in the following sections.

Transfer Invitation The transfer function has two methods for indicating to the MAC sublayer that transmission is possible.

The HC_Sync_Indication primitive notifies an HCS user that the next channel-access cycle is now commencing and he can initiate a user data transfer.

The HCS user learns from the HC_Free_Indication primitive that he can now invoke the user data transfer function at any time. This state is ended again by means of an HC_Status_Indication primitive with the content *Transfer unsuccessful*.

If an HCD user has called in the transfer function even though he has not received an invitiation to do so, the HCSDU transfer will be rejected with the message *Transfer unsuccessful* through an HC_Status_Indication primitive.

Acknowledgement The acknowledgement packet AK-HCPDU is used to acknowledge the receipt of a DT-HCPDU. It is created after the unicast transmission of a DT-HCPDU has been successfully completed.

The sender of a DT-HCPDU directed to a single HCSAP address waits for an acknowledgement of receipt at the end of the transmission. The received AK-HCPDU must contain the identification of the DT-HCPDU sent and must arrive within the designated time. If the identificaiton does not agree or there is no acknowledgement, the transmission is considered to have errors.

HCPDU Transmission An HCS user activates the *User Data Acceptance* function through the HC_Data_Request primitive. The user data acceptance function carries out and acknowledges the HCSDU transfer request of the HCS user. The procedure generates a DT-HCPDU from the HCSDU and the transmission parameters. This consists of an LBR part and an HBR part. The HBR part consists of 52-byte long blocks. The last block is stuffed with filler bits up to this value. The checksums are calculated for the data.

The DT-HCPDU is transmitted as soon as the channel can be obtained exclusively for this station on the basis of the channel-access procedure or directly if the channel is regarded as idle.

Transmission is successful if a group of HCSAP addresses could be contacted or the individual station has acknowledged the receipt with an AK-HCPDU. The HCS user is notified of the outcome of the transmission through an HC_Status_Indication primitive with the status *Transfer successful* or *Transfer unsuccessful*.

The DT-HCPDU receiving procedure is invoked when a DT-HCPDU is received. The packet is evaluated if the destination address of the packet corresponds to the address of the HC entity and the checksum test was successful. Otherwise the DT-HCPDU is rejected.

If the destination address is the address of an individual HCSAP then the acknowledgement procedure is invoked to acknowledge that receipt was successful.

13.7 The Physical Layer

13.7.1 Tasks

The *physical layer* represents the lowest layer in the HIPERLAN reference model. It forms the interface of a station to the radio channel and is responsible for the following tasks:

- Modulation and demodulation of a prescribed bit stream on the carrier of a radio connection.
- Establishment and maintenance of bit and packet synchronism between senders and receivers.
- Transmission or receipt of a prescribed number of bits over a special carrier frequency at a requested time.
- Addition and removal of synchronization sequence.
- Coding and decoding in accordance with a forward error-correction scheme.
- Notification of received signal strength.
- Determination of channel state to enable a decision on delaying channel-access.
- Adaptation of transmit and receive signal level.

13.7.2 Services of the Physical Layer

The standard lists the service primitives of the physical layer to facilitate a description of the procedures of the layer. This list is somewhat abstract, and can differ from implementation to implementation.

The services of the physical layer allow HIPERLAN-CAC sublayers to exchange data between CAC entities. The characteristics of the physical layer relating to data transmission are described below on the basis of the service primitives. With respect to the radio channel, these primitives conform to the corresponding primitives of wireline WANs.

The physical layer scans the radio channel at regular intervals, and notifies the CAC sublayer of the channel status through an HPh_Channel-State_Indication. This process is illustrated in Figure 13.25. If the channel is free, the CAC sublayer is signalled that data can be transmitted. Table 13.11 presents an overview of the different possible channel states.

Data cannot be transmitted directly, but instead must first pass through a priority phase and a contention phase in the access cycle. This involves the transmission of different bursts, as described in Section 13.6.3.1.

Table 13.11: Parameters of HPh_Channel-State_Indication primitive

Parameter	Description	Limits
Channel state	Indication of whether the channel is busy or idle	Channel state ∈ {`BUSY, IDLE`}

Table 13.12: Parameters of HPh_Signal_Request primitive

Parameter	Description	Limits
Action	Whether to start or stop transmission	Action ∈ {`start, stop`}

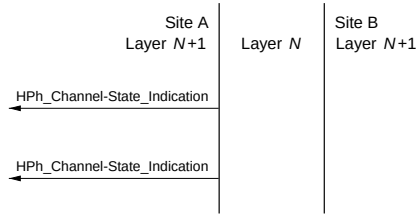


Figure 13.25: Sequence of primitives for indication of channel status

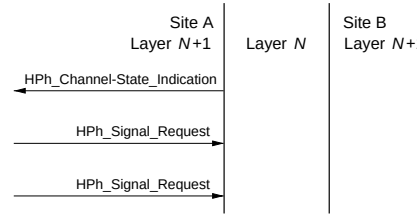


Figure 13.26: Sequence of primitives for transmission of a burst

The physical layer receives an HPh_Signal_Request from the CAM of the CAC sublayer when a burst is to be transmitted. It then begins to transmit the burst over the radio channel. The physical layer does not end the transmission of the burst until another HPh_Signal_Request has been received (see Figure 13.26). Table 13.12 lists the parameters for an HPh_Signal_Request.

If a station was successful during the priority and the contention stages, the physical layer receives an HPh_Unitdata_Request from the CAC sublayer to transmit the data packet. If the data was received successfully, the physical layer of the receiving station forwards it with an HPh_Unitdata_Indication to the CAC sublayer and is then requested through an HPh_Unitdata_Request to transmit an Acknowledgement to the sending station.

If the Acknowledgement is successfully received by the physical layer of the sending station, the CAC sublayer of the sender is notified through an HPh-UNITDATA indication, as shown in Figure 13.27, that the data packet has been successfully transmitted.

The example in Figure 13.28 shows the primitives that occur in the different layers if a transfer takes place.

13.7.3 Transmission Rates and Modulation Procedures

Acknowledgements and data packets are modulated differently and with different signal rates by the physical layer. Data packets are modulated with

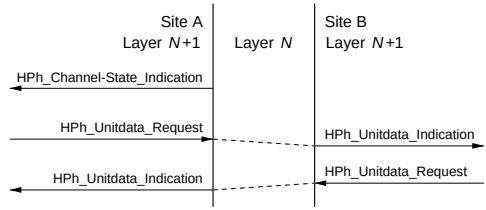


Figure 13.27: Sequence of primitives with data transmission

Table 13.13: Physical characteristics of the physical layer

	High-bit-rate data	Low-bit-rate data
Frame	Data packet	Acknowledgement
Signal rate [Mbit/s]	23.5294	1.4706
Deviation in signal rate [bit/s]	±235	±15
Modulation procedure	GMSK	FSK
BT	0.3	—

Gaussian Minimum Shift Keying (GMSK) and transmitted at a high bit rate (HBR) of 23.5294 Mbit/s. *Frequency Shift Keying* (FSK) is used in the modulation of acknowledgements before they are transmitted at a low bit rate (LBR) of 1.4706 Mbit/s (see Table 13.13).

The selected radio frequency can only deviate a maximum 10^{-5} from the mid-frequency listed in Table 13.2.

13.7.4 Packet Structure

There are two types of frames: data packets in which data is transmitted and acknowledgement frames (Acknowledgement).

A data packet contains the following information:

- *Low-bit-rate* information
- Synchronization and training sequence, 450 *high-rate bits*
- Minimum of one data block with 496 *high-rate bits*; maximum 47 data blocks

Figure 13.29 (a) shows the structure of a data packet. The numbers relate to the bits with high transmission rates. According to Figure 13.29 (b), the Acknowledgement packet only consists of low-bit-rate information.

13.7.5 Receiver Sensitivity

The transmitter power of data packets transmitted by stations should be a maximum of 1 W. Three categories of receiving and transmitting classes are

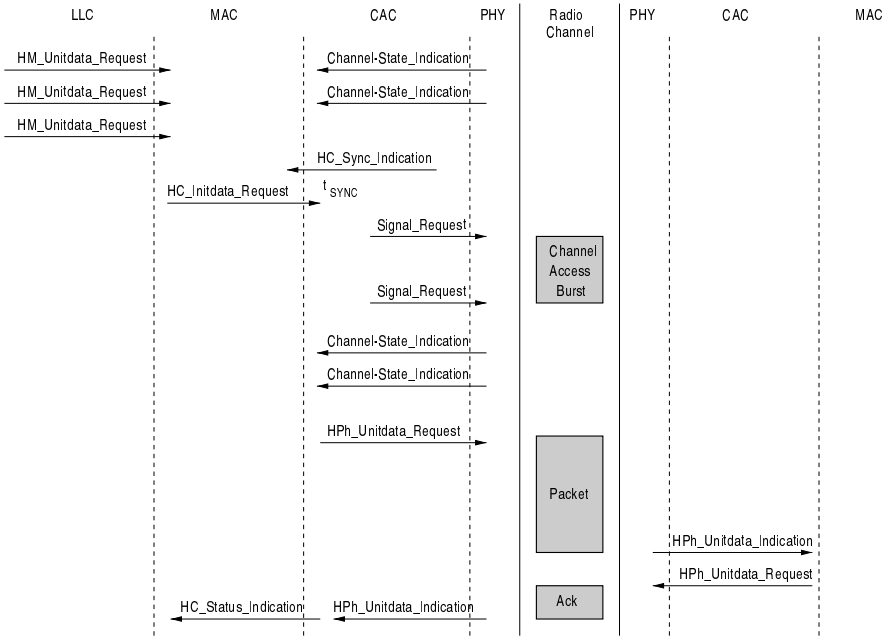


Figure 13.28: Primitives occurring in a transfer

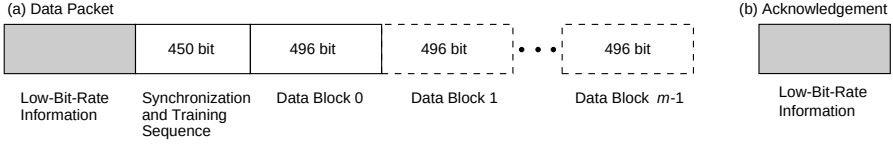


Figure 13.29: Structure of a data packet (a) and of an Acknowledgement (b)

differentiated in HIPERLAN. The allowed combinations of transmitter and receiver sensitivities with the corresponding values are provided in Table 13.14.

Because the aim is to ensure a *bit-error ratio* of 0.001 only, data packets with a *carrier-to-interference* (C/I) ratio of $\geq$13 dB are successfully accepted for receipt. Data packets with a C/I ratio of $<$13 dB cannot be decoded error-free.

HIPERLAN evaluates the channel state on the basis of the received power level. The channel is regarded as occupied or free depending on the threshold value. This threshold value must be set so that non-HIPERLAN interference has no effect on the decision. It is known that HIPERLAN transmissions are bursty in nature. Furthermore, it is assumed that noise has a relatively constant power level. From this knowledge, the channel evaluation scheme can be adapted to the receiving level of the noise so that transmission is possible during low-noise periods.

Table 13.14: Allowed combinations of transmitter and receiver classes

	Transmitter class		
Receiver class	A (+10 dBm)	B (+20 dBm)	C (+30 dBm)
A (−50 dBm)	Allowed	—	—
B (−60 dBm)	Allowed	Allowed	—
C (−70 dBm)	Allowed	Allowed	Allowed

For this purpose, the receive level is divided into areas, which are allocated *signal level numbers* (SLN). The parameters for categorization of the receive level are listed in Table 13.17. The channel detected by the physical layer is considered to be free if the received SLN is smaller than the current threshold value (*Maximum Adaptive Defer Threshold*, MADT). With HIPERLAN the standard threshold value is set at SLN 1. The maximum value for this threshold is at SLN 21, in other words at a level of around −40 dBm.

For adjustment, the threshold value is selected two SLN levels higher than the lowest one during a period of 5 ms received power levels. After every 100 ms, the current threshold value is lowered by one level (*aging of defer threshold*). At the end of a successful transmission, the station measures the level in the channel synchronization cycle and adjusts the threshold value to two SLNs above if this value is at least two levels lower than the current MADT.

13.8 HIPERLAN Parameters

Parameters of EY-NPMA Channel-Access Procedure

Table 13.15 lists the parameters for channel-access using the EY-NPMA mechanism as defined in the standard. Durations are given in units of HBR-bit periods (HBRBP) as in Table 13.13 in Section 13.7.3.

Parameters and Data Formats of MAC Sublayer

Table 13.16 lists the parameters for timers within the MAC protocol and the constants as preallocations in the different HMPDUs.

Parameters of the Physical Layer

Table 13.17 presents the signal level numbers (SLN) of the physical layer which are used in categorizing the receive level.

Table 13.15: Constants of EY-NPMA channel-access procedure in HIPERLAN

Parameter	Significance	Value
m_{CAP}	Number of priority slots	5
m_{ES}	Number of elimination slots	12
m_{YS}	Number of yield slots	14
i_{PS}	Duration of a priority slot	256 HBRBP
i_{PA}	Duration of a priority burst	256 HBRBP
i_{ES}	Duration of an elimination slot	256 HBRBP
i_{ESV}	Duration of an elimination-survival-verification interval	256 HBRBP
i_{CS}	Duration of a channel-access synchronization interval	256 HBRBP
i_{YS}	Duration of a yield slot	64 HBRBP
i_{ACKW}	Duration between middle of last bit of DT-HMPDU and middle of first bit of AK-HMPDU	512 HBRBP
i_{IDLE}	Dur. of interval before a channel is considered *free*	1700 HBRBP
p_{E}	Probability of a burst in the next elimination slot	0.5
p_{Y}	Probability of listening in in next yield slot	0.9

Table 13.16: Constants for the parameters in the MAC protocol

Parameter	Significance	Value
Max_Hops	Maximum number of *Hops* for a TC-HMPDU	128
T_{wp}	Period of validity of *Wake-Pattern-Declaration* entries	30 s
T_{dmp}	Period of validity of *Multicast-Pattern-Declaration* entries	30 s
T_{ah}	Period of validity of alias entries	30 s
T_{hr}	Time between two calls of *Hello* function	10 s
T_{hh}	Period of validity of *Hello* entries	20 s
T_{tcr}	Time between two calls of topology control function	20 s
T_{tch}	Period of validity of the entries	40 s
$\mathtt{T_{lrr}}$	Time between two calls of the lookup function	2 s

Table 13.17: Allocation of SLNs for receive power

SLN	Level [dBm]	Tolerance [dB]
0	<SLN1	
1	−75.0	+3, −5
2	−73.3	+4, −5
3	−71.7	±5
4	−70.0	±5
5	−68.3	±5
...	...	...
29	−28.3	±5
30	−26.7	±5
31	−25.0	±5

Table 13.18: U–NII frequency bands

Band of operation [GHz]			Maximum transmit power [mW]
5.15	...	5.25	50
5.25	...	5.35	250
5.725	...	5.825	1000

13.9 Scope and Purpose of WLAN IEEE 802.11

The purpose of the IEEE 802.11 standard is "To provide wireless connectivity to automatic machinery, equipment or stations that require rapid deployment, which may be portable or hand-held, or which may be mounted on moving vehicles within a local area" [5].

The standard describes the functions and services required to operate within IEEE 802.11 compliant ad hoc and infrastructure networks. It defines MAC procedures and PHY signalling techniques. Furthermore, it describes procedures to provide privacy of user information.

IEEE 802.11 has been designed for use in *Industrial, Scientific and Medical* (ISM) bands. The *Federal Communications Commission* (FCC) in the USA prescribed maximum power levels only, band edge interference, and the requirement to use spread spectrum in order to minimize interference with already existing communication systems. It designated the frequency bands 902–928 MHz, 2400–2483.5 MHz and 5725–5850 MHz to these ISM bands [10].

IEEE 802.11 has defined two PHY standards for the 2.4 GHz ISM band: one using the *Frequency Hopping Spread Spectrum* (FHSS) technique, and one using the *Direct Sequence Spread Spectrum* (DSSS) technique. An alternative is the specification of an *Infrared* (IR) physical layer. IEEE 802.11 stations using any of the three technologies operate at a data rate of 1 Mbit/s (optionally 2 Mbit/s). Recently work started on the specification of a 20 Mbit/s PHY at 5 GHz. Target frequencies are those opened by the FCC in 1997 for *Unlicensed Information Infrastructure Networks* (U–NII); see Table 13.18.

A key issue of 802.11 is that a mobile station is able to communicate with any other mobile or wired station transparently, which means that above the MAC layer, 802.11 appears like any other 802.x LAN and offers comparable services (see Figure 13.30). This implies that mobility is handled at MAC level [1].

The protocols are specified for slowly moving stations, usually indoors but not limited to this, communicating among each other (ad hoc mode) or with stations beyond their direct communication range with the support of an infrastructure (infrastructure mode). The communication is packet-oriented [9].

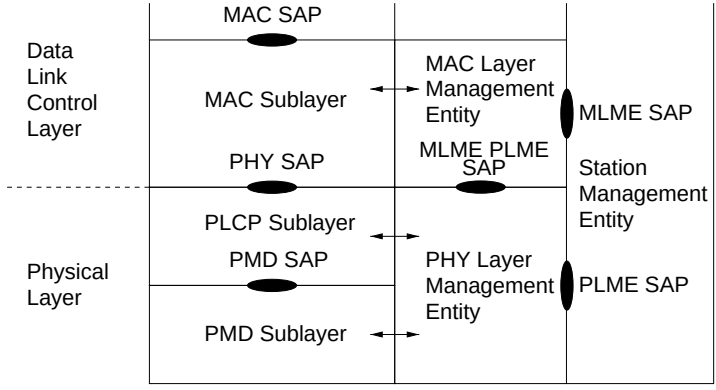


Figure 13.30: IEEE 802.11 reference model

13.9.1 Architecture of IEEE 802.11 Networks

The IEEE 802.11 network architecture is hierarchical [5]. Its basic element is the *Basic Service Set* (BSS), which is a set of IEEE 802.11 stations (STA) controlled by a single *Coordination Function* (CF). The *Distributed Coordination Function* (DCF) is mandatory for each BSS, whereas the *Point Coordination Function* (PCF) is optional [7].

An *Independent BSS* (IBSS) is the most basic IEEE 802.11 type of network. It is an ad hoc network consisting of a minimum of two stations. A BSS may also be part of a larger network, the so-called *Extended Service Set* (ESS). The ESS consists of multiple BSS interconnected by the *Distribution System* (DS). The ESS appears as a single BSS to the IEEE 802.2 LLC layer.

Logically the BSS and the DS work on different media. The BSS operates on the *Wireless Medium* (WM), whereas the DS uses the *Distribution System Media* (DSM). As the IEEE 802.11 architecture is specified independently of any specific media, the WM and DSM may or may not be the same.

Stations connected to the DS are called *Access Points* (AP), which provide the *Distribution System Services* (DSS) in order to enable the MAC to transport MAC SDUs between stations that cannot communicate over a single instance of the WM.

A *Portal* is the logical point where a non IEEE 802.11 LAN is connected to the DS. This allows the communication across different types of LANs.

Figure 13.31 comprises the components of the 802.11 architecture.

13.9.1.1 Address Space

IEEE 802.11 only specifies the use of the WM address space. Therefore the address space of the DSM and the integrated wired LAN may differ from the WM. IEEE 802.11 uses the 48-bit address space, so it is most likely that the address spaces of WM, DSM and wired LAN is the same.

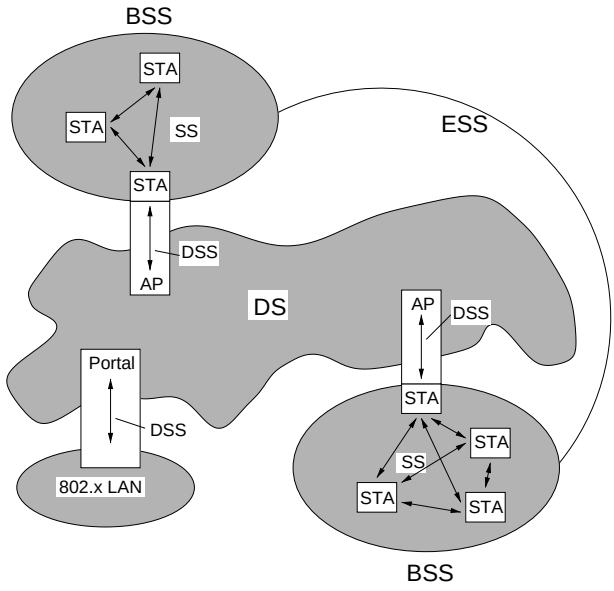


Figure 13.31: Components of the IEEE 802.11 architecture

13.9.2 Services of IEEE 802.11 Networks

The services of IEEE 802.11 are sorted into two categories: the *Station Services* (SS) and the *Distribution System Services* (DSS). The services are listed in Table 13.19.

In an *Independent Basic Service Set* (IBSS) or ad hoc network, only the *Station Services* are available. Besides the *MSDU Delivery*, which will be explained in detail in Section 13.10, the station services provide access and confidentiality control[5]:

Authentication The identity of stations is delivered to each other. Here, only link level authentication is provided. End-to-End system or user-to-user authentication is outside the scope of IEEE 802.11. Nevertheless these services are supported.

Deauthentication An existing authentication is cancelled.

Privacy The contents of link level messages are protected from eavesdropping. IEEE 802.11 specifies an optional privacy algorithm: *Wired Equivalent Privacy* (WEP).

Table 13.19: Services of IEEE 802.11 networks

Station services	Distribution system services
Authentication	Association
Deauthentication	Disassociation
Privacy	Distribution
MSDU delivery	Integration
	Reassociation

The *Distribution System Services* are used to distribute messages in the *Distribution System* and to support mobility.

The *Distribution* service delivers MSDUs within the DS, but it is not specified in IEEE 802.11. The standard provides the DS with sufficient information in order to be able to determine the target AP of an MSDU. This is done by the three association-related services (*Association*, *Reassociation* and *Disassociation*). The *Integration* service enables the delivery of MSDUs between non 802.11 LAN and DS via a *Portal*. The *Integration* service depends on the *Distribution* service, and performs media and address space translation if necessary. The *Integration* service is outside the scope of IEEE 802.11.

Mobility is supported by association services. Three types of mobility are distinguished in 802.11:

No-transition Stations are stationary or move within a BSS.

BSS-transition Stations move from one BSS to another BSS within the same ESS.

ESS-transition Stations move from one BSS in one ESS to another BSS in a different ESS. This kind of mobility is not supported by 802.11; running services will be disrupted.

The association services needed for mobility support are as follows:

Association makes an AP aware of a station in the BSS. It is always invoked by the station. A station cannot associate with more than one AP at the same time. This ensures that the *Distribution* service finds a unique AP for a message in the DS. A station selects the appropriate AP out of several possible APs by scanning. *Association* is sufficient to handle *No-transition* mobility and necessary but not sufficient to support *BSS-transition* mobility.

Disassociation cancels an existing *Association*. The DS is informed about the *Disassociation* by the related AP.

Reassociation transfer an existing *Association* from one AP to another, thus supporting *BSS-transition* mobility by informing the DS about the

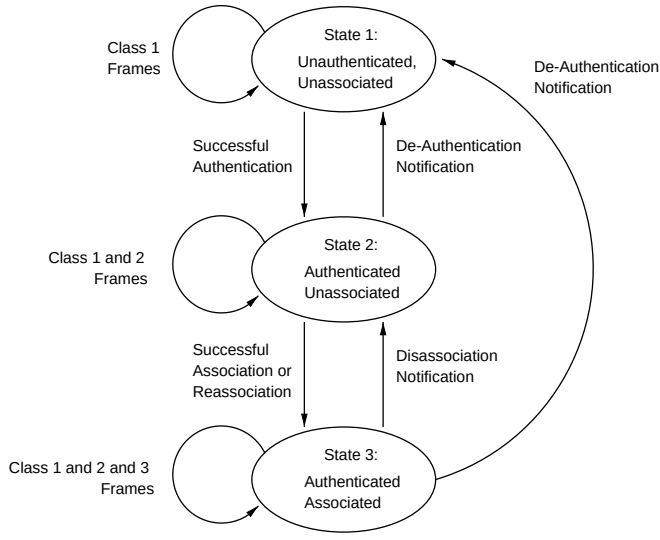


Figure 13.32: Relationship between IEEE 802.11 services

change of AP. In order to speed up the *Reassociation* procedure, a *Pre–authentication* may be invoked prior to the *Reassociation*.

The relationship between the services is shown in Figure 13.32. The frame types allowed to be transmitted between two stations are grouped in three classes corresponding to the state of the station. In an IBSS only class 1 and 2 frames are allowed, since there is no AP and therefore no *Association* is possible [5].

13.10 IEEE 802.11 MAC Sublayer

The IEEE 802.11 MAC protocol provides two types of service: asynchronous and contention-free. The asynchronous type of service is provided by the *Distributed Coordination Function* (DCF), which implements as the basic access method the *Carrier Sense Multiple Access with Collision Avoidance* (CSMA/CA) protocol. The contention-free type of service is provided by the *Point Coordination Function* (PCF), which basically implements a polling access method. Unlike the DCF, the implementation of the PCF is not mandatory. Furthermore, the PCF itself relies on the asynchronous service provided by the DCF as shown in Figure 13.33.

13.10.1 Address Mapping

The IEEE 802.11 standard uses 48-bit address representation. In order to avoid an additional address mapping, five types of addresses are used:

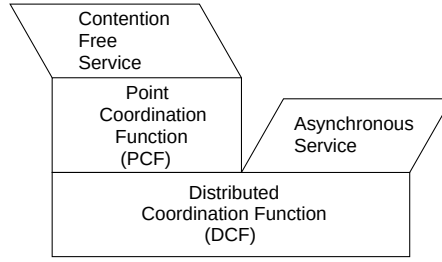


Figure 13.33: IEEE 802.11 MAC architecture

BSS Identifier (BSSID) The BSSID uniquely identifies each BSS. In an infrastructure BSS, the BSSID is the MAC address of the AP serving the BSS. In an independent BSS (IBSS), the BSSID is formed by a random number.

Destination Address (DA) The MAC address of the final recipient of an MSDU.

Source Address (SA) The MAC address of the originator of a MSDU.

Receiver Address (RA) The MAC address of the intended immediate receiver on the wireless medium for the MPDU.

Transmitter Address (TA) The MAC address of the recent station transmitting this MPDU on the wireless medium.

MAC Data Frames have four address fields. The type of addresses assigned to these fields depend on the *To DS* and *From DS* bit settings (see Table 13.20) [9].

13.10.2 MAC Services

As a service provider, the MAC sublayer provides its services to the LLC sublayer at the MAC SAP (see Figure 13.30).

The MAC offers the *Asynchronous Data Service*, which provides the LLC with the ability to exchange *MAC Service Data Units* (MSDU). This service is offered on a best-effort connectionless basis. There are no guarantees that the submitted MSDU will be delivered successfully. Two classes of services are offered: one allowing re-ordering of MSDUs inside the MAC, the other guaranteeing the strict order of MSDUs.

Broadcast and multicast transport is part of the asynchronous data service that will experience a lower quality of service compared to unicast MSDUs owing to the characteristics of the WM [5].

The 802.11 MAC offers a security service by using the WEP. This includes:

1. Confidentiality

Table 13.20: MAC Data Frame Address Field content

To DS	From DS	Meaning	Address Field 1	Address Field 2	Address Field 3	Address Field 4
0	0	Data Frame within IBSS Management and Control Frames	DA	SA	BSSID	N/A
0	1	Data Frame to DS	DA	BSSID	SA	N/A
1	0	Data Frame from DS	BSSID	SA	DA	N/A
1	1	WDS Frame distributed from one AP to another AP	RA	TA	DA	SA

Table 13.21: 802.11 MAC primitives and their parameters

Primitive	Parameter
MA-UNITDATA.request	Source address, destination address, routing information, data, priority, service class
MA-UNITDATA.indication	Source address, destination address, routing information, data, reception status, priority, service class
MA-UNITDATA-STATUS.indication	Source address, destination address, transmission status, provided priority, provided service class

2. Authentication
3. Access control in conjunction with layer management

13.10.2.1 MAC Service Primitives

The service primitives supported by the IEEE 802.11 MAC are taken from the ISO/IEC 8022-2 standard [6]. The service primitives at the MAC SAP and their parameters are listed in Table 13.21.

The meaning of the parameters listed in Table 13.21 are:

Source address (SA) Address of the MAC sublayer entity receiving the MA-UNITDATA.request primitive.

Destination address (DA) An individual or a group MAC sublayer entity address.

Routing information Must be null for 802.11.

Data MSDU that is transported without modification by the MAC between to LLC sublayer entities. The size of the MSDU must be less or equal to 2304 octets for 802.11.

Priority 802.11 offers two priority levels: *Contention* and *Contention-Free*.

Service class Two classes are defined: *Reorderable Multicast* and *Strictly Ordered*.

Reception status The result of the reception is reported, namely *Success* and *Failure*.

Transmission status Information about the transmission passed back to the local LLC sublayer entity. Possible values are:

1. Successful
2. Undeliverable
3. Excessive data length
4. Non-null source routing
5. Unsupported priority
6. Unsupported service class
7. Unavailable priority
8. Unavailable service class

13.10.3 MAC Protocol

The IEEE 802.11 MAC protocol is based on a *Carrier Sense Multiple Access with Collision Avoidance* (CSMA/CA) protocol. The time to sense the carrier is defined by the *Interframe Space* (IFS). The collision avoidance is done by a random backoff procedure [9].

13.10.3.1 Interframe Space

The time between two frames is called *Interframe Space* (IFS). In order to determine whether the medium is free, a station has to use the carrier sense function for a specified IFS. The standard specifies four different IFSs, which represent three different priority levels for the channel-access. The shorter the IFS, the higher the priority. The IFSs are specified as time gaps on the medium and are independent of the channel data rate. Owing to the different characteristics of the different PHY specifications, the IFS time durations are specific for each PHY.

Some relations between the IFS are shown in Figure 13.34. The IFS are listed in order, from the shortest to the longest [5].

Short IFS (SIFS) The SIFS is used for the immediate acknowledgement (ACK frame) of a data frame, the answer (*Clear To Send* (CTS) frame) to a *Request To Send* (RTS) frame, a subsequent MPDU of a fragmented MSDU, response to any polling by the PCF, and any frames of the AP during the *Contention-Free Period* (CFP).

Point Coordination Function IFS (PIFS) The PIFS is used by stations operating under the PCF to gain access to the medium at the start of the CFP.

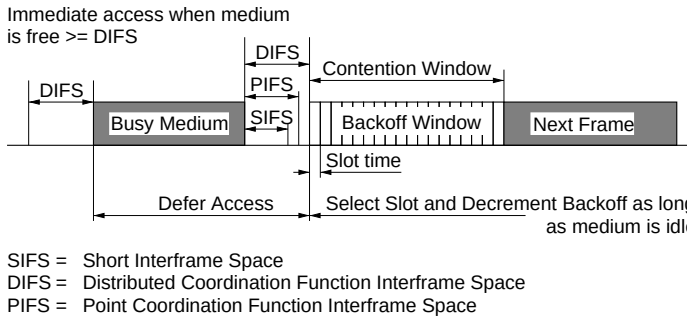


Figure 13.34: Interframe space relationship

Distributed Coordination Function IFS (DIFS) The DIFS is used by stations operating under the DCF to gain access to the medium to transmit data or management frames.

Extended IFS (EIFS) The EIFS is used by the DCF whenever the PHY indicates that a frame transmission did not result in a correct *Frame Check Sequence* (FCS). The EIFS allows another station to acknowledge what was, to this station, an incorrectly received frame.

13.10.3.2 Distributed Coordination Function

According to the DCF (see Figure 13.35), a station must sense the medium before initiating the transmission of a packet. If the medium is sensed as being idle for a time interval greater than a DIFS then the station transmits the packet. Otherwise, the transmission is deferred and the backoff process is started. Specifically, the station computes a random number uniformly distributed between zero and a maximum called *Contention Window* (CW). The random number is multiplied by the *slot time*, resulting in the backoff interval used to set the backoff timer. This timer is decremented only when the medium is idle, whereas it is frozen when another station is transmitting. Each time the medium becomes idle, the station waits for a DIFS and then periodically decrements the backoff timer.

As soon as the backoff timer expires, the station is authorized to access the medium. If two or more stations start transmission simultaneously, a collision occurs. Unlike wired networks (e.g. with CSMA/CD in IEEE 802.3), in a wireless environment collision detection is not possible. Hence, as shown in Figure 13.36, a positive acknowledgement is used to notify the sending station that the transmitted frame has been successfully received. The transmission of the acknowledgement is initiated at a time interval equal to the SIFS after the end of the reception of the previous frame.

If the acknowledgement is not received in a specified time interval, the station assumes that the transmitted frame was not successfully received, and

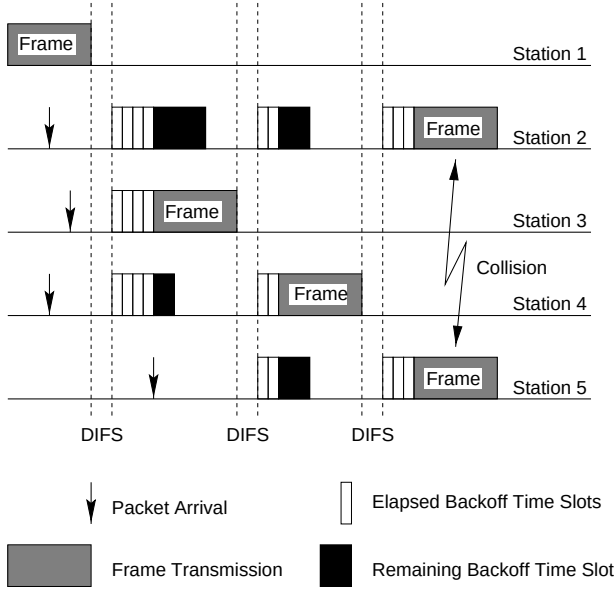


Figure 13.35: Basic access mechanism

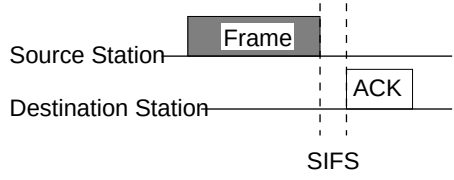


Figure 13.36: Acknowledgement mechanism

hence schedules a retransmission and enters the backoff process again. However, to reduce the probability of collisions, after each unsuccessful transmission attempt the *Contention Window* is doubled until a predefined maximum (CW_{max}) is reached. After a successful transmission, the *Contention Window* is reset to CW_{min}.

After each frame transmission, a station must execute a new backoff process. Therefore at least one backoff is in between two transmissions of the same station [1].

In radio systems based on medium sensing, a phenomenon known as the *hidden-station problem* may occur. This problem arises when a station is able to successfully receive frames from two different stations but the two stations cannot receive signals from each other. In this case a station may sense the medium as being idle even if the other one is transmitting. This results in a collision at the receiving station.

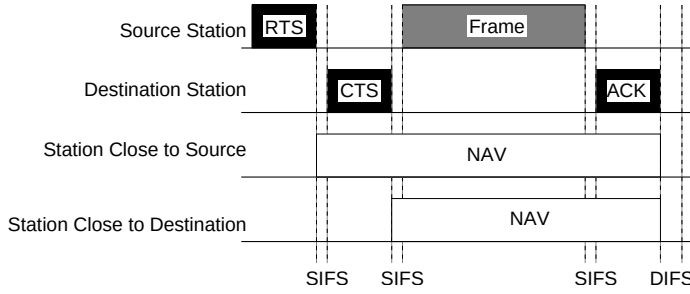


Figure 13.37: RTS/CTS mechanism

To deal with the hidden-station problem, the IEEE 802.11 MAC protocol includes a mechanism based on the exchange of two short control frames (see Figure 13.37): a *Request-to-Send* (RTS) frame that is sent by a potential transmitter to the receiver and a *Clear-to-Send* (CTS) frame that is sent by the receiver in response to the received RTS frame. If the CTS frame is not received within a predefined time interval, the RTS frame is retransmitted by executing the backoff algorithm described above. After a successful exchange of the RTS and CTS frames, the data frame can be sent by the transmitter after waiting for a SIFS. The implementation of the RTS packet is optional, whereas all stations must be able to answer to a RTS with the belonging CTS.

The RTS and CTS frames include a duration field that specifies the time interval necessary to completely transmit the data frame and the related acknowledgement. This information is used by stations that can hear either the transmitter or the receiver to update their *Net Allocation Vector* (NAV), a timer that, unlike the backoff timer, is continuously decremented irrespective of the status of the medium. Since stations that can hear either the transmitter or the receiver refrain from transmitting until their NAV has expired, the probability of a collision occurring because of a hidden station is reduced. Of course, the drawback of using the RTS/CTS mechanism is an increased overhead, which may be significant for short data frames.

Furthermore, the RTS/CTS mechanism can be regarded as a way to improve the MAC protocol performance. In fact, when the mechanism is enabled, collisions can obviously occur only during the transmission of the RTS frame. Since the RTS frame is usually much shorter than the data frame, the waste of bandwidth and time due to the collision is reduced.

In both cases the effectiveness of the RTS/CTS mechanism depends upon the length of the data frame to be protected. Consequently, the RTS/CTS mechanism relies on a threshold, the *RTS threshold*: the mechanism is enabled for data frame sizes over the threshold and disabled for data frame sizes under the threshold [1].

The RTS/CTS is useful also while operating overlapping BSS or IBSS.

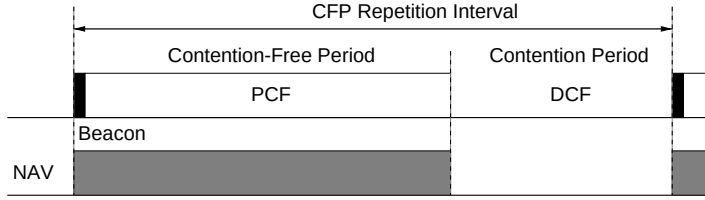


Figure 13.38: Relationship between CFP and CP

Table 13.22: 802.11 MAC Data Subtypes under the PCF

Station	Data Subtypes
PC	Data+CF-Poll, Data+CF-Ack+CF-Poll, CF-Poll, CF-Ack+CF-Poll, CF-End, CF-End+CF-Ack
PC and CF-Pollable	Data, Data+CF-Ack, CF-Ack, Null Function

13.10.3.3 Point Coordination Function

In order to support time-bounded services, the IEEE 802.11 standard defines the *Point Coordination Function* (PCF) to permit a *Point Coordinator* (PC) to have priority access to the medium. Usually an AP in an infrastructure based network acts as a PC.

Although PCF is optional, all stations are able to obey the medium access rules of the PCF, because it is based on the DCF. Stations that are able to respond to polls by the PC are called *Contention-Free-Pollable* (CF-Pollable). Only these stations are able to transmit frames according to the PCF (besides the AP).

The PCF controls the frame transfers during the so-called *Contention-Free Period* (CFP), which alterates with the *Contention Period* (CP) under the control of the DCF. The CFP is periodically repeated in time at the *Contention-Free Repetition Rate* (CFPRate) and starts with the transmission of a beacon (see Section 13.10.4). The beacon contains the maximum duration of the CFP (`CFPMaxDuration`), and all stations in the BSS except the PC set their NAV to CFPMaxDuration, thus guaranteeing the control of the PCF for this amount of time. Figure 13.38 shows these relations.

The PC gains control at the nominal beginning of the CFP by waiting PIFS after the medium is sensed idle instead of DIFS. It maintains control for the entire CFP by waiting a shorter time than stations using the DCF access procedure. The PC maintains a *Polling List*, which consists of the *Association Identifier* (AID) of the stations requesting polling. A *CF-Pollable* station may request to be added to the polling list during *Association* or *Reassociation*.

Special Data Subtypes are defined for the use during CFP in order to enable "piggybacking" of polls and acknowledgements (see Table 13.22). A *CF-Poll*

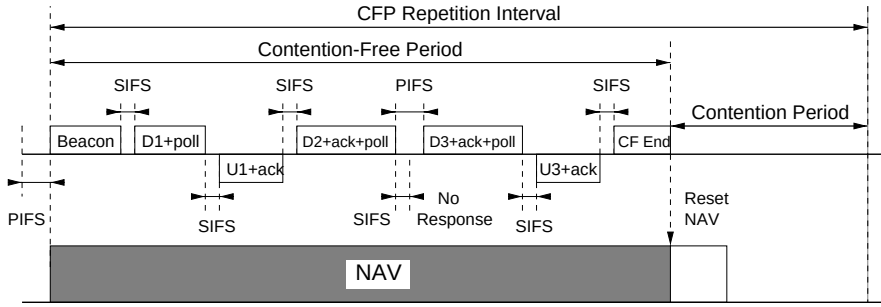


Figure 13.39: Example of PCF frame transfer

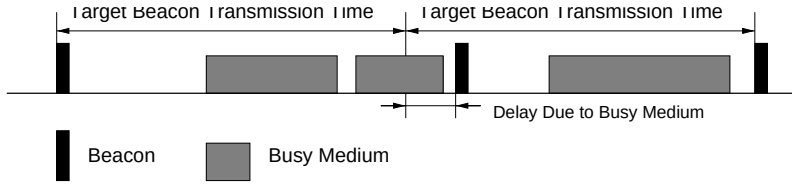


Figure 13.40: Delayed beacon transmission on a busy medium

is used by the PC to poll a station for the transmission of a data frame. *CF-Ack* is the acknowledgement to a successfully received frame under the PCF, either by a station or the PC. The *Null Function* is used to indicate that no data has to be transmitted. If all stations on the polling list have been polled and no more data has to be transmitted by the PC during one CFP, the PC may prematurely stop the current CFP by sending a *CF-End*. On receiving a *CF-End*, all stations reset their NAV.

Figure 13.39 shows an example of a sequence of frame transmissions during a CFP. Usually the gap between two transmissions under the PCF is SIFS unless a station does not respond to a *CF-Poll*. In the later case the PC regains control of the medium after a PIFS.

13.10.4 Synchronization

All stations within a single BSS will be synchronized to a common clock while maintaining a local timer. This is done by performing a *Timing Synchronization Function* (TSF). The TSF is different for infrastructure BSS and IBSS.

The synchronization is maintained by broadcasting the TSF timer in a so-called *beacon*. Beacons are transmitted periodically at the *Target Beacon Transmission Times* (TBTT). In case the medium is sensed busy at TBTT, the transmission of the beacon is delayed as shown in Figure 13.40 [8].

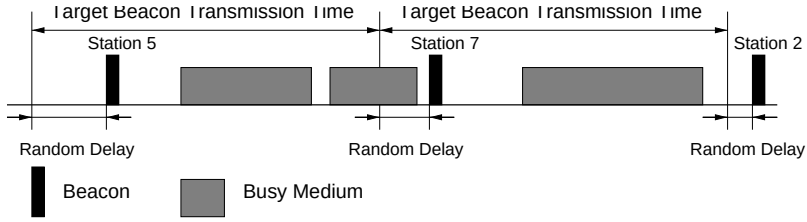


Figure 13.41: Beacon transmission in an IBSS

13.10.4.1 Beacon Generation in Infrastructure Networks

In an infrastructure network the access point is responsible for the transmission of the synchronization beacon. At each TBTT, the AP schedules a beacon as the next frame to be transmitted according to the PCF procedures described in Section 13.10.3.3.

13.10.4.2 Beacon Generation in an IBSS

In an independent BSS the beacon generation is distributed among all stations. At each TBTT all members of a IBSS perform the following procedure:

1. Suspend the decrementing of the backoff timer for any pending non-beacon transmission.
2. Calculate a random delay uniformly distributed in the range between zero and $2 \cdot CW_{min} \cdot slottime$.
3. Wait for the period of random delay similar to the backoff algorithm.
4. If a beacon arrives in between then the random delay timer and the beacon transmission are cancelled.
5. If the random delay timer expires, send a beacon.

Performing the aforementioned procedure, the transmission of the beacon in an IBSS is performed by different stations using different random delays as shown in Figure 13.41.

13.10.5 Power-Saving Mode

IEEE 802.11 compliant devices will most probably be battery-powered. Therefore the standard supports a power conservation mode. A station may be in one of two power states:

Awake A station is fully powered.

Doze A station is able neither to receive nor to transmit, and consumes very low power.

The power management defines two modes, which determine how stations transit from one power state to the other:

Active Mode (AM) A station will be in the *Awake* state. A station on the polling list of a PC will be in *Active Mode* for the duration of the CFP.

Power Save (PS) A station will be in the *Doze* state. It listens to selected beacons (transition to Awake state), and checks whether there are MSDUs to be received or not. If no MSDUs are to be received or transmitted, or all MSDUs are received or transmitted, the station resumes to the *Doze* state.

A station will buffer all MSDUs addressed to stations in PS mode. Multicast and broadcast MSDUs are announced by a field in the beacon called the *Delivery Traffic Indication Message* (DTIM). Only selected beacons contain a DTIM. The multicast and broadcast MSDUs are transmitted immediately after the DTIM beacon [8].

The procedures to announce unicast MSDUs to stations in PS mode are different for infrastructure and independent BSS.

13.10.5.1 Power Management in an Infrastructure BSS

Stations changing power management modes have to inform the AP via a successful frame exchange. Stations having buffered cells within the AP are announced in a *Traffic Indication Map* (TIM), which is included in all beacons generated by the AP. A station receiving and interpreting the TIM may poll the AP for the buffered frames during the *Contention Period* using the *PS-Poll* frame; therefore it is not necessary that a station listens to each single beacon. CF-pollable stations do not poll for frames, since these are transmitted during the CFP.

13.10.5.2 Power Management in an IBSS

As the beacons are generated by different stations (see Section 13.10.4) within the IBSS, the TIM is not sufficient to announce traffic to stations in PS mode. Instead, a station announcing unicast traffic to another station uses the *Ad Hoc Traffic Indication Message* (ATIM). The receiving station acknowledges the reception of the ATIM, and remains in the Awake state for the whole beacon interval, waiting for the announced MSDUs to be delivered.

All stations in PS mode have to wake up for a so-called *ATIM Window*, which follows the TBTT. During the *ATIM Window*, only beacons and ATIMs will be transmitted. After the ATIM, every station in PS mode knows whether it has to remain in the *Awake* state or is allowed to go to the *Doze* state. Figure 13.42 shows the basic operation of power management in an IBSS.

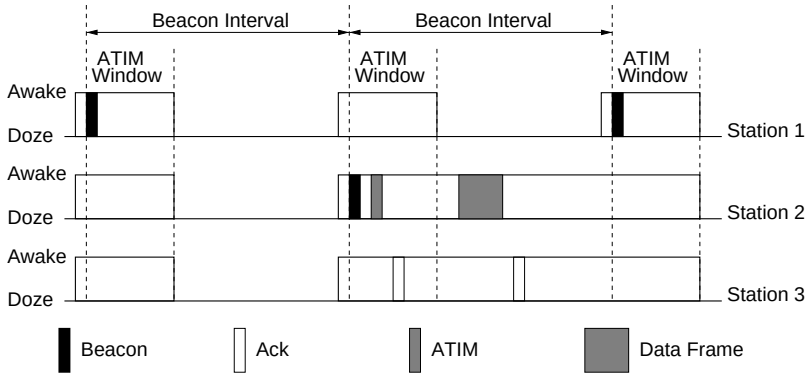


Figure 13.42: Power management in an IBSS

13.11 IEEE 802.11 Physical Layer Specification

13.11.1 Scope

The IEEE 802.11 *Physical Layer* (PHY) consists of two sublayers: the *Physical Layer Convergence Procedure* (PLCP) sublayer and the *Physical Medium-Dependent* (PMD) sublayer (see Figure 13.30). The PLCP sublayer maps the 802.11 MAC PDUs into a framing suitable for transmitting and receiving user data and management information between the associated PMD entities. The PMD system describes the method of transmitting and receiving data through a wireless medium.

13.11.2 Physical Layer Service Definition

Two groups of service primitives are distinguished: *peer-to-peer* and *sublayer-to-sublayer* service primitives.

The *peer-to-peer* service primitive is:

PHY-DATA This primitive is used to transfer octects of data between the PHY and MAC sublayers.

The *sublayer-to-sublayer* service primitives are:

PHY-TXSTART With this primitive, the start of the transmission of a MPDU is requested by the MAC layer.

PHY-TXEND The MAC layer requests the end of a transmission.

PHY-RXSTART The PHY layer indicates to the local MAC entity that the PLCP has received a valid *start frame delimiter* (SFD) and PLCP header.

Table 13.23: PHY service primitive parameters

Parameter	Associated primitive	Value
DATA	PHY-DATA.request PHY-DATA.indication	Octect value X'00'–X'FF'
TXVECTOR	PHY-TXSTART.request	Set of parameters
RXVECTOR	PHY-RXSTART.request	Set of parameters
STATUS	PHY-CCA.indication	BUSY, IDLE
RXERROR	PHY-RXEND.indication	NoError, FormatViolation, CarrierLost, UnsupportedRate

Table 13.24: FHSS specific service primitive parameters

Parameter	Associated parameter	Value
LENGTH	TXVECTOR, RXVECTOR	1–4095
DATARATE	TXVECTOR, RXVECTOR	1, 1.5, 2, 2.5, 3, 3.5, 4, 4.5
RSSI	RXVECTOR	0–RSSI Max

PHY-RXEND With this primitive, the end of a MPDU currently received is indicated to the local MAC entity.

PHY-CCA The PHY layer indicates the state of the medium to the local MAC entity. Possible states are IDLE and BUSY. It is generated whenever the state of the medium changes.

PHY-CCARESET With this primitive, the MAC layer requests to reset the *clear channel assessment* (CCA) state machine.

Table 13.23 lists the parameters belonging to the PHY service primitives.

The architecture of the IEEE 802.11 MAC is intended to be PHY-independent. However, some PHY implementations require the interaction with the *MAC sublayer management entity* (MLME) through the PHY SAP using the PHY service primitives. Therefore the parameter lists *TXVECTOR* and *RXVECTOR* are PHY-dependent. Two fields are mandatory, namely *DATARATE*, specifying the transmit or receive data rate, and *LENGTH*, determining the length of an MPDU.

13.11.3 Frequency Hopping Spread Spectrum

13.11.3.1 FHSS Specific Service Parameters

The *Frequency Hopping Spread Spectrum* (FHSS) dependent definitions of *TXVECTOR* and *RXVECTOR* are listed in Table 13.24.

The *receive signal strength indicator* (RSSI) is an optional parameter indicating the energy observed at the antenna used to receive the current PPDU.

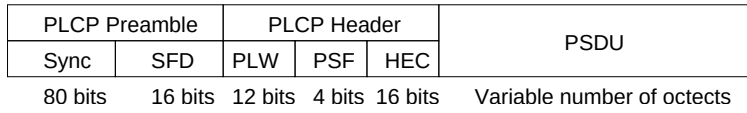


Figure 13.43: FHSS frame format

Table 13.25: Frequency range, channels and hopping sets

Geographical region	Operating frequency [GHz]	Regulatory range [GHz]	Hopping set
North America	2.402 ... 2.480	2.400 ... 2.4835	79
Europe	2.402 ... 2.480	2.400 ... 2.4835	79
Japan	2.473 ... 2.495	2.471 ... 2.497	23
Spain	2.447 ... 2.473	2.445 ... 2.475	27
France	2.448 ... 2.482	2.4465 ... 2.4835	35

RSSI is intended to be used in a relative manner, since the absolute accuracy of the RSSI is not specified.

13.11.3.2 FHSS Frame Format

The *Physical Layer Convergence Procedure* (PLCP) maps an MAC PDU into a frame format designed for the FHSS radio transceiver.

The frame format of the FHSS PLCP is shown in Figure 13.43. It consists of three parts: the PLCP Preamble, the PLCP Header and the PSDU.

The PLCP Preamble consists of a 80-bit SYNC field and a 16-bit *Start Frame Delimiter* (SFD). The SYNC field is used to detect a receivable signal, and to give frequency correction and synchronization with the packet timing. The SFD defines the frame timing.

The PLCP Header consists of a 12-bit *PSDU Length Word* (PLW), a 4-bit *PLCP Signalling Field* (PSF) and a 16-bit *Header Error Check* (HEC). The PLW determines the number of octects contained in the PSDU, the PSW defines the data rate of the PSDU, and the HEC provides an error detection facility for the PLCP Header.

13.11.3.3 FHSS Regulatory Requirements

The regulatory requirements addresses the operating frequency range, the regulatory range and the hopping sequences, as listed in Table 13.25.

13.11.3.4 FHSS Modulation and data rates

The FHSS PMD operates at two data rates: 1 Mbit/s and 2 Mbit/s. 1 Mbit/s is mandatory, whereas 2 Mbit/s is optional. For operation at 1 Mbit/s a two-level *Gaussion Frequency Shift Keying* (GFSK) modulation with a nominal

PLCP Preamble		PLCP Header				PSDU
Sync	SFD	SIGNAL	SERVICE	LENGTH	CRC	
144 bits	16 bits	8 bits	8 bits	16 bits	16 bits	Variable number of octects

Figure 13.44: DSSS frame format

bandwidth bit period (BT) = 0.5 is used. For operation at 2 Mbit/s a four-level GFSK modulation with a nominal bandwidth bit period (BT) = 0.5 is used.

13.11.3.5 FHSS Receiver Sensitivity

The sensitivity is defined as the minimum signal level required for a *Frame Error Ratio* (FER) of 3% for PSDUs of 400 octects generated with pseudo-random data. For the 1 Mbit/s FHSS the sensitivity should be less than or equal to −80 dBm. For the 2 Mbit/s FHSS the sensitivity should be less than or equal to −75 dBm.

13.11.4 Direct Sequence Spread Spectrum

13.11.4.1 DSSS Frame Format

The *Physical Layer Convergence Procedure* (PLCP) maps an MAC PDU into a frame format designed for the DSSS radio transceiver.

The frame format of the DSSS PLCP is shown in Figure 13.44. It consists of three parts: the PLCP Preamble, the PLCP Header and the PSDU.

The PLCP Preamble consists of a 128-bit SYNC field and a 16-bit *Start Frame Delimiter* (SFD). The SYNC field is used to detect a receivable signal, and to give frequency correction and synchronization with the packet timing. The SFD defines the frame timing.

The PLCP Header consists of an 8-bit SIGNAL field, an 8-bit SERVICE field, a 16-bit LENGTH field and a 16-bit CRC field. The SIGNAL field defines the data rate of the PSDU, the SERVICE field is reserved for future use, the LENGTH indicates the number of microseconds required to transmit the PSDU, and the CRC field allows error detection in the PLCP Header.

13.11.4.2 DSSS Regulatory Requirements

The regulatory requirements addresses the operating frequency range, the number operating channels and the maximum transmit power levels, as listed in Table 13.26.

13.11.4.3 DSSS Modulation and Data Rates

The symbol duration is exactly 11 chips long. The following 11-chip Barker sequence is used as the PN code sequence:

Table 13.26: Frequency range, channels and power levels

Geographical region	Frequency channels [GHz]	Number of channels	Transmit power
North America	2.412 ... 2.462	11	1000 mW
Europe	2.412 ... 2.472	13	100 mW (EIRP)
Japan	2.484	1	10 mW/MHz
Spain	2.457 ... 2.462	2	100 mW (EIRP)
France	2.457 ... 2.472	4	100 mW (EIRP)

$$+1, -1, +1, +1, -1, +1, +1, +1, -1, -1, -1$$

The DSSS PMD operates at two data rates: a *basic access rate* and an *enhanced access rate*. The *basic access rate* is based on 1 Mbit/s *Differential Binary Phase Shift Keying* (DBPSK) modulation. The *enhanced access rate* is based on 2 Mbit/s *Differential Quadrature Phase Shift Keying* (DQPSK) modulation.

13.11.4.4 DSSS Receiver Sensitivity

The sensitivity is defined as the minimum signal level required for a *Frame-Error Ratio* (FER) of 8 % for PSDUs of 1024 bytes. The sensitivity is specified for the 2 Mbit/s DQPSK modulation, and should be less than or equal to −80 dBm.

13.11.5 Infrared

The infrared (IR) PHY uses light in the 850–950 nm range. The PHY IR is not directed, but uses both line-of-sight and reflected energy. The operation is limited to indoor environments and the typical range is around 10 m. IEEE 802.11 IR devices are able to deal with disturbances from other common consumer devices operating in the same spectral range, such as infrared remote controls.

13.11.5.1 IR Frame Format

The *Physical Layer Convergence Procedure* (PLCP) maps an MAC PDU into a frame format designed for the IR radio transceiver.

The frame format of the IR PLCP is shown in Figure 13.45. It consists of three parts: the PLCP Preamble, the PLCP Header and the PSDU.

The PLCP Preamble consists of a SYNC field that is 57–73 slots long and a 4-slot *Start Frame Delimiter* (SFD). A slot corresponds to one of the L-positions of a symbol, and has a 250 ns duration. The SYNC field is used to

PLCP Preamble		PLCP Header				PSDU
Sync	SFD	DR	DCLA	LENGTH	CRC	
57-73 slots	4 slots	3 slots	32 slots	16 bits	16 bits	Variable number of octects

Figure 13.45: IR frame format

perform clock recovery (slot synchronisation), automatic gain control, signal–to–noise ratio estimation, and diversity selection. The SFD is provided to perform bit and symbol synchronization.

The PLCP Header consists of a 3-slot *Data Rate* (DR) field, a 32-slot *DC Level Adjustment* (DCLA) field, a 16-bit LENGTH field and a 16-bit CRC field. The DR field defines the data rate of the LENGTH, CRC and PSDU fields, the DCLA field is required to allow the receiver to stabilize to the dc level, the LENGTH indicates the number of microseconds required to transmit the PSDU, and the CRC field allows an error detection in the PLCP Header.

13.11.5.2 IR Regulatory Requirements

Worldwide, there are currently no frequency and bandwidth allocation regulatory restrictions on infrared emissions.

13.11.5.3 IR Modulation and data rates

The IR PMD operates at two data rates: a *basic access rate* and an *enhanced access rate*. The *basic access rate* is based on 1 Mbit/s 16-*Pulse Position Modulation* (PPM). A group of 4 data bits is mapped to one 16-PPM symbol. The *enhanced access rate* is based on 2 Mbit/s 4-PPM. A group of 2 data bits is mapped to one 4-PPM symbol.

13.11.5.4 IR Receiver Sensitivity

The sensitivity is defined as the minimum irradiance at the photodetector plane required for a *Frame-Error Ratio* (FER) of $4 \cdot 10^{-5}$, with a PSDU of 512 octets and with an unmodulated background IR source between 800 nm and 1000 nm, with a level of 0.1 mW/cm^2. For 1 Mbit/s the sensitivity should be less than or equal to $2 \cdot 10^{-5}$ mW/cm^2. For 2 Mbit/s the sensitivity should be less than or equal to $8 \cdot 10^{-5}$ mW/cm^2.

13.12 W-LAN Abbreviations and Acronyms

ADA	Alias Destination Address
ASA	Alias Source Address
ATM	Asynchronous Transfer Mode
CAC	Channel-Access Control

CAM	Channel-Access Mechanism
CCITT	Comité Consultatif International des Télégraphique et Téléphonique
CEPT	Conférence Européenne des Postes et Télécommunications
DA	Destination Address
DECT	Digital Enhanced Cordless Telecommunications
DMPD	Deferred Multicast Pattern Declaration
ETSI	European Telecommunications Standards Institute
FIFO	First-In First-Out
HBRBP	High-Bit-Rate Bit Period
HC	Hop Counter
HC-	HIPERLAN-CAC-
HCS	HIPERLAN-CAC Service
HCSAP	HIPERLAN-CAC Service Access Point
HID	HIPERLAN Identifier
HIPERLAN	High-Performance Radio Local Area Network
HM-	HIPERLAN-MAC-
HMPDU	HIPERLAN-MAC Protocol Data Unit
HMQoS	HIPERLAN-MAC Quality of Service
HMS	HIPERLAN-MAC Service
IEEE	Institute of Electrical and Electronics Engineers
ISDN	Integrated Services Digital Network
ISO	International Standardization Organization
ITU	International Telecommunications Union
LAN	Local Area Network
LLC	Logical Link Control
MAC	Medium-Access Control
MRSC-SN	Multipoint Relay Set Change Sequence Number
MRT	Multipoint Relay Table
MSAP	MAC Service Access Point
MSDU	MAC Service Data Unit
NRML	Normalized Residual MSDU Lifetime
OSA	Originator Address
OSI	Open Systems Interconnection
PCMCIA	Personal Computer Memory Card Interface Association
RIB	Routing Information Base
RML	Residual MSDU Lifetime
SA	Source Address
SDU	Service Data Unit
TDA	Target Destination Address
T-AH	Alias Holding Timer
UP	User Priority
WPD	Wake Pattern Declaration
WLAN	Wireless Local Area Network

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14
Mobile Satellite Communication*

14.1 Fundamentals

In principle communications satellites provide the same connectivity as terrestrial (wireless and wireline) networks. The advantages of satellites, such as fast wide-area coverage, flexible transmission parameters and cost independence due to distance, are compared with the disadvantages, such as restricted channel capacity because of the frequencies available, orbital positions, need for line-of-sight connectivity and high initial investment besides relatively long signal propagation times. As a result, only certain application areas have been developed for satellites in the past.

14.1.1 Application Areas

Satellites are being widely used for distribution functions, e.g., for transmitting television and radio programmes as well as for data. Existing communications networks can be totally bridged through the use of satellite systems. Satellite communication, which until recently was almost only exclusively used for navigation and aviation as well as in land vehicles, is a branch that opens up a totally new world of applications. Satellite paging along with GPS (*Global Positioning System*) and GLONAS have recently been introduced for civil use. The interest in global personal communications is leading to big efforts in the development of new satellite systems that operate at the low orbital heights.

Currently mobile satellite systems are being used mainly in areas where no other terrestrial communications systems are available (on the open seas, in the desert, in rural regions, etc.). These systems are also attractive to users who operate internationally and otherwise use different kinds of terrestrial mobile radio systems, requiring them to carry terminals with different standards.

A differentiation is made between worldwide, regional and national systems, depending on the coverage area of a satellite system. In terms of institutional and organizational structure, a distinction is made between international, national and private operators of satellite systems. Tables 14.1–14.5 present a compilation of the known parameters of all systems currently in development.

*With the collaboration of Branko Bjelajac and Alexander Guntsch

Table 14.1: Narrowband satellite systems concentrating on telephony applications—Part 1

System	Globalstar	ICO	IRIDIUM	Odyssey	Ellipso	ECCO (later: Aries)
Company	Loral, Qualcomm, Alcatel Espace	ICO Global Communications	Motorola	TRW, Teleglobe Canada	Mobile Communications Holdings	Constellation Inc., Telebras
Orbit	LEO (circular)	MEO (circular)	LEO (circular)	MEO (circular)	MEO (circ. + ell.)	LEO (circular)
Path altitude [km]	1414	10 355	780	10 354	520–7846 (ell.) 8040 (circ.)	2000
No. of sats. + spare sats.	48 + 8	10 + 2	66 + 12	12 + 2 on the Ground	14 + 3	11 + 1 (later add'l 35 + 7)
No. of orbits	8	2	11	3	3 (2 ellipt. + 1 circular)	1 equat. (later add'l 7)
No. of ground stations	100	12	21	7	≥ 20	11 (more later)
Inclination [°]	52	45	86	50	116.5 (ellipt.)/ 0 (circ.)	0 (later 62)
Min. elevation [°]	10	10	10	20	25–30	5
Cells/sat.	16	163	48	37	61	32
ISL	—	—	4/sat.	—	—	—
Access methods	CDMA	FDMA/TDMA	FDMA/TDMA	CDMA	CDMA	CDMA
Duplex method		FDD	TDD			
Cluster size	6		180 (global) 5 (USA) 4 (Europa)	6.3		
Error handling	FEC $r = \frac{1}{3}$–$\frac{1}{2}$		FEC MS: $r = \frac{4}{3}$ Grnd.st.: $r = \frac{1}{2}$	FEC $r = \frac{1}{2}$		FEC Voice: $r = \frac{1}{2}$ Data: $r = \frac{1}{4}$

Table 14.2: Narrowband satellite systems concentrating on telephony applications - Part 2

System	Globalstar	ICO	IRIDIUM	Odyssey	Ellipso	ECCO (later: Aries)
Modulation	QPSK		QPSK	QPSK		BPSK
No. of channels/sat.	2700/ 2.4 kbit/s	4500/ 4.8 kbit/s	4070/2.4 kbit/s	2800		1000
No. of channels in entire system	130 000	45 000	283 000	27 600		11 000 (later 46 000)
Channel bandwidth [kHz]		25.2				
Transmission rate [kbit/s]	2.4–9.6	4.8–38.4	2.4	2.4–9.6	0.3-9.6	bis 9.6
Voice transmission rate [kbit/s]	2.4/4.8/9.6	4.8	2.4	9.6		
Bit-error ratio voice/data	$10^{-3}/10^{-6}$		$10^{-2}/10^{-3}$	$10^{-3}/10^{-5}$		$10^{-3}/10^{-6}$
Frequency UL [MHz]	1610–1621.35	1980–2010	1621.35–1626.5	1610–1621.35	1610–1621.35	1610–1621.35
Frequency DL [MHz]	2483.5–2500	2170–2200	1621.35–1626.5	2483.5–2500	2483.5–2500	2483.5–2500
Bandwidth UL+DL [MHz]	27.85[a]	70	5.15	27.85[a]	27.85[a]	27.85[a]
Frequency GW–Sat [GHz]	5.091–5.25	5.15–5.25	29.1–29.3	29.1–29.4	15.45–15.65	5.05–5.25
Frequency Sat–GW [GHz]	6.875–7.055	6.975–7.075	19.4–19.6	19.3–19.6	6.875–7.075	6.825–7.025
Satellite weight [kg]	450	2300	689	2000	689 & 877	425
Antenna type	Planar horn		Planar	Dual refl.		
Satellite intelligence	Available	Available	Available	No	No	No
Satellite transmission mode	Transparent	Regenerative	Regenerative, switching	Transparent	Transparent	Transparent
Transmitter power [W]	1100	5000	1430	4500	2300	815
Power reserve [dB]	3–10 dyn.		16	6		
Max. delay [ms]	11.5		8.22	44.3		
Eavesdropping security	Possible		High	Possible		

Table 14.2: Narrowband satellite systems concentrating on telephony applications—Part 2 (continued)

System	Globalstar	ICO	IRIDIUM	Odyssey	Ellipso	ECCO (later: Aries)
Terminal (mobile)	Dual-mode	Dual-mode	Dual-mode	Dual-mode	Dual-mode	Dual-mode
Terminal trans. power [W]		3.8				
Services	Voice, data, fax, RDSS, SMS	Voice, data, fax, RDSS, SMS	Voice, data, fax, RDSS, SMS	Voice, data, fax, RDSS, SMS	Voice, data, fax, RDSS, SMS	Voice, data, fax, RDSS, SMS
RDSS accuracy [km]	0.3–2		0.5			
Coverage area	74° S–74° N	Global	Global	Global	55° S–90° N	23° S–23° N (later global)
Availability [%]	90–95		90–95	99.5 (User) 99.9 (Grnd.st.)		
No. of users [Mill]	2–5		1.4	2.3	1.0	> 1.0
System cost [Mill. US$]	2.6	2.6	4.4	3.2	0.9	0.55 (later 1.7)
Price of terminal [US$]	750	1000	2000–3000	550	1000	1500
Cost/min [US$]	0.35–0.53	1–2	3	0.65	0.12–0.5	
Start comm. operations	1998	2000	1998	1998	2000	2000
Useful life [years]	7.5	12	5–10	15	5	5
German partners	DB Aerospace		Vebacom			
Licence	01/95 granted by FCC	10/95 frequency alloc. by ITU	01/95 granted by FCC	01/95 granted by FCC	06/97 granted by FCC	06/97 granted by FCC

[a]The UL and DL frequency bands are occupied by CDMA systems simultaneously

Table 14.3: Narrowband satellite systems concentrating on message transfer

System	Orbcomm	E-Sat	Faisat	GE Starsys	GEMnet	LEO One
Company	Magellan Systems, Teleglobe, Orbital Science	Echostar Communications	Final Analysis, Polyglott Enterp., VITA	GE American Comm., CLS North America	CTA Orbital Sciences	dBX Corp.
Orbit	LEO	LEO	LEO	LEO	LEO	LEO
Path altitude [km]	775	1260		1067	1000	950
No. of satellites	28 + 8	6	26 + 4	24	38	48
No. of orbits	8		6	4		8
No. of ground stations	min. 1 per country					USA 3, others in other countries
Inclination	45°					50°
Intersatellite links	—	—	—	—	—	—
Access method	CDMA			CDMA		
Transmission rate [kbit/s]	0.3–2.4		UL: 1.2–19.2 DL: 1.2–38.4			2.4–9.6
Frequency uplink [MHz]	148–149.9		Fin. An.: 455–456, 459–460 VITA: 148–149.9	148–149.9		148–150.05
Frequency downlink [MHz]	137–138		400–401 (Fin. An. and VITA)	137–138		137–138
Bandwidth UL+DL [MHz]	2.9			2.9		3.05
Frequency GW–Sat [MHz]	148–149.9					148–150.05
Frequency Sat–GW [MHz]	137–138					400.15–401
Satellite weight [kg]	46			80		125

Table 14.3: Narrowband satellite systems concentrating on message transfer (continued)

System	Orbcomm	E-Sat	Faisat	GE Starsys	GEMnet	LEO One
Terminal (communicator)	×	×	×	×	×	×
Services (Monitoring, control, message transfer)	×	×	×	×	×	×
Coverage area	Global	N. America	Global	Global	Global	65° S–65° N
System costs [Mill. US$]	0.35	0.05	0.25	0.17	0.16	0.25
Price of terminal [US$]						100–500
Start comm. operations	1998	1998	2002	1999	1999	2000
Useful life	4				5–7	5

Table 14.4: Broadband satellite systems concentrating on data transfer

System	Teledesic	Celestri LEO[a]	M-Star	SkyBridge
Company	Teledesic	Motorola	Motorola	Alcatel Espace, Loral Space & Comm.
Orbit	LEO	LEO	LEO	LEO
Path altitude [km]	1375	1400	1350	1457
No. of satellites	288	63	72	64
No. of orbits	12	7	12	16
No. of ground stations		2 control + 6 antenna centres, many GWs	6	
Inclination	40°	48°	47°	
No. of cells per sat.	576	260		
Intersatellite links	8 per sat. (visual)	6 per sat. (visual)	4 per sat.	No
Access method	DL: Asynch. TDMA UL: MF-TDMA	DL: FDM/TDM UL: FDM/TDMA		
No. of channels per satellite	125 000 at 16 kbit/s			
No. of channels in total system	Simult. 36 Mill.	Simult. 395 000 at 64 kbit/s, in total 1.8 Mill. users at 64 kbit/s		
Transmission rate [Mbit/s]	UL: up to 2, DL: up to 64	Up to 155.52	2.048–51.84	0.016–60
Frequency UL [GHz]	28.6–29.1 and 27.6–28.4[a]	28.6–29.1 & 29.5–30	47.2–50.2	14–14.5
Frequency DL [GHz]	18.8–19.8 and 17.8–18.6[a]	18.8–19.8 & 19.7–20.2	37.5–40.5	11.7–12.7

[a]Motorola in 1998 has joined the Teledesic consortium

Table 14.4: Broadband satellite systems concentrating on data transfer (continued)

System	Teledesic	Celestri LEO	M-Star	SkyBridge
Bandwidth UL+DL [MHz]	2600	2000	6000	1500
Frequency GW–Sat [GHz]	In above band	In above band	In user-sat. band	12.75–13.25 and 13.75–14 and 17.3–17.8
Frequency Sat–GW [GHz]	In above band	In above band	In user-sat. band	10.7–11.7
Terminal	Fixed terminal (Ant.-Ø 0.16–1.8 m)	Fixed terminal	Terminal (Ant.-Ø 0.66–1.5 m)	Fixed terminal
Trans. power terminal [W]	0.01–4.7			
Services	Multimedia, video, data	Multimedia, video, data	Data, high-rate network connection	Multimedia, video, data
Coverage area	95 % of area and 100 % of population	70° S–70° N (with Celestri[b] global)	Global	Global
System costs [Mill. US$]	9	12.9	6.1	3.5
Start comm. operations	2002	2002	2002	2002
Useful life [years]	10	8		
Licence from FCC	Granted March 1997	Application accepted	Application accepted	Application accepted

[a]Gigalinks
[b]The Celestri system consists of the Celestri-LEO, M-Star and Millenium (GEO systems)

Table 14.5: Broadband GEO satellite systems concentrating on data transfer

System	Spaceway	Express-way	Millenium (Celestri-GEO)	Astrolink	GE Star	PAC 1–8 and Galaxy Sat.	Inmarsat-3
Company	Hughes	Hughes	Motorola, Vebacom	Lockheed Martin Comm.	GE Americom	PanAmSat	Inmarsat Contractors
Orbit	GEO	GEO	GEO	GEO	GEO	GEO	GEO
Path altitude [km]	35 786	35 786	35 786	35 786	35 786	35 786	35 786
No. of satellites	9 (Type HS702)	14	4	9	9	16	5
No. of different satellite positions				5	5		
No. of ground stations						7	
No. of cells per sat.	48				44		
Intersatellite links	Yes			Yes	No		
Access method	UL: TDMA/ FDMA DL: TDM			UL: TDMA/ FDMA DL: TDM			
No. of channels per sat.	276 480 at 16 kbit/s			64 at 125 MHz, 4 at 250 MHz			
No. of channels in total system	248 832	588 000		576 at 125 MHz			
Transmission rate [kbit/s]	16–6000			16–9600	384–40 000		
Frequency uplink [GHz]	28.35–28.6 and 29.25–30	V- and Ku-band	29.5–30	28.35–28.6 and 29.25–30	28.35–28.6 and 29.25–30	5.925–6.425	
Frequency downlink [GHz]	19.7–20.2 + 5 GHz in-band 17.7–18.8	V- and Ku-band	18.55–18.8 and 19.7–20.2	19.7–20.2 + 5 GHz in-band 17.7–18.8	19.7–20.2 + 5 GHz in-band 17.7–18.8	3.7–4.2	

Table 14.5: Broadband GEO satellite systems concentrating on data transfer (continued)

System	Spaceway	Express-way	Millenium (Celestri-GEO)	Astrolink	GE Star	PAC 1-8 u. Galaxy Sat.	Inmarsat-3
Bandwidth UL+DL [GHz]	2[a]		1.25	2[a]	2[a]	1	
Frequency GW–Sat [GHz]	In-band user-sat.			In-band user-Sat.	In-band user-sat.	14.0–14.5	
Frequency Sat–GW [GHz]	In-band user-sat.			In-band user-sat.	In-band user-sat.	11.7–12.2	
Terminal	Terminal (Ant.-Ø 0.7 m)	Terminal	Terminal	Terminal	Terminal	Terminal	Terminal
Services	Multimedia, video, data, voice		Video, data broadcast	Data, video, Internet	Data, video, audio	Voice, data, fax, multi-media, RDSS	Voice, data, fax, Pos. det.
Coverage area	Continents ex. parts of Russia	Global	America	Global	America, Europe, Asia, West Pacific, Caribbean	Global	70°§–70° N
System costs [Mill. US$]	3.2	3.9	2.3	9	4		0.69
Price of terminal [US$]	1000			Several 100			
Start comm. operations	2000		2001	2000		In operation	since 1997
Useful life [years]	15					11	13
Licence from FCC	05/97			05/97	05/97		

[a]Several systems share frequency band

14.1.2 Satellite Organizations

Until now, the commercial operations of international satellite systems have almost exclusively been carried out by state-run operating companies such as Intelsat, Eutelsat and Inmarsat (see Appendix B). The primary responsibility of Intelsat is to provide regular radio services worldwide, namely telephony and data links as well as the transmission and distribution of TV and radio programmes. Satellites for regular radio services specifically for Europe are operated by Eutelsat *(European Telecommunications Satellite Organization)*. The organization Inmarsat (*International Maritime Satellite Organization*) was set up to provide a worldwide maritime satellite radio service.

These operating organizations are all structured in a similar way. They are based on international agreements that were ratified as national law by the participating countries. These member countries establish the objectives of the organization and entrust national telecommunications authorities or operating companies with creating satellite earth stations. The satellite organizations of the countries have no customer contact, but instead provide satellite capacity and, among other things, specify the frequency ranges available for the operation of satellite systems. The operating companies design the satellite services for the customers.

State-run operators of satellite systems are competing against private operators, international companies and consortia, which mainly concentrate on implementing new concepts such as worldwide satellite-based mobile communications systems. These organizations focus on providing almost complete wide-area coverage of the earth's surface using *Low Earth Orbit* (LEO) and *Medium Earth Orbit* (MEO) satellite systems. Some of these satellite systems are introduced in the following sections, accompanied by advantages and disadvantages of geostationary and non-geostationary satellite systems.

14.1.3 Satellite Orbits

Path height is possibly the most important aspect of describing satellite systems. As shown in Figure 14.1, the categorization breaks down into LEO, MEO, *Highly Elliptical Orbit* (HEO) and *Geostationary Orbit* (GEO) systems. LEO system satellites are located between a height of 200 km and the inner Van Allen belt (named after James Alfred van Allen, b. 1914, an American physicist. The Van Allen belts are two radiation belts of the earth (zones with high-intensity ionizing radiation)) at a height of 1500 km. The satellites of the MEO systems are located between the two Van Allen belts, between 5000 and 13 000 km. LEO and MEO satellites are in circular orbits. In addition to the circular systems, there are also HEO systems that are able to achieve a better coverage of densely populated areas using elliptical orbits without having to forfeit the advantages of low orbits.

During the 1960s, developers of the first systems started discussing the advantages and disadvantages of LEO, MEO and GEO systems. The technical

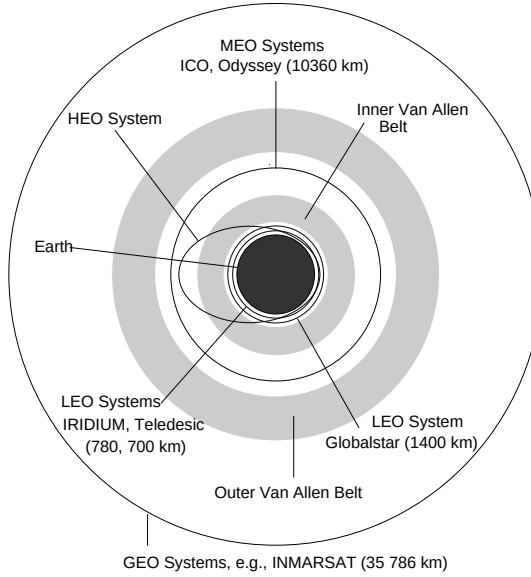


Figure 14.1: Orbits of satellite systems

advantages of LEO systems, such as lower signal propagation times and path loss, were weighed up against the practicability of geostationary systems. The first test satellites such as TELSTAR operated in the lower orbits. Before the mobile applications of satellites were considered as a possibility, GEO systems were viewed as being superior. The satellite launches, particularly during the 1960s, were still unreliable and global coverage was less important than it is today, so the GEO systems, which can manage with only one satellite, were favoured. With interest in mobile communications over satellites increasing, there was a renewed effort to speed up the development of systems with low earth orbits.

14.1.4 Elevation Angles and Coverage Zones

With LEO satellite systems the radio coverage zone on the ground is divided, following the usual cellular principles, into individual cells by a satellite to allow the reuse of frequency bands within the entire coverage zone. The size of the illuminated zone is established by the minimum elevation angle ϵ_{min}, which can be determined from the maximum possible distance between a mobile terminal and the satellite (see Figure 14.2).

The elevation ϵ is

$$\epsilon = \arccos\left(\frac{r_e + h}{d}\,\sin\phi\right) \tag{14.1}$$

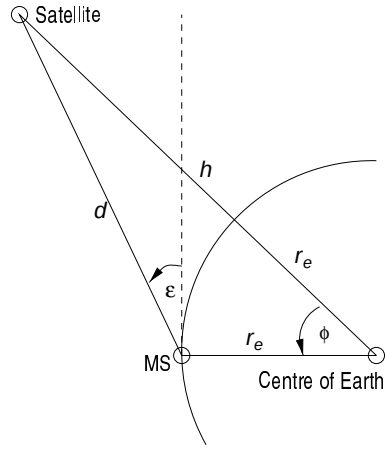


Figure 14.2: Elevation angle in LEO systems

Theoretically, elevation angles of 0° are possible. However, there is a tendency to maintain the minimum elevation angle ϵ_{min}, which for practical reasons is typically at 10° to avoid larger areas without radio coverage because of shadowing [12]. The radius of a coverage zone depends on the earth's radius r_e and the path height h of the satellite above the earth's surface:

$$r_{cov} = r_e \left[\arccos\left(\frac{r_e}{r_e + h} \cos \epsilon_{min} \right) - \epsilon_{min} \right] \tag{14.2}$$

With a given maximum distance d_{max} and a minimum elevation ϵ_{min}, the orbit height works out as

$$h = \sqrt{d_{max}^2 + 2 d r_e \sin \epsilon_{min} + r_e^2} - r_e \tag{14.3}$$

The area covered by the satellite is then

$$A_{cov} = 2\pi r_e^2 (1 - \cos \phi) \tag{14.4}$$

These equations can be used to determine the radius of the coverage zone, the orbit speed of the satellite and the travel time around the earth. If the number of cells per coverage zone is available then the size of the cells can be established and an initial prediction made about the amount of time a user spends in a cell. The above values for some orbit heights are given in Table 14.6. For example, an IRIDIUM satellite (at a 780 km height) travels at approximately 28 500 km/h over a stationary earth point, and is only visible from there for about 10–12 min (see Figure 14.39).

14.1.5 Frequency Regulation for Mobile Satellites

Frequency allocation is an important aspect in the planning of satellite systems. Since it is the international authorities who allocate the frequencies,

Table 14.6: Circular satellite orbits

Orbit height [km]	Travel time	Orbit speed [km/s]	Radius of coverage zone [km]
400	1 h 32 min	7681	1333
600	1 h 36 min	7569	1737
800	1 h 40 min	7463	2065
1000	1 h 45 min	7361	2349
1200	1 h 49 min	7263	2592
1400	1 h 54 min	7168	2805
2000	2 h 07 min	6906	3320
35786	23 h 56 min	3075	6027

operators of global satellite systems must negotiate licensing agreements with all the countries in which they want to offer their services. Member states of the International Telecommunications Union (ITU) are committed to the resolutions of the World Radio Conference (WRC).

Commercial satellite radio is currently using frequencies mainly in the C- and Ku-bands. In the C-band these are 5.925–6.425 GHz range for the uplink and 3.7–4.2 GHz for the downlink; and in the Ku-band they are 11.7–12.2 GHz for the downlink and 14–14.5 GHz for the uplink. Because of the growing interest in satellite communications, additional frequencies in the S- and the K/Ka-bands were reserved for satellite radio at WRC 1992. The frequency bands 1980–2010 for the uplink and 2170–2200 MHz for the downlink were made available in the S-band for the satellite segment of UMTS. With the availability of 3.5 GHz of bandwidth each (27.5–31 GHz and 17.7–21.2 GHz), this means that considerably more bandwidth is now available in the Ka-band than was provided in the previous frequency bands.

Since the initial networks are mostly being built by American companies, applications for licences are first made to the American regulatory body, the *Federal Communications Commission* (FCC). These licences then establish the pattern for regulation in other countries. Accordingly, the first frequencies allocated by the FCC for mobile satellite systems were in the 1610–1626.5 MHz band. As is shown in Figure 14.3, 5.15 MHz was allocated to the IRIDIUM-TDMA system and 11.35 MHz to the CDMA systems to be shared by Odyssey and Globalstar. In case only one of the CDMA systems becomes operational, the frequency band 1616.25–1621.35 MHz will be renegotiated.

14.2 Geostationary Satellite Systems

The geostationary orbit is at an altitude of 35 786 km above the earth's surface. Satellites in this circular orbit have an angular velocity that is the same as the rotational speed of the earth. Therefore to an observer on the ground the satellite is at in a fixed position. Because a geostationary satellite can

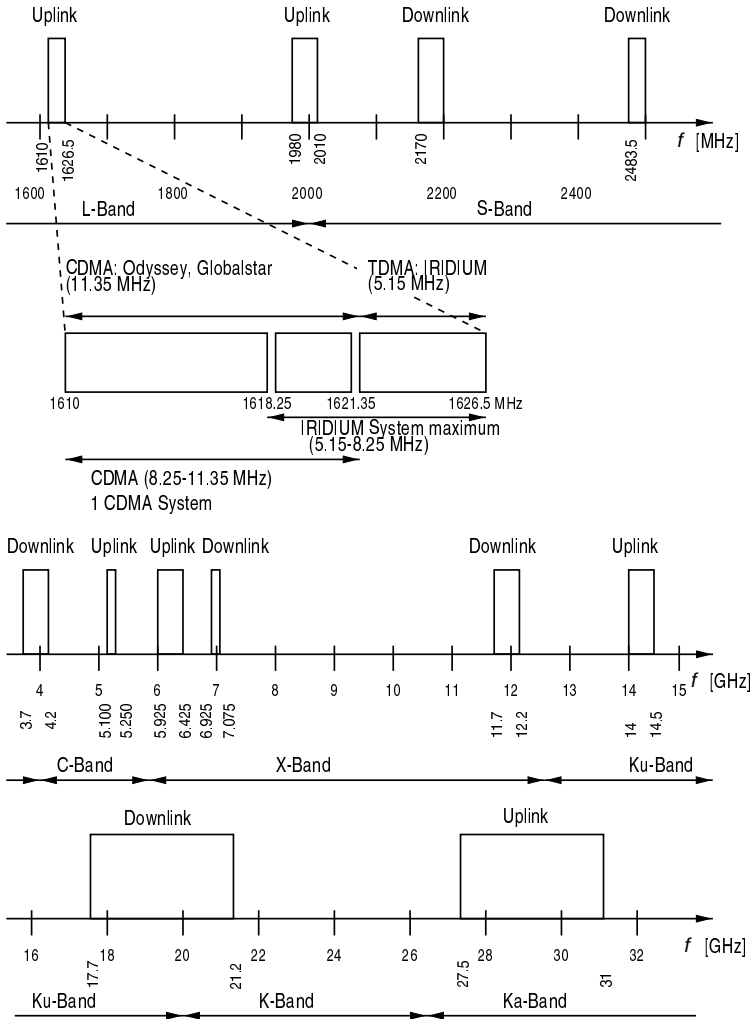


Figure 14.3: Frequencies for satellite radio

drift up to 60 km from its position owing to the influence of other celestrial bodies, positional corrections are essential for the satellite. Ground stations can follow these positional changes better than mobile stations.

The advantages of geostationary satellite systems compared with non-geostationary systems include:

- Simple configuration.
- Coverage of large areas with only one satellite. Large distances can easily be bridged.

- Minimal routing problems in the coverage areas because of coverage zones of the satellites.
- The low relative movement of the earth only causes insignificant Doppler shifts of the signals.

The disadvantages of using a GEO system are as follows:

- A need for high transmitter power and large receiving antennas (because of the distances attenuation is at approximately 200 dB).
- Regions at a very high geographical latitude cannot be serviced. The elevation angle, which decreases the higher the latitude, falls below an elevation of 10° at a degree of latitude $\varphi > 72°$, so that perfect connection can no longer be guaranteed. For example, in Germany (47°–55° North) elevation angles between 20° and 30° are achieved.
- Coverage to urban areas is a problem because of shadowing. Reception in these degrees of latitude is only possible through the use of directional antennas.
- Signal propagation times are large because of the large distances. One way propagation delay is approximately 125 ms. Therefore ARQ (automatic repeat request) data link control protocols with error handling designed for wired networks are not suitable.
- It is expensive to launch satellites into a geostationary orbit compared with lower orbits.

These disadvantages are dominant as far as utility of GEOs are concerned for mobile communication. Such orbits are primarily suitable for stationary services.

European telecommunications network operators use the Inmarsat satellite system operated by Inmarsat, an organization established in 1979 with its headquarters in London and supported by 63 member countries. The mobile radio services offered by Inmarsat were initially restricted to maritime applications. Inmarsat is still the only supplier today that offers global mobile satellite communications on the sea, on land and in the air.

The Inmarsat satellite system consists of 11 satellites: 4 operating satellites and 7 reserve satellites. Reserve satellites can be leased from different organizations.

All Inmarsat satellites are located in a geostationary orbit. The four operating satellites are distributed in the equatorial plane so that they can cover the entire globe, except for the polar regions. The reserve satellites are located in positions in the immediate vicinity of the operating satellites so that they can be replaced if necessary.

The four coverage zones of the operating satellites are the West Atlantic, the East Atlantic, the Indian Ocean and the Pacific. As can be seen in Figure 14.4,

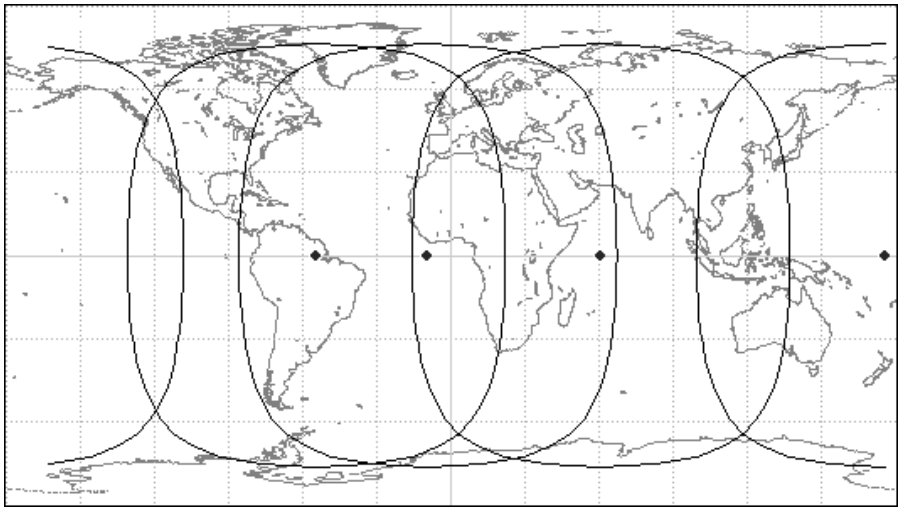


Figure 14.4: Configuration of Inmarsat operating satellites

there is a difference in how much the coverage zones overlap each other. The largest overlap is between the West Atlantic and the East Atlantic. The reason for this is to provide better servicing of the large traffic between America and Europe.

14.2.1 Inmarsat-A

The Inmarsat-A service caters for the following applications:

- Self-dialled telephone connections from/to user terminals.
- Self-dialled (half-duplex) telephone connections from/to user terminals.
- Calls to terminals and retrieval of data stored in terminals. The information is provided only if the caller provides unique identification.
- Use of telephone connections to transmit fax messages and data (up to 9600 bit/s with a modem).
- Data transfer from terminals at a data rate of 64 kbit/s.
- Special applications for broadcast transmission, video freezing frames and commentator transmission routes.

Inmarsat-A is the first of six systems being operated by Inmarsat. It went into operation in 1979, and transmits analogue modulated signals. Portable terminals require a parabolic antenna of approximately 1 m diameter and weighing 20–60 kg. With stationary terminals the antennas are adjusted by hand in the direction of the desired satellite.

Table 14.7: Technical data on Inmarsat systems

Inmarsat	A	B	M	C	Aero
Channel access	Aloha	Aloha	S-Aloha	Aloha	Aloha
Modulation	BPSK	QPSK	BPSK	BPSK	BPSK
Coding	BCH	1/2 FEC	1/2 FEC	1/2 FEC	1/2 FEC
Transmis. rate [kbit/s]	4.8	24	3	0.6	0.6

14.2.2 Inmarsat-B

The Inmarsat-B system has been in existence since 1993 as a further development of Inmarsat-A, the major improvement being the change over to digital transmission. The traffic channels transmit digitally using forward error correction. It has greatly improved the quality of voice and data transmission. The frequency band and the capacity of the satellites are being used more economically. Improvement in satellite technology have produced a reduction in the transmission costs for a call. The communications services offered by Inmarsat-B are comparable to those of Inmarsat-A (see Table 14.7).

14.2.3 Inmarsat-C

Inmarsat-C was developed for worldwide telex and data transmission (X.25) from small mobile terminals. The following applications are some of those available:

- Data transmission in X.25 mode from/to mobile terminals (*Mobile Earth Station*, MES) at data rates of up to 600 bit/s.
- Text transmission from the MES in X.25 mode and output of messages on fax equipment.
- Data transmission in X.400 mode to mailboxes.
- Position location of MES by a central switching centre.
- Formation of closed user groups through the allocation of appropriate telephone numbers.

The net bit rate transmitted in both directions is 600 bit/s. Dynamic intermediate storage (*store-and-forward* transmission) located in the earth station guarantees reliable data transfer. An earth station automatically makes another request for erroneous or missing data blocks. The data blocks are not forwarded to the receiver until the message has arrived error-free at the destination.

Signals are scrambled and spread by the sender to reduce tramsmission errors that occur owing to fast fading. The length of messages is limited to 32 kbytes, primarily because of the intermediate storage.

The terminals used in the Inmarsat-C service are relatively small and weigh approximately 5 kg. They consist of a transmit/receive antenna unit for all directions, a transmitter/receiver, an operating unit and a printer (if required). The power consumption in receiving mode is 25 W and in sending mode 100 W pulsed.

14.2.4 Inmarsat-Aero

The Inmarsat-Aero service, which was introduced in 1992, enables communication from and to airplanes (see also the TFTS service in Section 4.1). Two types of antenna are available:

- Low-gain antennas (0 dBi) for data transmission, mainly for internal operating purposes for airplanes.
- High-gain antennas (12 dBi), which allow higher data rates, thereby enabling voice links to the ground.

14.2.5 Inmarsat-M

The Inmarsat-M service, a digital expansion of the Inmarsat-A service, has been available since 1993. Inmarsat-M terminals are small and considerably lighter in weight than Inmarsat-A terminals. Mobile radio users can make use of the following communications services:

- Telephone
- Fax
- Data transfer (up to 4800 bit/s)
- Variations of the Inmarsat-C service

Because of the low transmission rate of 3 kbit/s, the voice quality with Inmarsat-M is poorer than in connections with Inmarsat-A and -B. Because of the terminal's smaller antenna (30–50 cm) and lower transmitter power, the gain in the Inmarsat-M system is 8 dB less than with Inmarsat-A. It was necessary to reduce the data rate to 3 kbit/s to achieve good transmission quality despite the poor power budget.

14.3 Non-Geostationary Satellite Systems

The advantages of LEO and MEO/ICO systems compared with GEO systems are:

- Lower transmitter power required because of low orbit heights.
- Higher average elevation angle. Because of the large number of satellites, the satellite with the shortest distance to the mobile radio user can always be selected.

- High operating reliability through increased redundancy.
- Good coverage of regions with highest geographical latitude (e.g., polar regions).
- Small signal propagation time.

However, there are also disadvantages due to the lower orbits:

- Short link duration to satellites due to changing elevation angles.
- Smaller coverage area per satellite.
- Complex system control.

At their nearest point to earth (*perigee*), HEO satellites approach up to a couple of hundred kilometres from the earth's surface in order to be able to reach the furthest distance (*apogee*) in the area of the geostationary orbit. They are used there for communication purposes because that is where their orbit speeds are the lowest and a satellite therefore remains visible for a long time from a certain point. Satellites in these orbits are always passing through the Van Allen belt and are consequently subject to increased radiation.

14.3.1 ICO

In September 1994 Inmarsat founded a company called ICO (Intermediate Circular Orbit) to introduce and operate the Inmarsat-P21 project (see Table 14.1). ICO's biggest investors are INMARSAT (150 million US$) and Hughes Space and Communications (94 million US$). The ICO system is the only mobile satellite system that has been initiated by an operator of an existing satellite system.

ICO will be using 10 satellites in two orbits with 45° inclination to the equatorial plane (see Figure 14.22; the coverage zones of the ICO system are shown in Figure 14.5). Maximum use will be made of path diversity. It is very probable that two satellites will be visible at the lower elevation, and the satellite with the best channel quality will be selected. It is anticipated that the greatest proportion of users of the ICO system will be accessing through GSM dual-mode terminals, and will use the satellite system only if the GSM network cannot guarantee an adequate radio connection.

ICO will be dispersing 12 earth stations (*Service Area Nodes*, SAN) across the globe (see Figure 14.6), and these will be interconnected over a broadband network (P-network). Because there are no plans for links between satellites, the P-network will forward wide-area connections to the nearest network gateway, which will create the link to other networks such as PSTN, PLMN and PSDN.

The *terrestrial terminal and control* (TT&C) stations are responsible for operation and maintenance of the satellites, as well as for control of the configuration through position tracking. Data (such as battery state and position)

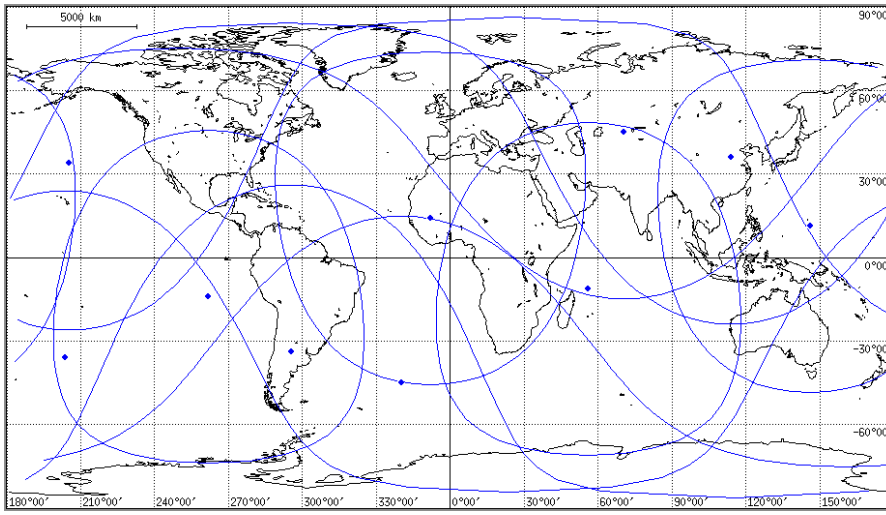


Figure 14.5: Coverage zones of the ICO system

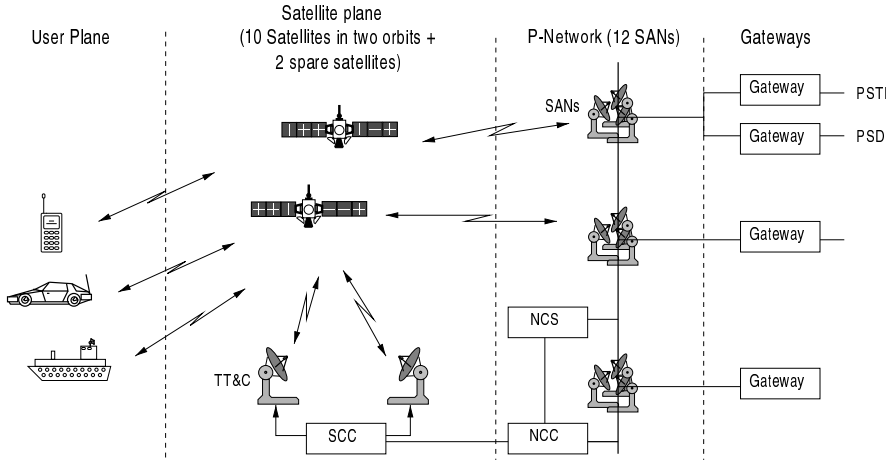


Figure 14.6: Architecture of the ICO system

are recorded. The *Network Control Stations* (NCS) check traffic loads and the connection quality of the different links, and are responsible for channel allocation. The *Network Control Centre* (NCC) carries out system management for the entire system [4, 9].

ICO is planning to use 10 MHz in each of the frequency bands 1980–2010 MHz and 2170–2200 MHz for the uplink and the downlink of satellite user connections (see Figure 14.3). Frequencies from the bands between 5100 and 5250 MHz as well as between 6925 and 7075 MHz are planned for the

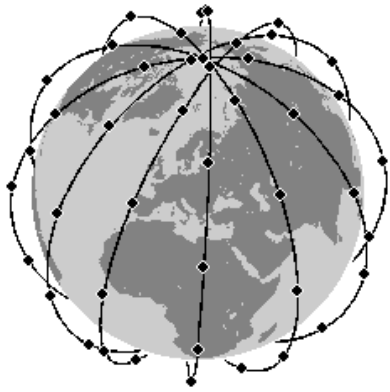


Figure 14.7: Orbits of the IRIDIUM satellite system

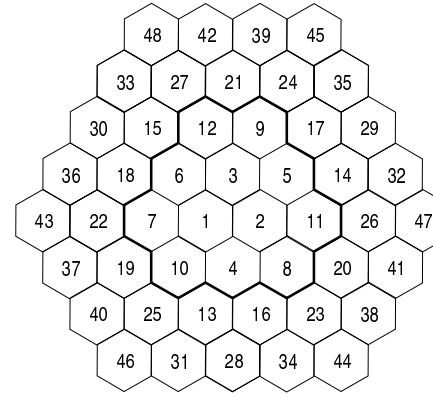


Figure 14.8: Idealized cell structure of the IRIDIUM system

satellite-to-SAN link; see Figure 14.6. A TDMA/FDMA method with six time slots per carrier will be used in the uplink and the downlink and 163 cells per satellite. The satellites are designed so that they can carry 4500 voice channels.

14.3.2 IRIDIUM

In June 1990 Motorola Inc. announced its development of the IRIDIUM system. IRIDIUM is a LEO system based on a constellation of 66 satellites. The satellites cruise at a height of 780 km on polar orbits with an orbit inclination of 86° around the earth. As can be seen in Figure 14.7, 11 satellites are always arranged on six orbits so that the entire surface of the earth is covered (see Figure 14.9).

The coverage zones of each satellite are divided into 48 cells through the use of 48 *spot beams*, with the frequencies reused according to a 12-cell cluster configuration; see Figure 14.8. The satellite antennas are *phased-array antennas* which compensate for the difference in path loss between inner and outer cells using the different antenna gain of the respective radiation diagram.

The first satellites were launched into orbit in July 1997. Commercial operation has been opened in November 1998. The IRIDIUM system uses *inter-satellite links* (ISL), for which 200 MHz are being provided in the Ka-band, as is also the case for the gateway satellite link (see Figure 14.3). The ISLs reduce the number of gateways required. Furthermore, coverage over oceans could not be guaranteed without ISLs. ISLs exist to each of the four adjacent satellites. The difficulty is with interorbital links, because the distances and directions for these links are constantly changing. The routing in the ISL-based satellite network is controlled from the NCSs on the ground.

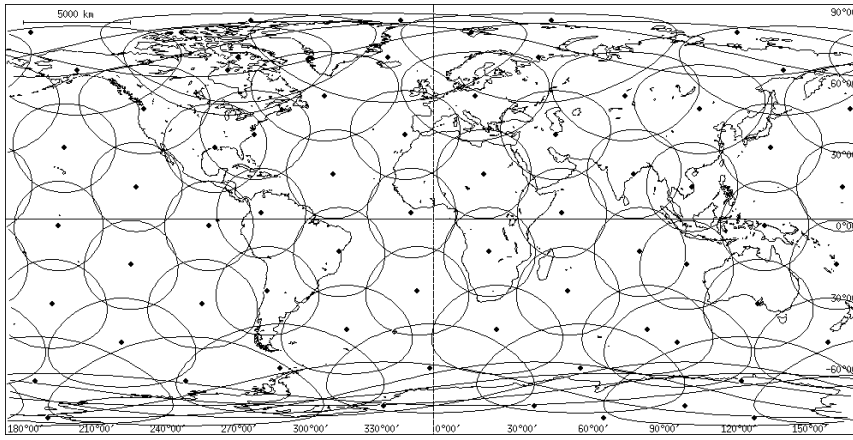


Figure 14.9: Coverage zones of the IRIDIUM system

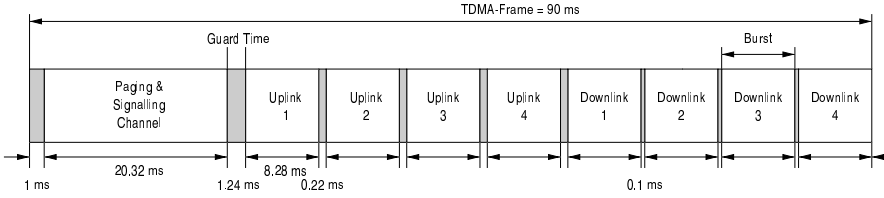


Figure 14.10: TDMA frame in IRIDIUM

The mobile terminals are pocket-sized dual-mode devices that enable connections to be established to satellites as well as to a terrestrial mobile radio network. The transmission rates are 4800 bit/s for speech and 2400 bit/s for data. Multiple access is through the use of a combined FDMA/TDMA/TDD procedure. The TDMA frame planned is 90 ms long, and contains four uplink and four downlink channels as well as a channel for signalling and paging (see Figure 14.10).

The IRIDIUM system uses a QPSK modulation technique with 50 kbit/s. Frequency planning for IRIDIUM before the allocation of an additional 5.15 MHz by the FCC is shown in Figure 14.11. The channel bandwidth is 31.5 kHz and the frequency spacing is 41.67 kHz [8].

The 5.15 MHz frequency band allocated by the FCC to the IRIDIUM system is divided into 124 carriers, each containing four duplex channels, thereby providing 496 channels. Since the frequencies are reused four times per satellite (12 cells per cluster, 48 cells per satellite) it is theoretically possible that 1984 connections could exist simultaneously per satellite because of the given cell structure. However, owing to the limited battery power of 1400 W available a satellite can only maintain 1100 connections at a time.

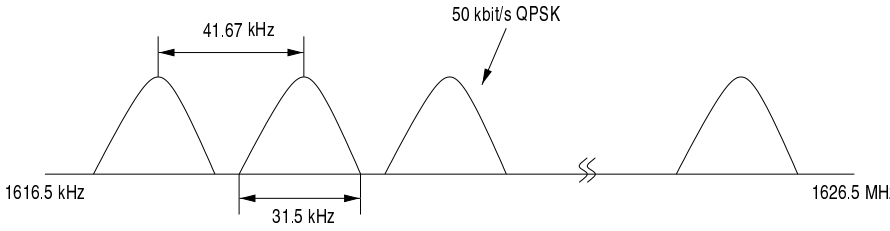


Figure 14.11: Frequency plan in IRIDIUM system with 10 MHz bandwidth

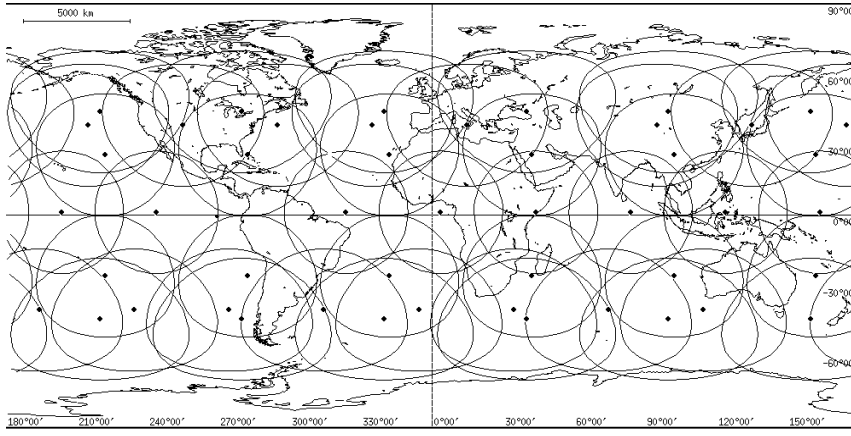


Figure 14.12: Coverage zones of the Globalstar system

14.3.3 Globalstar

In contrast to IRIDIUM, which was developed for worldwide coverage, Globalstar, another representative of LEO systems, supplies coverage only to areas between a 70° northern and southern latitude (see Figure 14.12). Its main purpose is to provide coverage to the industrial countries. Direct (intersatellite) links between the satellites are not planned. The transmission and access technologies closely follow the standard TIA-IS95; see Section 4.3.

Forty-eight satellites circle eight circular orbits at a height of 1400 km (see Figure 14.13). Two satellites are constantly visible below an elevation angle of at least 10° in the coverage area of the system. The coverage area of a satellite is divided into 16 cells, which are grouped into six or nine segments in a concentric arrangement around the foot of the satellite (see Figure 14.14). A synchronous CDMA channel allocation method is used to facilitate handovers. Communication between mobile station and satellite is only possible if an earth station is located in the area being illuminated by the satellite, because the satellites have only a transponder function and no onboard switching at all. The signal of a connection between two mobile stations must therefore cover

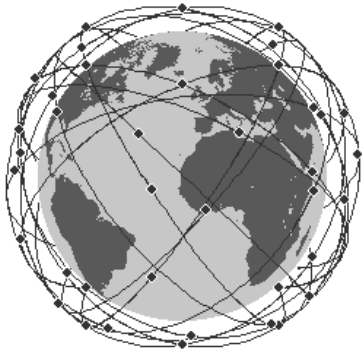


Figure 14.13: Orbits of the Globalstar satellite system

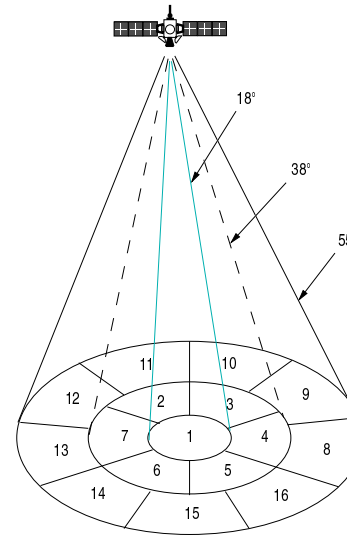


Figure 14.14: Cell layout on Globalstar downlink

twice the distance (two hops). The signal propagation times are short anyway (approximately 70 ms) because of the low orbital height. For example, in North America six earth stations are required to cope with the traffic volume.

14.3.4 TELEDESIC

There has been a sharp increase in the demand for broadband communications systems during the last few years. However, coverage with fibre optics will not be widespread in the world, even in the next few years. The LEO and MEO satellite systems offer services with up to a 9.6 kbit/s transmission rate only, and, this naturally does not meet the requirements for multimedia data transmission. Services such as video conferencing, telelearning and video telephony require a much larger bandwidth.

TELEDESIC, a company founded in June 1990 by Bill Gates, the owner of Microsoft, and Craig McCaw, one of the owners of the largest mobile radio network operators worldwide, is planning a satellite system that will be offering broadband services up to 2 Mbit/s transmission rate per terminal over satellite (see Table 14.4). The first satellites are to be launched into their orbits in 1999, and operations are to start in 2002.

The TELEDESIC system is supposed to be compatible with fibre-optic networks and to provide the quality and services of fibre-optic technology in remote, rural areas at the price of fibre optics in urban areas.

In contrast to other systems, TELEDESIC will operate in the Ka-band (30 GHz downlink, 20 GHz uplink) in order to provide broadband services.

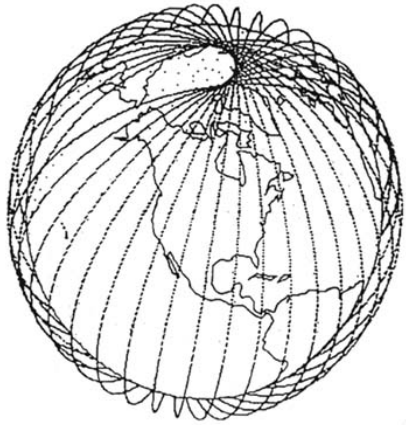


Figure 14.15: Satellite orbit in the TELEDESIC system

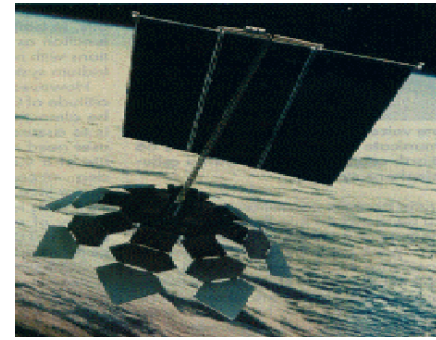


Figure 14.16: A TELEDESIC satellite

Communication in the Ka-band is heavily affected by attenuation caused by rain.In the TELEDESIC system 924 satellites (840 operating and 84 in reserve) operate in 21 orbits at 700 km above the earth's surface. The orbit inclination is 98° so that there is almost global coverage except for a small gap at the poles (see Figure 14.15), this also reduces the loss due to rain. Four reserve satellites are distributed evenly over each orbit. If one of the satellites malfunctions, the other operating satellites and the closest reserve satellite spread out over the respective orbit. Because of the large number of satellites, there is a minimal elevation of 40°, which helps to reduce the effects of heavily elevation-dependent rain attenuation. The TELEDESIC system has been developed to offer 99.9 % availability.

Initially eight satellites are to be launched into orbit at the same time in order to keep the number of rocket launches to a minimum. Weighing 700 kg, the satellites will be much lighter in weight in comparison with other systems. Figure 14.16 shows a TELEDESIC satellite [13]. It consists of a quadratic *solar panel* of 12 m edge length, with a total of 24 plates arranged in a daisy shape with eight *daisy leaves*. The plates contain the phased-array antennas. The octagonal baseplate contains eight pairs of ISL antennas along with a drive unit. The power supply unit is attached to the top of the solar panel. The satellites are so designed that they can be launched by as many as 20 different types of rockets.

The TELEDESIC network uses links in the 60 GHz band for communication between satellites (ISL) in order to feed connections as closely as possible to the destination user in the fixed network; intersatellite links exist to eight neighbouring satellites. The network transmits in packet-switched mode and supports broadband communication based on the ATM standard (*Asyn-*

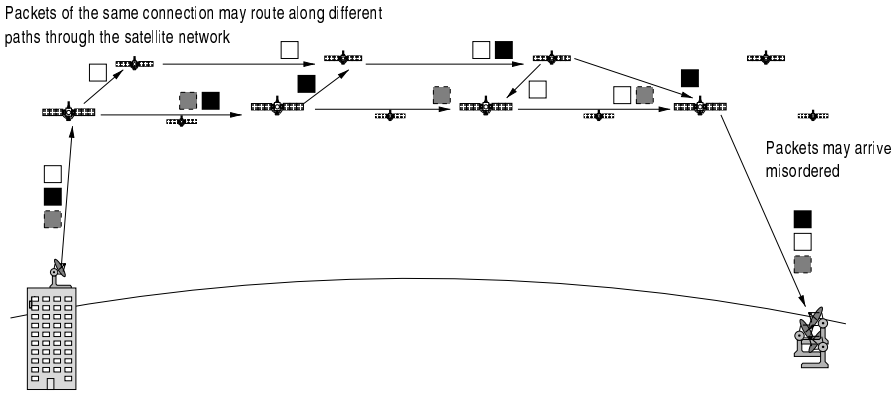


Figure 14.17: Packet-switched connectionless TELEDESIC network

chronous Transfer Mode). The data packets are 512 bits long. The packet header contains the address and the sequencing information, which are used to restore the sequence at the receiver, as well as fields for error protection and user information. Deviating from the ATM standard, TELEDESIC uses a connectionless protocol (see Figure 14.17). The packets that arrive at the receiver over different paths must therefore be buffered and restored again in the right sequence. Each node independently routes the packet (*Distributed Adaptive Routing Algorithm*) over the path that is currently able to guarantee the minimum delay.

TELEDESIC is also breaking new ground in its approach to cell layout. Cells in the TELEDESIC system are ground-based, which makes cell planning easier but on the other hand requires more complicated antenna technology. The earth's surface is covered by 20 000 *supercells*. These supercells are divided into bands, which at the equator contain 250 supercells but towards the poles contain fewer supercells so that the North-South boundaries shift between the bands. As Figure 14.18 shows, a supercell consists of nine square small cells.

Ground-based cells require much less administration than satellite-based cells. The TELEDESIC system uses *steering phased-array antennas* (see Section 14.4.1.1). As illustrated in Figure 14.20, the satellite controls its spot beam and compensates for the movement of the satellite and earth when the satellite passes over the cell cluster. In contrast with satellite systems without ground-based cells, there is no *interbeam handover* between antennas of the same satellite.

A supercell is supplied by an antenna which serves the nine cells in consecutive order (SDMA). This cycle which is made up of transmit times of 2.276 ms and guard times of 0.292 ms is 23.111 ms long (see Figure 14.19). The cells are scanned according to the numbers given in Figure 14.18. The combina-

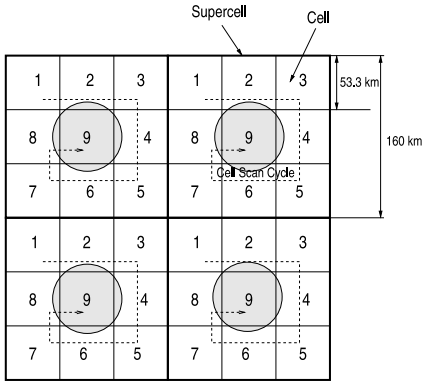


Figure 14.18: Cell structure of the TELEDESIC system

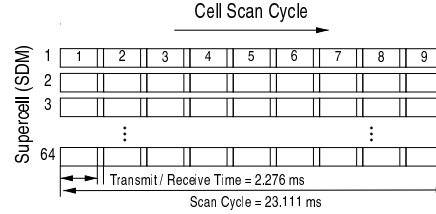


Figure 14.19: Cell scan cycle

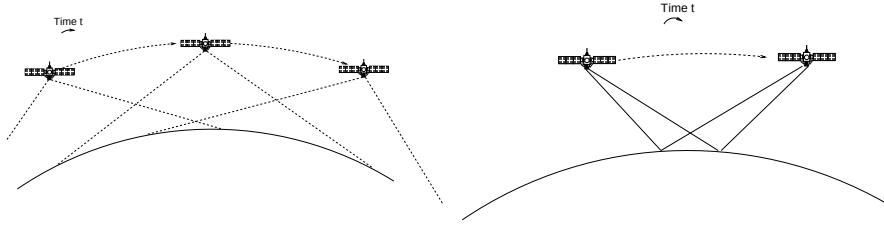


Figure 14.20: Left: Coverage zone which moves with the satellite. Right: Ground-based cells

tion of a large number of satellites and the different multiplexing procedures produces a high level of system capacity. With the TELEDESIC system each frequency used is repeated 350 times. Each satellite can scan 64 supercells simultaneously. Each cell has 1800 channels with a 16 kbit/s transmission rate. A 400 MHz bandwidth is being planned for both the uplink and the downlink. As illustrated in Figure 14.21, the 400 MHz on the uplink will be distributed over 1800 FDM channels. The time slot belongs to the 16 kbit/s channel, and a terminal totally occupies one or more channels for the duration of a call. On the other hand, an ATDMA ($A = adaptive$) method is used on the downlink. The channels are not allocated permanently, because of the connectionless transmission. A terminal must therefore read all the packets and check the headers to determine whether the packet belongs to its own connection before the sequence of the packets is restored. The packets are separated from one another through certain bit sequences to enable detection of the beginning and end of a packet. In this connection, the satellite always only transmits as long as packets are still in the queue.

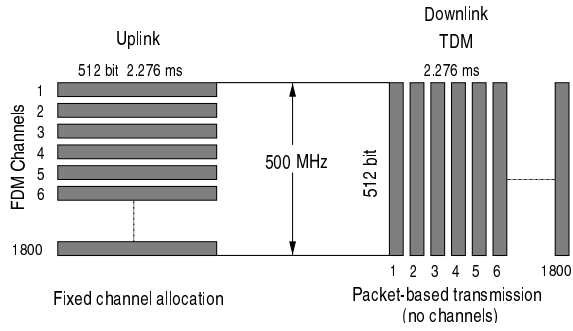


Figure 14.21: Channel allocation in TELEDESIC

A user connection can occupy up to 128 channels, so data rates between 16 and 2048 kbit/s are supported. However, data rates of 155.52 Mbit/s to 1.24416 Gbit/s are supported for special applications.

Fixed and mobile terminals are planned for the TELEDESIC system. The transmitter power of the terminals varies between 0.01 W and 4.7 W, and the antenna diameter between 16 cm and 1.8 m, depending on climatic conditions, requirements for availability and transmission rate [16].

14.3.5 Odyssey

A MEO system, the Odyssey satellite system consists of 12 satellites that travel around the earth at an altitude of 10 370 km on three orbits with an inclination of 55° (see Figure 14.22). The antennas of the satellites can be controlled dynamically and are mainly directed towards coastal areas and the interior. Similarly to the TELEDESIC system, the adaptive antennas are controlled to create ground-based cells (see Section 14.3.4). Because of the approximately six-hour earth circulation time and the relatively large illumination zones, handovers are normally not required during a connection. The coverage zones of the Odyssey system are shown in Figure 14.23.

Connections between users of the system are carried out over 1 of 10 to 11 earth stations distributed around the world. For example, two earth stations (one each on the West and East coast) are sufficient for good-quality operation in North America. A channel-oriented CDMA method is used as the access procedure. The first starts originally were planned for 1997, and initially six satellites were intended to be put into operation in early 1998. This schedule has been delayed, i. e., until end 1998 Odyssey has not been operational at all.

14.4 Antennas and Satellite Coverage Zones

There are many points to be considered in modelling antennas, because of the vital importance of the antenna characteristics to the performance of the

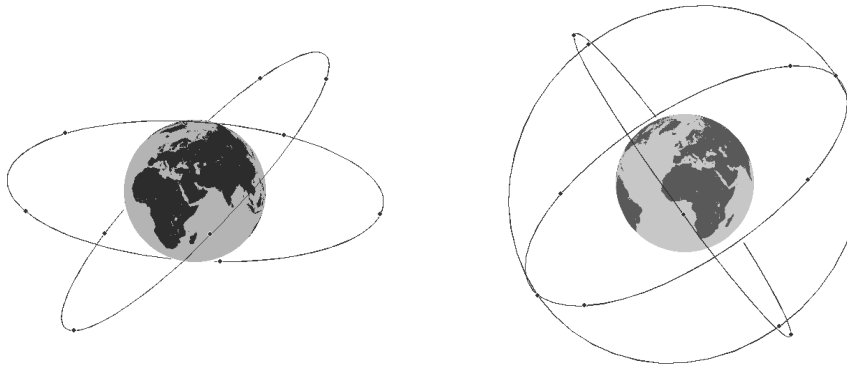


Figure 14.22: Orbits of the ICO (left) and Odyssey (right) satellite systems

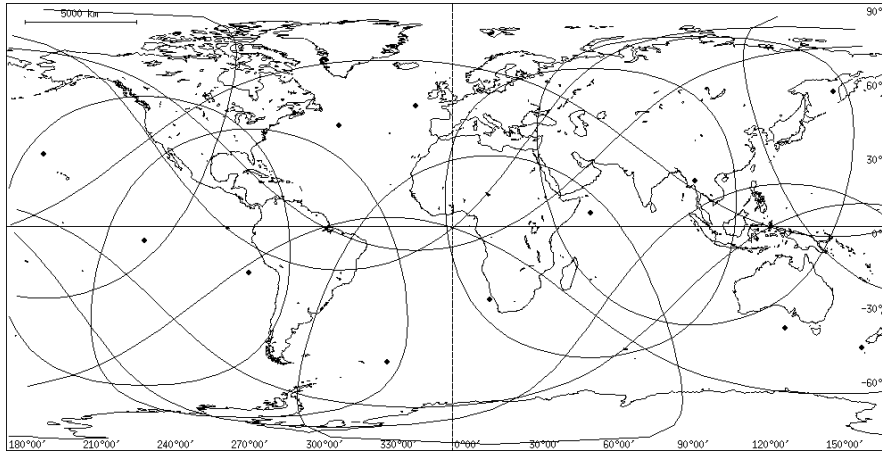


Figure 14.23: Coverage zones of the Odyssey system

system. Antennas determine the size of the cells and consequently the area of the satellite coverage zone. It is desirable that antennas have a high gain in the area of a cell but at the same time that the gain drop dramatically at a cell boundary in order to prevent undesirable interference. Splitting up the satellite coverage zones into cells provides a possibility for the multiple use of frequencies.

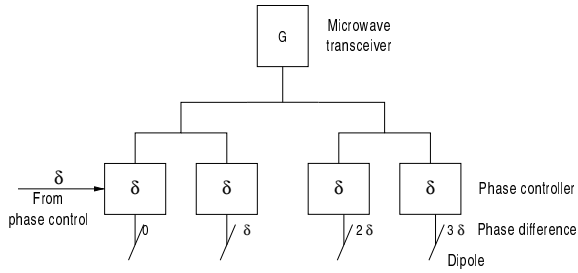


Figure 14.24: Principle of phased-array antennas

14.4.1 Antennas

14.4.1.1 Phased-Array Antennas (PAR)

The satellite systems planned for use in non-geostationary orbits may use *phased-array antennas*.

Individual dipoles act as omnidirectional beams. However, if groups of dipoles are used, a certain directional pattern is created through the phase difference at the receiver, which in a distant field is only dependent on the direction. The directional pattern of a dipole group is dependent on the layout, phases and amplitudes of the dipoles. This is exploited when phased-array antennas are used. The phases of the dipoles are then controlled electrically, as shown in Figure 14.24 for one antenna with four dipoles. This provides for high speeds of beam steering.

Satellite systems with PAR are lighter in weight than mechanically controlled antennas. Phased-array antennas prove to be advantageous, even on the ground, compared with those that swivel mechanically, because with high-gain antennas two antennas are needed owing to the slowness of the swivelling when mechanically swivelled antennas are used in handovers between different satellites.

14.4.1.2 Parameters

The main tasks of a satellite antenna are:

- To receive the desired signal with a given polarization in a specific frequency band.
- To suppress unwanted signal components from other senders transmitting in the same frequency band (co-channel interference).
- To transmit radio waves within a frequency band in the desired direction.
- To keep transmitter power as low as possible through focussing.

Maximum Gain and Antenna Efficiency The maximum gain of an antenna with the equivalent electromagnetic antenna surface A_{eff} is given by:

$$G_{max} = \frac{4\pi}{\lambda^2} A_{eff} \tag{14.5}$$

with λ = wavelength of the transmit frequency. An antenna with a circular aperture and a geometric radiation surface of

$$A = \pi D^2/4 \tag{14.6}$$

(with D = antenna diameter) then has an effective antenna surface

$$A_{eff} = \eta A. \tag{14.7}$$

The antenna efficiency η is dependent on its physical state, and is typically in the region of 0.55–0.75 (0.55 for a parabolic reflector).

According to Equation (14.5), $G_{\max}$ is given by

$$G_{\max} = \eta \left(\frac{\pi D}{\lambda} \right)^2 \tag{14.8}$$

Depending on the angle θ, the antenna gain decreases compared with the maximum gain for $\theta = 0$.

Characteristics Figure 14.25 shows the modelled characteristics of satellite antennas in CCIR-Rep. 558-4. The characteristics of the steep drop are described by

$$G(\theta) = G_{\max} - 3 \left(\frac{\theta}{\theta_o} \right)^2 \quad [\text{dB}] \tag{14.9}$$

For an angle $\theta > \alpha_{SL}$ the assumption is that the antenna gain is very small (-100 dB). This means that signals incident on the antenna from an angle $\theta > \alpha_{SL}$ cannot be received. Another way of looking at this would be that the antenna has no transmitting effects for $\theta > \alpha_{SL}$.

Antennas with spherical characteristics are normally assumed for mobile users. The antenna gain is constant at 0 dB for all angles.

14.4.2 Satellite Coverage Area and Cell Structure

The size and shape of cells produced by satellite antennas is determined by the characteristics of the antenna used. According to Figure 14.25, there are two approaches to defining the angle parameters α of the antenna characteristics [2]:

1. With a satellite antenna with *beam forming*, the parameter angle α of the antenna characteristics can be specified as the spatial angle β between the cell centre and a point further away on the earth's surface

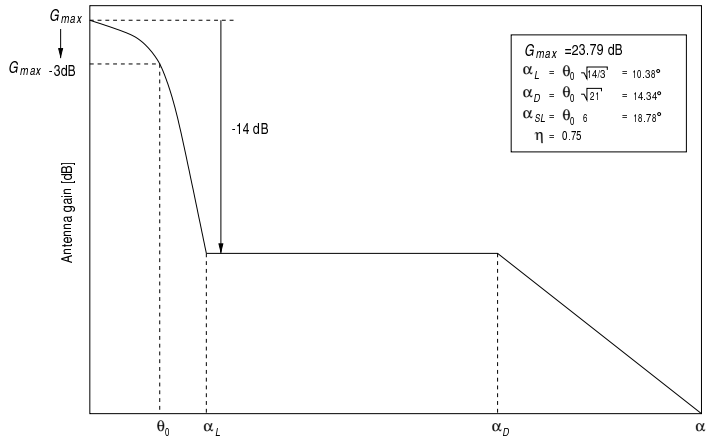


Figure 14.25: Characteristics of satellite antennas (values for IRIDIUM)

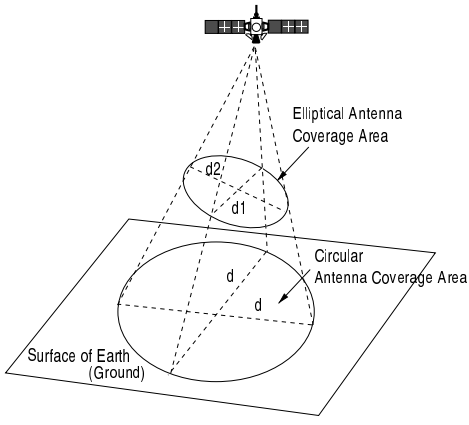


Figure 14.26: *Beam forming* by the satellite antenna

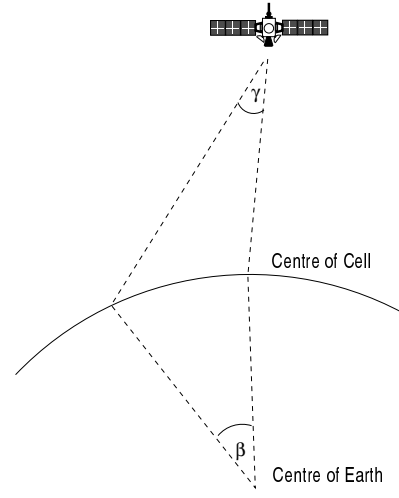


Figure 14.27: Definition of antenna parameter angles β and γ

seen from the earth's centre (see Figure 14.27). The cells are then of circular shape with an antenna gain of $G = G_{max} - 3$ dB on the edge curves (see Figure 14.26).

2. If *beam forming* is not possible on the satellite antenna, the parameter angle α of the antenna characteristics is specified as spatial angle γ seen by the satellite between the cell centre and a point further away on the earth's surface (see Figure 14.27). In this case the cell boundaries and

consequently also the curves with the same antenna gain are no longer circles.

14.4.2.1 Satellite Coverage Area

The coverage area of a satellite is determined by its illumination zone. The coverage area can be of any shape. The assumption is made that a satellite can shape its coverage area (*beam forming*, Assumption 1). In the model the area is restricted by a circle on the earth's surface. The offset angle 2ϕ includes the edge curve of the coverage zone. This angle is the spatial angle that delineates the coverage area of a satellite from a central point on the ground. The following relationship holds for ϕ:

$$\phi = \arccos\left(\frac{r_e}{r_e + h}\cos\ \epsilon_{min}\right) - \epsilon_{min} \tag{14.10}$$

r_e is the earth's radius, h is the orbital height of the satellite and ϵ_{min} is the minimum elevation angle of the satellite system.

14.4.2.2 The Cell Structure of an Individual Satellite

In the model the cells are arranged symmetrically at the *Subsatellite Reference Point*. For a satellite system similar to IRIDIUM, 48 cells are arranged at the *Satellite Base Point* per satellite on the earth's surface (see Figure 14.28). For a system similar to Globalstar, 16 cells appear in two rings. The size of a cell is determined by the antenna chacteristics. The cell edge forms a closed curve on which the signalling of the user to his satellite is reduced by half (−3 dB) compared with the position in the cell centre.

14.4.2.3 The Cell Structure in a Satellite System

The overlapping of all the coverage areas of satellites is usually more distinct than the overlapping of the cells of an individual satellite. The overlapped areas of the coverage areas are particularly large at the earth's poles because of the predefined orbits. If the coverage zones of two satellites overlap near the equator then up to five satellites could be overlapping near the poles. In satellite systems this produces very complex cell structures, which are constantly changing because of the changing overlapping of the satellite coverage areas. The cell structures are formed by the antenna characteristics and free-space attenuation.

14.4.3 Radio Propagation

Because of large distances between a satellite and a transmit/receive station, a large free-space attenuation L_0 undergo signals given by

$$L_0[\text{dB}] = 92.4 + 20\log f_c[\text{GHz}] + 20\log d[\text{km}] \tag{14.11}$$

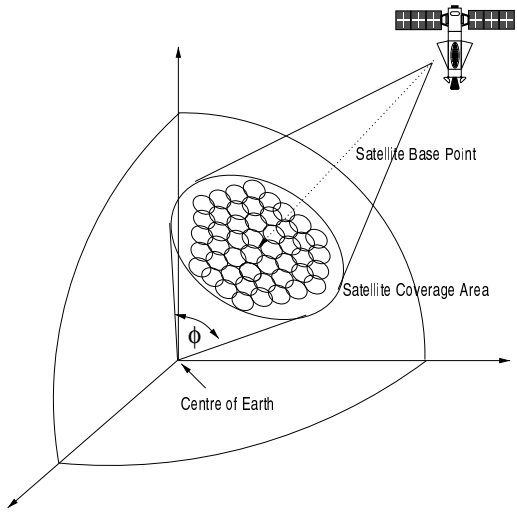


Figure 14.28: Position of cells in satellite coverage area

The *free-space attenuation* increases logarithmically with the path length, which corresponds to 6 dB with each doubling of the distance. Atmospheric and rain attenuation further increase the signal degradation. Though C-band losses due to atmospheric and rain attenuation are relatively small, this effect has to be taken into account at higher frequencies (over 10 GHz) (see Figure 2.4). The signal is also affected by the following factors:

1. Change in polarization and resulting additional interference
2. Increase in noise temperature

14.4.3.1 Path Loss in LEO and MEO Satellite Systems

As mentioned above, the free-space attenuation depends on the frequency and the distance of the mobile user or base stations from the ground to the satellite. The existence of different constellations of the systems produces different free-space attenuations. There is a big difference between maximum and minimum distance owing to the low flight paths and relatively large illumination zones, especially with the LEO and MEO constellations, illustrated in Figure 14.29 for a LEO constellation.

According to Equation (2.5), the free-space attenuation is dependent on the frequency used and the distance between satellite and mobile station. The different distances that can exist between users and their connecting satellites produce varying levels of free-space attenuation. The distance d between a user and a satellite depends on the *elevation angle*:

$$d = \sqrt{r_e^2 + h^2 - (r_e \cos\epsilon)^2} - r_e \sin\epsilon \tag{14.12}$$

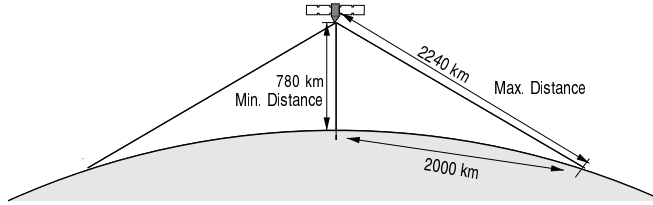


Figure 14.29: Distances in a LEO system constellation

Table 14.8: Comparison of mininimum and maximum free-space attenuation [dB]

IRIDIUM	Mobile station (uplink)	−156.36	−163.28	6.92
	Mobile station (downlink)	−157.13	−164.06	6.93
	Ground station (uplink)	−167.28	−174.21	6.93
	Ground station (downlink)	−164.69	−171.62	6.93
Globalstar	Mobile station (uplink)	−161.51	−167.44	5.93
	Mobile station (downlink)	−162.30	−168.23	5.93
	Ground station (uplink)	−172.45	−178.38	5.93
	Ground station (downlink)	−169.85	−175.78	5.93
Odyssey	Mobile station (uplink)	−178.81	−181.08	2.27
	Mobile station (downlink)	−179.59	−181.86	2.27
	Groud station (uplink)	−189.74	−192.01	2.27
	Ground station (downlink)	−187.15	−189.42	2.27

r_e is the earth's radius, h is the height of the satellite and ϵ is the elevation angle.

Examples of minimum and maximum free-space attenuation resulting within a satellite illumination zone are given in Table 14.8 for three satellite systems with different orbit heights: IRIDIUM (780 km), Globalstar (1389 km) and Odyssey (10 354 km).

The difference between maximum and minimum free-space attenuation (last column in Table 14.8) in LEO system constellations is considerably greater than in the MEO constellation Odyssey, because the difference in distance between the satellite base point and the edge of the illumination zone is greater.

14.4.3.2 Attenuation Caused by Weather Conditions (3 . . . 180 GHz)

With frequencies over 1 GHz, scattering and absorption of electromagnetic energy due to rain cause considerable attenuation to signal amplitude. These effects increase at higher frequency.

Table 14.9: Rain rate statistics [mm/h] of different regions of Europe [NASA Reference Publication 1082(03)]

Percent annually	Zone C	Zone D_1	Zone B_2
0.001	78	90	70
0.01	28	35.5	23.5
0.1	7.2	9.8	6.1

Statistics of Rain Rate The problems increase with the statistical characteristics of the rain rate. The maximum rain rates often exceed average values. Table 14.9 lists the rain rates that are reached 0.001 %, 0.01 % and 0.1 % of the time in zones C, D_1 and B_2. These zones cover most of the area of Europe. Germany, France and the British Isles belong to Zone C, large parts of Southern Europe belong to Zone D_1 and Eastern Europe to B_2 [15].

Calculation of Rain Attenuation Different methods are available for predicting rain attenuation. According to the CCIR recommendation [15], the attenuation A is given by

$$A = aR^b L_{eff}[\text{dB}] \tag{14.13}$$

The value aR^b is the specific attenuation [dB/km] and L_{eff} the effective path length due to rain. The coefficients a and b are based on frequency, raindrop size, elevation and other factors, but can be approximated as follows [14]:

$$a = \begin{cases} 4.21 \cdot 10^{-5} f^{2.42}, & 2.9 < f < 54 \text{ GHz} \\ 4.09 \cdot 10^{-2} f^{0.699}, & 54 < f < 180 \text{ GHz} \end{cases} \tag{14.14}$$

$$b = \begin{cases} 1.41 \cdot f^{-0.0779}, & 8.5 < f < 25 \text{ GHz} \\ 2.63 \cdot f^{-0.272}, & 25 < f < 164 \text{ GHz} \end{cases} \tag{14.15}$$

The rain height h_r, which depends on the degree of latitude ϕ, is required in order to calculate the effective path length based on the CCIR recommendation:

$$h_r = 5.1 \text{ km} - 2.15 \log(1 + 10^{(\phi - 27)/25}) \text{ km} \tag{14.16}$$

The path length L can then be calculated from h_r and the height of the receiver over NN, h_0, as well as from the elevation angle θ as follows [15]:

$$L = \begin{cases} \dfrac{2(h_r - h_0)}{(\sin^2(\theta) + 2(h_r - h_0)/8500)^{\frac{1}{2}} + \sin(\theta)} \text{ km}, & \theta < 10^\circ \\ \dfrac{h_r - h_0}{\sin(\theta)} \text{ km}, & \theta \geq 10^\circ \end{cases} \tag{14.17}$$

The attenuation is then:

$$A = aR^b L r_p, \qquad \text{with} \quad r_p = \frac{90}{90 + 4L\cos\theta} \tag{14.18}$$

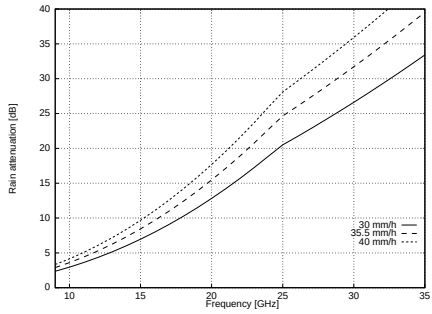


Figure 14.30: Rain attenuation versus frequency on the basis of different rain rates and 10° elevation

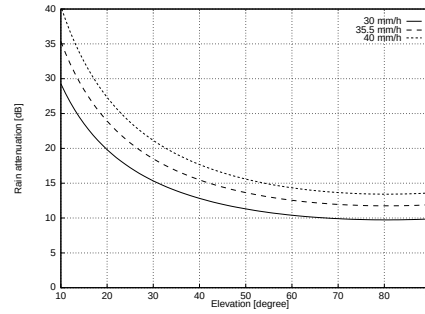


Figure 14.31: Rain attenuation versus elevation at 20 GHz

The basis used in system design is normally the value $R_{0.01}$ [15], which indicates the rain rate that 99.99 % of the time is actually lower, and the required transmission quality guaranteed for this rain rate. As can be extrapolated from the equations, the rain attenuation depends on many different factors, such as rain rate, frequency, elevation, geographical latitude, height of earth station, as well as other values such as raindrop diameter.

At frequencies above 10 GHz rain attenuation has a strong effect on connection quality. The dependency of the frequency was determined through a simulation of the geographical latitude of 40° and an elevation of 10° and illustrated in Figure 14.30. The rain attenuation increases dramatically the higher the frequency. Frequency selection therefore has a great impact on transmission quality, thereby limiting suitable frequency bands.

Figure 14.31 shows the simulated rain attenuation over the elevation at 20 GHz. Below a 30° elevation, the rain attenuation rises sharply as the elevation decreases. The aim therefore is to strive for constellations with high elevations. This can only be achieved through increasing the number of satellites, which results in high costs and high risks or through high orbits which in turn increases the signal propagation times.

14.4.4 Power Control

The blocking probability of calls depends in part on signal strength, so power control can be used to compensate for path loss. Otherwise mobile stations located a minimal distance from the satellite would have a high signal strength at the antennas of the satellite and consequently cause comparatively strong co-channel interference.

In general, power control serves the following purpose in satellite systems:

- Compensation of propagation attenuation.

- Compensation of deterioration in radio propagation conditions (e.g., through meteorological influences) in order to maintain a good-quality link.
- Compensation of co-channel interference. This occurs when more than one mobile station is maintaining a connection on the same channel at the same time.
- Protection of mobile station and satellite batteries through reduction of emitted signal power.

A possible solution to this problem would be for mobile stations operating in the proximity of the satellite base point to reduce their power by the following in favour of mobile stations on the edge of the satellite radiation zone:

$$L = 20 \log \left(\frac{4\pi f d_{max}}{c} \right) \tag{14.19}$$

d_{max} is the maximum distance between a satellite and a mobile station on the edge of the illumination zone, c is the speed of light and f is the frequency.

14.5 Interference in the Satellite Radio Network

Besides attenuation, signals experience disturbance due to interference from other signals. There are two different types of interference:

1. Co-channel interference caused by the use of the same radio channel in a neighbouring cluster.
2. Adjacent-channel interference caused by the use of adjacent radio channels in the same cluster.

14.5.1 Co-Channel Interference

Channel allocation methods are based on an evaluation of the carrier-to-interference (C/I) ratio measured at the antennas of the satellite and of the mobile user. A simple model is used here to explain the level of interference. No influences caused by weather (attenuation, polarization) or environmental influences (shadowing, fading) are taken into account. The received and transmitted signal strength is determined solely on the basis of free-space attenuation, the characteristics of the antenna used and the power control of the mobile stations. Interference in the simplified model is therefore caused by the user signal of a call and the noise signals received from calls of other mobile stations. Interference can only occur if both connections are using the same physical channel.

Figure 14.32 shows that interference can occur in the uplink and in the downlink for various reasons, thereby necessitating a separate calculation of C/I_{Uplink} and the C/I_{Downlink}. The carrier-to-interference ratios will be calculated below.

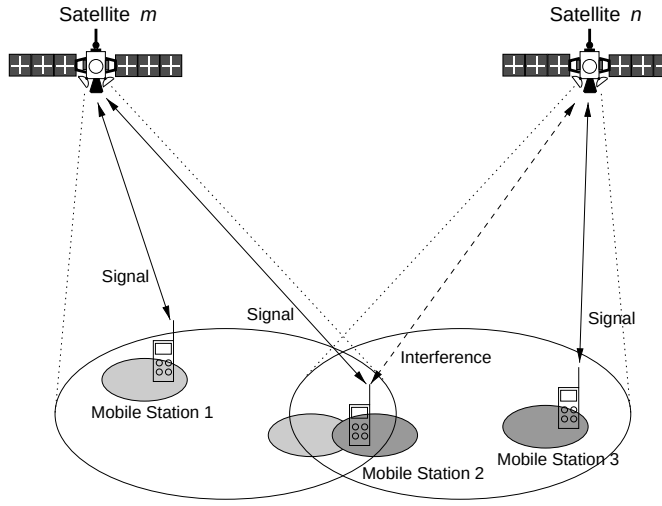


Figure 14.32: Co-channel interference caused by the use of the same channel

14.5.2 Uplink Carrier-to-Interference Ratio

The carrier-to-interference ratio C/I_{Uplink} measured by the satellite is

$$C/I_{\text{Uplink}} = \frac{C_{\text{Uplink}}}{\sum_{x=1}^{N} I_x(\text{mobile station}_x)} \tag{14.20}$$

C_{Uplink} is for the user signal power of a mobile station received on the channel of a satellite. I is the interference power received on the same channel from mobile stations located in the coverage area of the satellite and maintaining a connection to this or to a neighbouring satellite. N is the number of co-channel mobile stations that are operating in the illumination zone of the satellite.

For example, in Figure 14.33 mobile station 3 has set up a connection to satellite n (antenna 1); mobile station 2, on the other hand, has a connection to satellite m (antenna 5). Since both mobile stations are transmitting on the same channel, mobile station 2 is interfering with the signal sent by mobile station 3. C/I_{Uplink} of the connection to mobile station 3 received at satellite n (antenna 1) is therefore given by

$$C/I_{\text{Uplink}}(\text{Sat } n,\ \text{Ant 1, MS 3}) = \frac{C(\text{Sat } n,\ \text{Ant 1, MS 3})}{I(\text{Sat } n,\ \text{Ant 1, MS 2})} \tag{14.21}$$

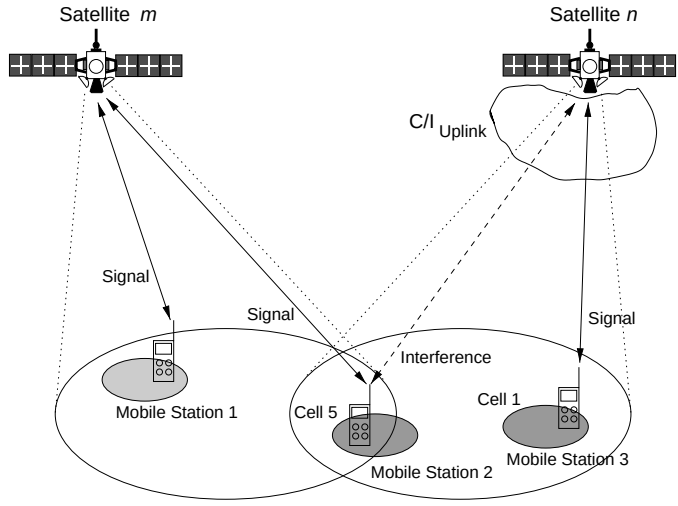


Figure 14.33: Development of uplink interference level

14.5.3 Downlink Carrier-to-Interference Ratio

The carrier-to-interference ratio C/I_{Downlink} at a mobile station is

$$C/I_{\text{Downlink}} = \frac{C_{\text{Downlink}}}{\sum_{x=1}^{N} \sum_{y=1}^{M_x} I_{xy}(\text{satellite antenna}_{xy})} \tag{14.22}$$

C_{Downlink} is the user signal level received on a channel at the mobile station. N is the number of satellites in whose coverage area a mobile station requesting a channel is located. M_x is the number of antennas for each of these satellites that is maintaining a connection to another mobile station on the same channel. I_{xy} is the interference level sent by the above-mentioned satellite antennas and received at the mobile station.

Figure 14.34 illustrates this example in more detail. Satellite n (antenna 1) has established a connection to mobile station 3. Satellite m (antenna 5), on the other hand, has a connection to mobile station 2. Since both satellites are transmitting on the same channel, satellite n (antenna 1) is interfering with the user signal of the satellite (antenna 5) at mobile station 2. C/I_{Downlink} of the connection to satellite m (antenna 5) received at this mobile station is then given by

$$C/I_{\text{Downlink}}(\text{Sat } m,\ \text{Ant 5, MS 2}) = \frac{C(\text{Sat } m,\ \text{Ant 5, MS 2})}{I(\text{Sat } n,\ \text{Ant 1, MS 2})} \tag{14.23}$$

14.5.4 Model of a Land Mobile Satellite Channel

A channel model is introduced to determine the bit error probability and incorporates a channel structure similar to the IRIDIUM system. Consider-

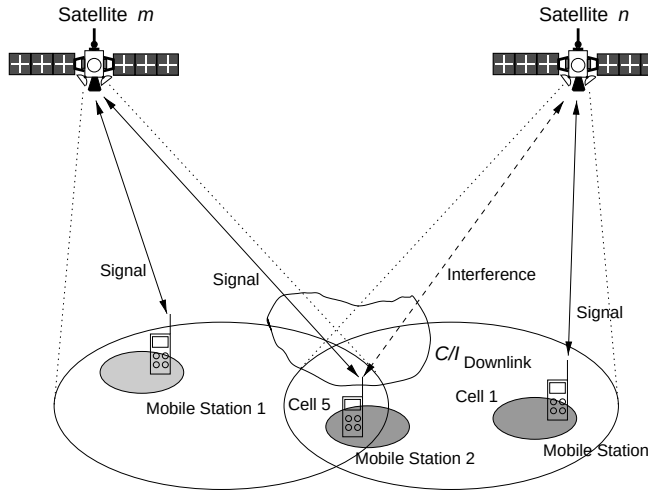


Figure 14.34: Development of downlink interference level

ation will only be given to the communications link between mobile station and satellite. Because of the higher transmit power of the ground station with a higher antenna gain (see [5]) and constant line-of-sight path, the link between satellite and ground station (*gateway*) is not considered important for the discussion here.

As is the case with terrestrial mobile radio channels, land mobile satellite channels are also affected by the following interference effects [1]:

Fading due to shadowing Obstacles such as trees and buildings on the transmission path cause short-term shadowing and consequently produce considerable signal attenuation. The shadowed areas become larger as the satellite elevation drops, which results in a minimum elevation angle necessary for transmission.

Multipath fading Objects in the immediate vicinity of a mobile radio user can cause reflection of the signal with different propagation times of the resulting paths at the receiver. The superposition (interference) of the reflected signal with the signal transmitted over the line-of-sight produces short-term signal fading.

In contrast to terrestrial channels, where a mobile station's speed of movement has a major effect on the duration of fading, with satellites this hardly makes any difference. For example, a satellite in the IRIDIUM system has an orbital speed (orbit time: 1 h 40‘ 27“) of around 27 800 km/h. Taking into account the earth's rotation (1670 km/h at the equator) through addition of the two velocity vectors produces the relative speed to earth, which is higher than the mobile station speed by many magnitudes. The IRIDIUM satellite

is approximately 150 times faster than an automobile travelling at 170 km/h. Consequently the fading duration is shorter by a factor of 50 than in terrestrial mobile radio networks.

Between 1984 and 1987, the German Aerospace Institute (DLR: Deutsche Forschungsanstalt für Luft- und Raumfahrt, Oberpfaffenhofen, Germany), studied channel behaviour for different elevation angles and in different environments over the geostationary MARCES satellite. A modulated carrier signal in the L-band (1.54 GHz) was beamed over satellite from a ground station in Spain (ESA in Villafrance) to a measuring vehicle in different locations in Europe (Stockholm, Copenhagen, Hamburg, Munich, Barcelona and Cadiz) [11].

The results show short-term fading signals in open areas, e.g., as caused by trees and bridges to a travelling automobile. Even stronger multipath and fading signals caused by shadowing were observed in urban areas. Shadowing is particularly more pronounced in cities, and leads to time durations that alternate between good and poor reception conditions.

The measurements taken were used to develop a model for a land mobile satellite channel that reproduces the stochastic behaviour of the signal carrier received. The basis of this model is formed by two channel states that mirror each other in good and poor reception conditions. A good channel state is represented by a direct line-of-sight connection to the satellite, whereas a poor state is characterized by shadowing of the direct transmission path. In both cases the satellite signal is reflected by a large number of objects located in the immediate vicinity of the mobile station. The respective signals are received by the mobile station with differing propagation times and amplitudes, and superpose themselves with those of the direct transmission path.

This so-called DLR model of the land mobile satellite channel appears in Figure 14.35. A distinction is made between the good channel state with Rice fading and the bad state with lognormally distributed Rayleigh fading and low receive power. The switch controlled through the Markov process reproduces a random succession of good and bad channel states. The transmit signal is multiplied with the fading signal $a(t)$ and an *AWGN signal* (white Gaussian noise) is then added.

The channel model is characterized by the Rice factor c, the mean value μ and the dispersion σ^2 of the lognormal attenuation, as well as by the average durations D_g and D_b of the good or bad channel states. The time part of bad channel states is indicated as shadowing factor A [10]. These parameters are given in [11].

The average duration of channel states depends on the following factors:

- Environment (*open, suburban, urban*)
- Elevation of visible satellite
- Speed at which mobile station is moving

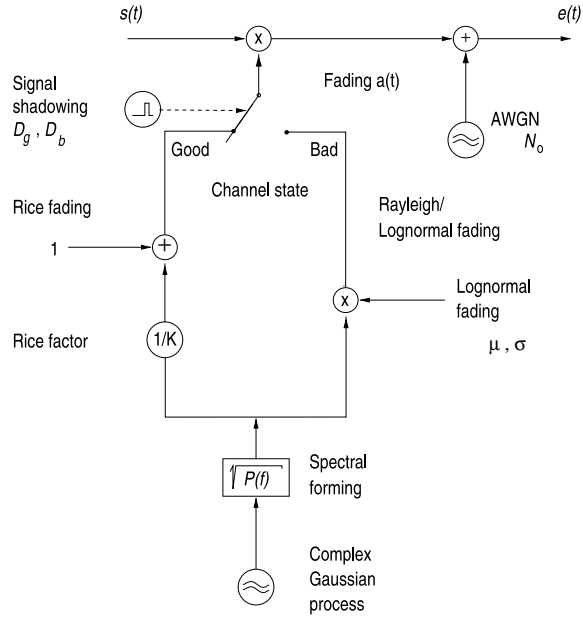


Figure 14.35: Analogue (DLR) model of land mobile satellite channel

These factors can be used to simulate the signal profile of a land mobile satellite channel for different receiving conditions. Figure 14.36 provides an example of simulation results for the *receive signal envelope*. The information digitally modulated to the carrier signal is recovered in the demodulator of the receiver, and the current channel quality is assessed through an evaluation of the error-correction unit in the receiver. The connection is jeopardized if the channel quality is not adequate, and must then be replaced by a connection with a better channel.

14.6 Handover in Mobile Radio Satellite Systems

A handover is a changeover to a different physical channel without the risk of the current connection being broken off. Handovers help to provide users with good-quality connections, and consequently are an important feature of the mobile radio network.

In contrast to the stationary cell structures of terrestrial cellular networks, where handovers are necessary because of user mobility, the cells in LEO satellite systems move in relation to the earth's surface at up to 7 km/s (see Figure 14.37).

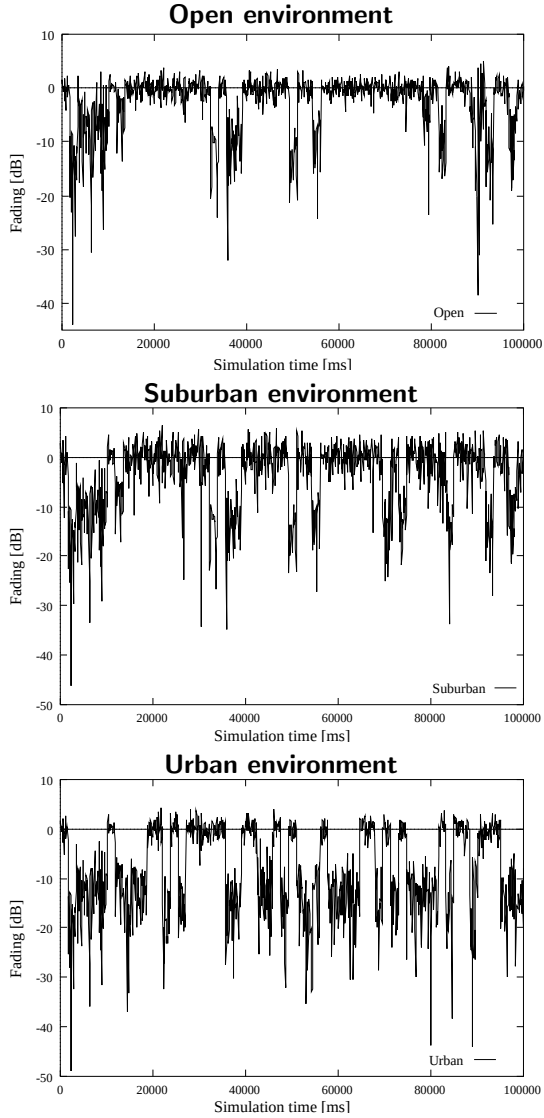


Figure 14.36: Signal profile in different environments, velocity $v = 108$ km/h, 21° satellite elevation

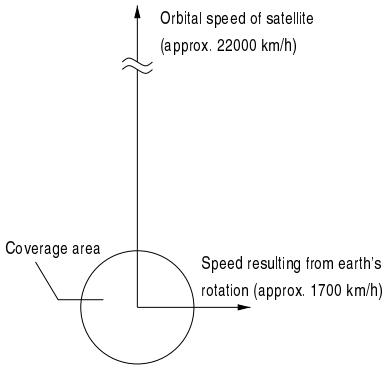


Figure 14.37: Movement of a coverage zone (spot)

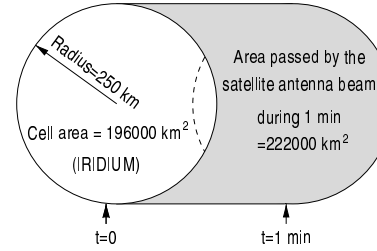


Figure 14.38: The area of an IRIDIUM system cell overflown in one minute

14.6.1 Frequency of Handovers

Compared with terrestrial mobile radio systems, the cells of satellite systems are very large; for example, with IRIDIUM the diameter of the cells is approximately 500 km. If these cells were stationary on the ground, there would seldom be a need for a handover during a call. If it could be assumed that communicating mobile radio users are distributed evenly in a certain area of the world, it would be possible to calculate the new area travelled during a given time interval and consequently the average number of handovers executed.

In one minute an IRIDIUM satellite cell travels over an area that is 1.12 times larger than the area of the cell (see Figure 14.38). Based on the assumption that a cell has a traffic volume of 80 Erl., the number of handover occurrences would be on average $80 \cdot 1.12 = 90$ per minute.

Based on an average call duration of 2 min and 80 Erl. of traffic, the number of new calls would be 40 per minute. The number of handovers would then be greater by a factor of 2.25 than with terrestrial systems.

With satellite systems communication is only possible over a line-of-sight link. Therefore the time span of the line-of-sight link is extremely important. It depends on the orbital speed of the satellite and the size of the coverage zone (spot), as well as the position and direction of movement relative to earth. Studies have shown that the maximum duration of a visit in a coverage zone can be calculated as

$$t_{max}(r) = \frac{2\, r_{cov}}{v_g} \tag{14.24}$$

The assumption is that the mobile user is not moving, because his speed compared to the orbital speed of the satellite and the speed based on the earth's rotation is negligible. In Equation (14.24) r_{cov} is the radius of the coverage zone and v_g the velocity of the satellite base point relative to the

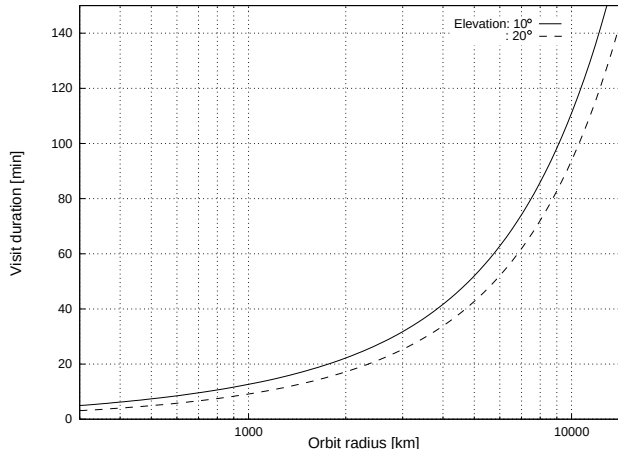


Figure 14.39: Maximum visit duration in a coverage zone

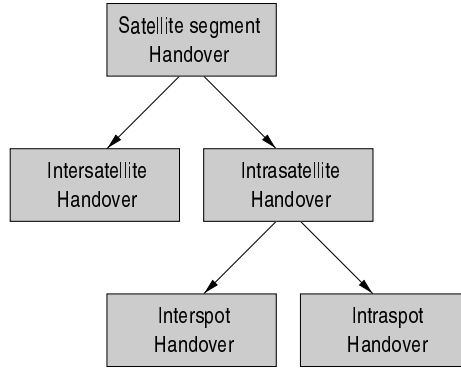


Figure 14.40: Types of handover in a satellite segment

earth's surface. For two elevation angles Figure 14.39 shows the maximum visit duration of a mobile user in a coverage zone over the orbit radius.

14.6.2 Types of Handover

A differentiation is made between intrasatellite handovers within the coverage zone of a satellite and intersatellite handovers between different satellites. Intrasatellite handovers are divided into intracell and intercell handovers (see Figure 14.40).

14.6.2.1 Intersatellite Handovers

Even stationary terminals are covered by the same satellite only for just a few minutes (see Figure 14.39).

When a connection is being established or a handover is being executed, the satellite that remains visible for as long a period of time as possible must be selected in order to keep the number of handovers required to a minimum. During this period of time, the channel parameters inevitably change considerably because unfavourable but still acceptable elevations occur. If a satellite is selected because of optimal channel parameters, it is more than likely that more handovers will have to be executed.

The objective is to keep the elevation angle at an acceptable level in order to guarantee satisfactory connection quality. There are two procedures known for this [3].

Maximization of current elevation angle With this strategy the satellite that is visible below the currently largest elevation angle is selected at all times during an existing connection. Handovers are also carried out if the connection quality is sufficient in order to maximize them. The shadowing probability is reduced because the connection is constantly being executed above the optimal elevation angle. The disadvantage of this procedure is that handovers could be executed unnecessarily because a satellite can provide sufficient connection quality even at a lower elevation.

Minimization of intersatellite handover rate This strategy allows handovers only when a given minimum elevation angle ϵ_{min} has been reached. For a new connection the satellite that is visible the longest is always selected on the condition $\epsilon \geq \epsilon_{min}$. This necessitates a position establishment by the mobile station.

With this procedure it is possible to keep the handover rate low, but in some cases the satellite selected cannot supply an acceptable connection quality.

Possible criteria for an intersatellite handover are as follows:

- The signal quality falls below a certain margin, i.e., the distance between satellite and user has become too great.
- The signal carrier-to-interference ratio is too low because other users are using the same channel and causing co-channel interference.

If a handover is initiated, the satellite that has the best signal quality or the best elevation angle is selected.

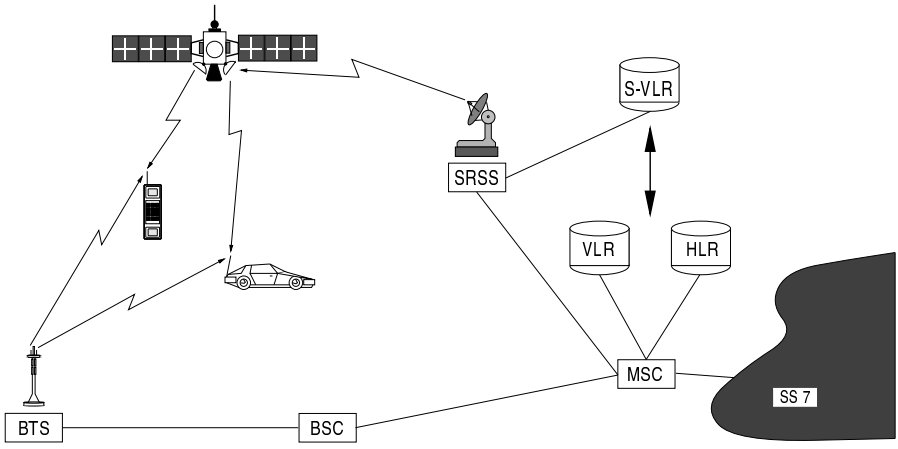


Figure 14.41: System level integration scenario in GSM systems

14.6.2.2 Intersegment Handover

The concept of an intersegment handover is used to describe an important function for integrating future LEO satellite systems into terrestrial cellular networks. An intersegment handover enables an active connection to be changed from a terrestrial cell to a satellite cell (*generated by a spotbeam antenna*) or the other way around.

A typical application would be the support of local cellular networks (e.g., of a city) by a satellite network in areas away from cities.

Intersegment handovers are discussed below based on the GSM system. A possibility for the integration of ground stations (*gateways*) into the GSM network is also introduced.

Integration at the network level This integration concept deals with the direct connection of satellite gateways to mobile switching centres (MSCs) of the GSM network (see Chapter 3). As illustrated in Figure 14.41, the gateway together with the *Controller* forms a *Satellite Relay Subsystem* (SRSS) which has a similar function as the *Base Station Subsystem* (BSS) of GSM. In addition, the SRSS has its own database, the *Satellite Visitor Location Register* (S-VLR), into which information on all active users located within the coverage zone (*footprint*) of the satellite is entered. Just like the VLR in GSM, this database can be used through the MSC or through the *Home Location Register* (HLR). The advantage of this kind of integration of satellite system and GSM is that it allows the use of existing protocols and interfaces for mobility management. It requires a close cooperation between satellite and GSM network operators [7].

Intersegment handovers in integrated GSM satellite systems Intersegment handovers, which correspond to intra-MSC handovers in GSM, are introduced below.

Figure 14.42 presents the handover protocol for changing to the GSM segment and Figure 14.43 that for the opposite direction. In both cases the relationship to the intra-MSC handover protocol described in Section 3.6.8 is apparent. Even the protocol elements are essentially identical. Changes to cell and channel identity appear to be necessary in the satellite segment to comply with the GSM handover protocol. The next paragraph deals with a peculiarity of satelllite channel access.

Random access to satellites Unlike terrestrial networks, in which mobile stations, except for the time-displacement factor (*timing advance*), synchronize with the logical time-multiplex channel for multiple access before they access the channel, synchronization in the satellite segment is not always possible during a handover. Because the signal propagation times are not known and depend on the current distance between satellite and user or gateway, it is not possible for a mobile station to be notified before it accesses the satellite channel of the signal propagation time correction that it must undertake in order to maintain the frame structure with the satellite.

Separate channels therefore exist in the satellite segment for the transmission of channel allocation requests (Handover_Access, Channel_Request) that require no synchronization with the satellite or with other users. Each mobile station is allowed to transmit without considering whether the channel is already occupied. The pure ALOHA procedure is an example of this kind of *Time-Random Multiple-Access* protocol (TRMA protocol).

If several mobile users are transmitting at the same time on the Pure ALOHA channel, collisions will occur and collided data packets have to be retransmitted. The packets are not transmitted again until a random waiting time has passed to ensure that another collision may not occur when they are resent.

If the duration of a data packet is T, as illustrated in Figure 14.44, and it is assumed that all packets are of the same length, then no other stations are allowed to send in the time frame between $-T$ and $+T$ ($t = 0$ at end of transmission) so that the communication can be received correctly. The phase of conflict, i.e., the time during which no other station is allowed to send, is then $2T$. It is double the length of a packet. Equivalently to the analysis of the Slotted ALOHA protocol in Section 2.8.1, independent access by mobile stations can be modelled by a Poisson arrival process for packets. The probability of a successful transmission during the time T, which then provides the throughput (packets/T) according to Equation (2.49) can be calculated to be

$$S = Ge^{-2G} \tag{14.25}$$

where G is the offered traffic (packets per timeslot).

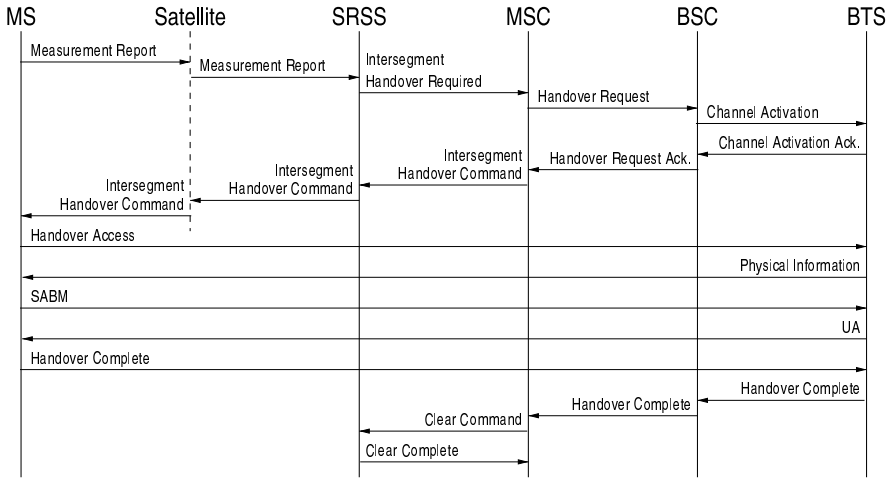


Figure 14.42: Intersegment handover protocol for changeover from satellite to GSM segment

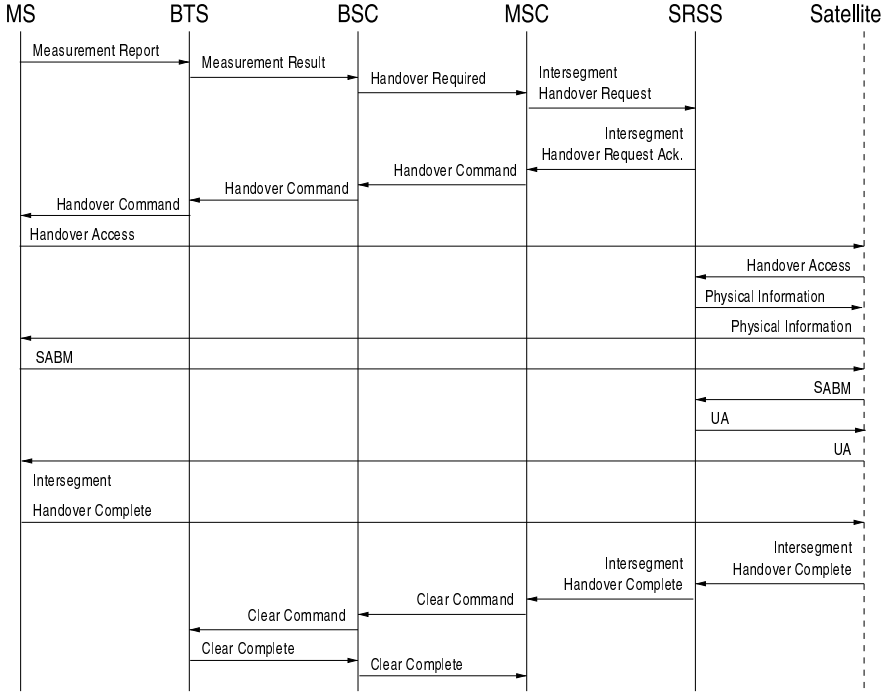


Figure 14.43: Intersegment handover protocol from GSM to satellite segment

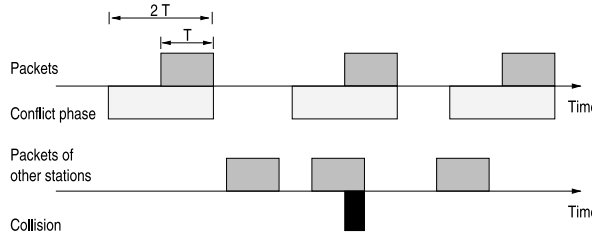


Figure 14.44: Pure ALOHA access protocol

The maximum throughput of pure ALOHA ($(2e)^{-1}$) is only half the value known from S-ALOHA ($1/e$).

14.7 Using Satellites to Link Wireless Access Networks to a Fixed Network

LEO satellite systems such as Globalstar and IRIDIUM achieve good channel quality with only line-of-sight links. Telephone calls from buildings or in shadowed areas are not possible. Wireless terrestrial access networks (WLL) (see Chapter 10) are used to link areas without coverage quickly to a telecommunications infrastructure. The issue discussed below is whether LEO satellite systems lend themselves to linking the base stations of WLL systems to the fixed network, thereby becoming competitors to point-to-multipoint radio relay systems, which are also being used for this purpose. Potential access networks that could be linked to the fixed network are shown in Figure 14.45:

- Wireless local-loop systems
- Base stations of cellular radio networks
- Private branch exchange systems

The advantages of satellite-supported point-to-multipoint radio relay systems compared with terrestrial radio relay systems include:

- Fast installation as long as suitable satellite systems are available.
- Low additional costs for infrastructure to handle new connections.

The disadvantages are:

- Higher signal propagation times.
- High system costs of satellite systems.

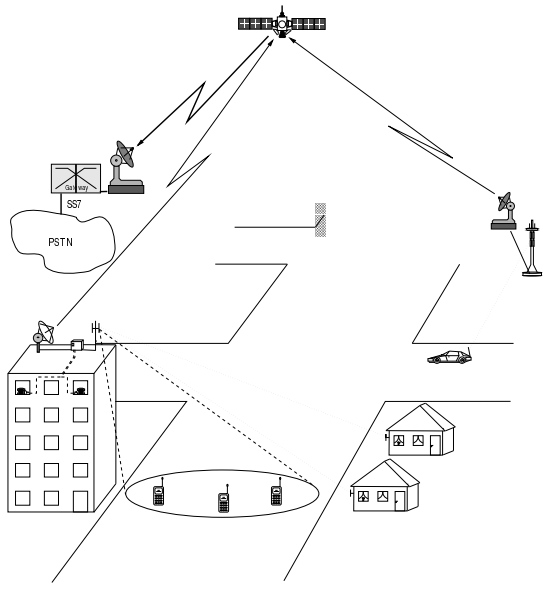


Figure 14.45: Scenarios showing the use of a satellite system as a transport network

This raises the question of how many small or large cities could be supplied coverage by a given LEO system. The example considered involves a penetration of 1 % as well as of 10 %, with the WLL systems of a city being modelled as a point source/sink. With satellite cell sizes of 600 km this approximation is sufficiently exact. However, the area over which the WLL systems extend should not be neglected in the capacity study of the satellite system.

14.7.1 Simple Fictional WLL System

Since the first LEO satellite systems only support low transmission rates, WLL systems, e.g., according to the DECT standard, can only be linked through the trunking of several satellite channels per, say, DECT channel. However, this would considerably reduce the number of calls that could be carried. Therefore a fictional WLL system with a transmission rate of 4.8 kbit/s adapted to the satellite channel is assumed here for the capacity study. The voice quality will then be low, but it can be assumed that there will be a large market for services at a lower quality than is normally the case with telecommunications networks in places like Central Europe, as long as the price is sufficiently low. The Internet is an example of a data transfer system with currently a poor quality (in terms of throughput) but a high acceptance.

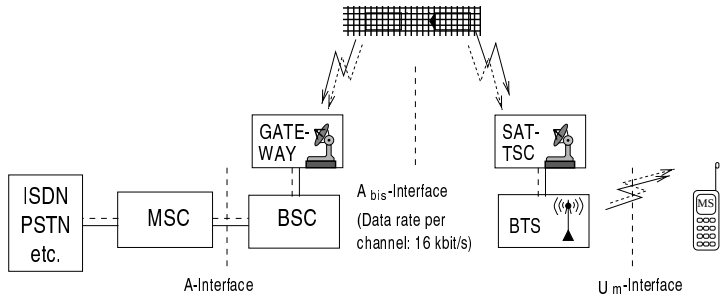


Figure 14.46: Connection of base stations over satellites

14.7.1.1 Linking Base Stations to Cellular Mobile Radio Networks

The location of base stations is selected from the standpoint of radio supply conditions and cost. Since the ideal positions in terms of radio supply frequently result in high wired connection costs, radio links are often used to link the base stations. LEO satellites could offer an alternative connection option.

Figure 14.46 illustrates the architecture of base stations linked over satellite. A new element that comes into play is the satellite-terminal controller (SAT-TSC) which is used to link base stations to the satellite system exploiting the higher gain of a directed antenna.

This kind of configuration places high demands on the satellite system in terms of delay time and bit-error ratio. There must be an assurance that no timers are running out in protocols and that messages are not retransmitted too often so that possible delay times do not become too large.

14.7.1.2 Potential Systems

Compared with the traffic to be carried, the capacity of the IRIDIUM system is too low even on the assumption that voice transmission will be at only 4.8 kbit/s (see scenario 1a in Figure 14.47). However, WLL connections could be implemented in the L-band for example (see scenario 1b in Figure 14.47) for linking GSM base stations.

The TELEDESIC system would be more appropriate as a transport network for WLL systems. Figure 14.48 presents a scenario with a DECT-WLL system. Since TELEDESIC has considerably more channels available than IRIDIUM, and DECT can be seen as representative of smaller WLL systems, this could be an interesting combination. Furthermore, the suitability for linking GSM base stations is being discussed. Figure 14.48 also shows the connection of larger users, e.g., GSM base stations in the Ka-band.

Rough calculations show that the IRIDIUM system would only be considered in special cases, and the TELEDESIC system only with major limitations, as the technology connecting WLL base stations to the fixed network if

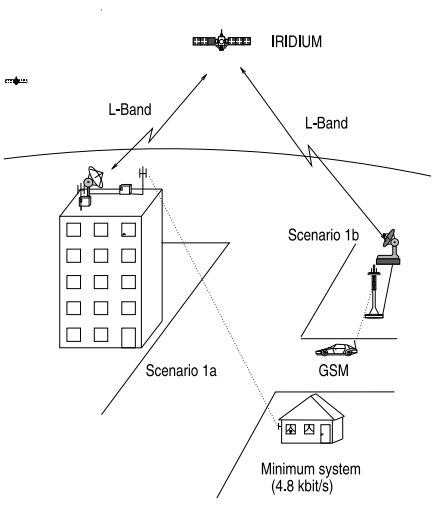


Figure 14.47: IRIDIUM for connection of a WLL system

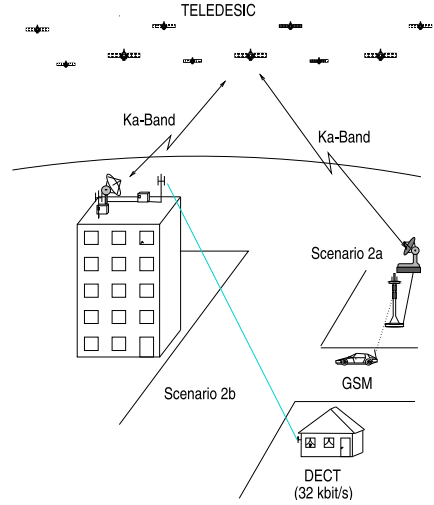


Figure 14.48: TELEDESIC for connection of DECT-WLL and GSM systems

a considerable proportion (10 %) of the traffic volume is to be carried, e.g., in Europe or the USA. The satellite cells are still too large for this application. The traffic that occurs in this area exceeds the capacity of satellite systems many times over. So the choice is to use this option only in isolated cases or to wait for satellite systems with definitely smaller cell radii and/or clearly higher capacity.

The situation is totally different when it comes to linking large rural areas or third world countries with very low traffic volumes per square unit to telecommunications facilities. Even the IRIDIUM system with its small traffic capacity could be an interesting option economically. The TELEDESIC system could be used there as a transport network for WLL systems for large regions without any restrictions [6].

14.8 Mobile Satellite Communication Abbreviations and Acronyms

ATDMA	Adaptive Time-Division Multiple Access
BPSK	Binary Shift Keying
CDMA	Code-Division Multiple Access
dB	Decibel
DL	Downlink

DLR	German Aerospace Research Institute
EUTELSAT	European Telecommunications Satellite Organization
FCC	Federal Communications Commission (USA)
FDD	Frequency-Division Duplexing
FDMA	Frequency-Division Multiple Access
FEC	Forward Error Correction
GEO	Geostationary Orbit
GLONAS	Russian Global Position System
GPS	Global Positioning System
GW	Gateway to/from fixed network
HEO	Highly Elliptical Orbit
ICO	Intermediate Circular Orbit
INMARSAT	International Maritime Satellite Organization
ISL	Inter Satellite Link
ITU	International Telecommunications Union
LEO	Low Earth Orbit
MEO	Medium Earth Orbit
MS	Mobile Station
NCC	Network Control Centre
NCS	Network Control Station
PLMN	Public Land Mobile radio Network
PSDN	Public Switched Data Network
PSTN	Public Switched Telephone Network
QPSK	Quadrature Phase Shift Keying
RDSS	Radio Data Satellite Service
SAN	Service Area Node
Sat	Satellite
SMS	Short Message Service
TDD	Time-Division Duplexing
TDMA	Time-Division Multiple Access
TRMA	Time-Random Multiple Access
TT&C	Terrestrial Terminal and Control
UL	Uplink

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15 UPT—Universal Personal Telecommunication*

People are becoming more mobile. At the same time they have a greater need to be reachable and to be able to reach others. A variety of different systems that address all the different categories of mobility already exist:

- Employees can be reached through a DECT terminal anywhere in their company.
- Tradesmen can be called over a *Digital Communication System* (DCS) within a city.
- Business people can be reached over the same telephone number via GSM anywhere in Europe, wherever they are located.
- Adventurous types are linked to civilization through the INMARSAT satellite system.

For each of these communications systems, the user requires a special terminal.

In contrast to these radio-based systems, wireline telecommunications systems capable of reaching almost every household in the industrialized countries are available. Fixed networks do not offer users mobility. To be reached at another terminal, the user has to inform the caller of his new location area in the form of a telephone number. With Universal Personal Telecommunication (UPT) the telecommunications network assumes responsibility for establishing a user's location through an interrogation of databases based on a personal telephone number. This provides mobility in telecommunications networks to a larger number of users without the need for special new terminals.

15.1 Classification of Telecommunications Services

The services of a telecommunications network first allow users to exchange information over the network in order to communicate. A service is what is

*With the collaboration of Matthias Fröhlich

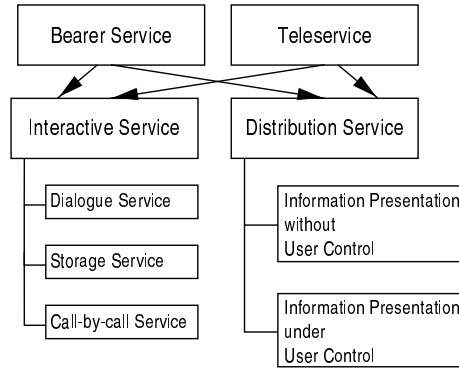


Figure 15.1: Classification of telecommunications services

made available to users by telecommunications authorities and private service providers for communication over public and private networks.

The ITU-T distinguishes [20] between two types of service:

Bearer services, also called communications services, are used to transfer data between precisely defined user–network interfaces. According to the ISO/OSI reference model, a bearer service is provided by layers 1–3 (the *Open Systems Interconnection* reference model (OSI-RM, [6, 25]) of the ISO (*International Organization for Standardization*) describes communication between open systems as seven interrelated layers). An example of this is the socalled B-channel in ISDN with 64 kbit/s of transmission capacity.

Teleservices offer users the possibility of communicating with one another over network terminals, and are provided by layers 1–7 of the OSI-RM. This also specifies the communications functions of the terminals. An example of this is the telephone service in ISDN.

With both services a separate distinction is made between interactive services and distribution services, which are subdivided further as shown in Figure 15.1.

Services are described according to the service features offered to the user. A differentiation is made between *general* connection features, *basic* service features and *extended* service features (see Figure 15.2).

Extended service features are only available in connection with a bearer or teleservice, and can be provided either outside of the network or within the network. Within the network they are subject to standardization and are specifically referred to as *supplementary services* (see Figure 15.3).

Value-added services (VAS) are extended service features that are provided by service nodes outside the network in OSI layers 4–7. These services usually contain storage, call-up or conversion functions for interfaces, protocols and

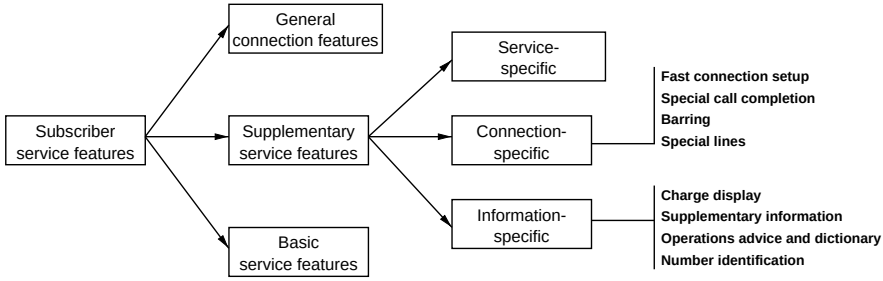


Figure 15.2: Breakdown of the service features of telecommunications services

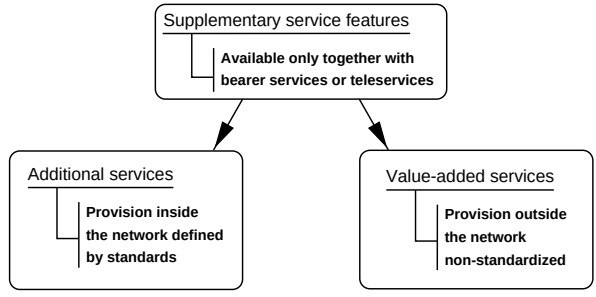


Figure 15.3: Separation of supplementary service features into additional and value-added services

bit-rate adaptation. They require an interface to a bearer or teleservice of the network.

Figure 15.4 presents a breakdown of the bearer services, teleservices, supplementary services and value-added services in a telecommunications network, with S representing the access interface of the user between terminal and network and Q the interface between value-added service and network.

15.2 Extended Service Features in ISDN and GSM

The current worldwide success of GSM is due to a great extent to the supplementary and value-added services offered. Owing to the harmonization between ISDN and GSM, most of the extended service features of ISDN are also available in GSM. The following section describes the most important supplementary and value-added services of ISDN. The service features offered additionally by GSM will be described in Section 15.2.2.

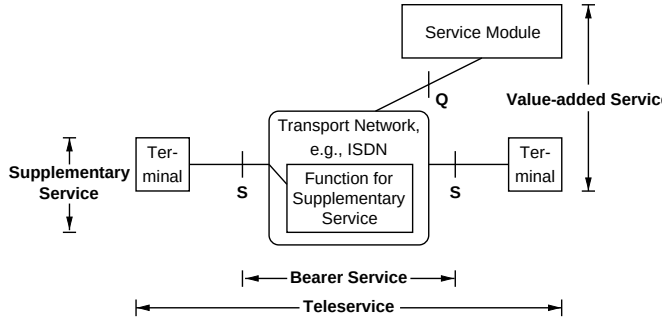


Figure 15.4: Breakdown of bearer services, teleservices, supplementary services and value-added services

15.2.1 Supplementary and Value-Added Services in ISDN

Because of full digitalization of its network, ISDN offers a variety of supplementary services compared with the analogue telephone network, the most important of which can be summarized into seven groups:

Number identification displays the telephone number of the party calling or of the party being called. The latter is of interest, for example, when call diversion is being activated. There are ways of keeping these supplementary services secret. In addition, more than one telephone number can be supported on one connection.

Call diversion allows an incoming call to be diverted immediately to another number after a brief wait or if a number is busy.

Special call completion includes supplementary service features such as call holding for enquiries as well as automatic callback or call-waiting if the number called is busy.

Group connections include conference and three-party calls as well as calls for closed user groups.

Charge functions allow to bill all or some incoming calls to the called subscribers account and to display the charges for these calls.

Supplementary information can be exchanged between the terminals using transparent user signalling.

Barring functions can be used for suppressing unwanted incoming calls, as well as for restricting specific or all outgoing calls.

For the first time, some of the supplementary and value-added services of ISDN are supporting the mobility of users, or, in other words, a personalization of the services is offered. Thus, for example, call diversion allows

incoming calls to be routed to another location, thereby supporting the user's mobility. Personalization is provided to a limited degree through the exclusive assignment of multiple-access telephone numbers to specific groups of people or the barring of certain telephone numbers for incoming calls.

15.2.2 Supplementary and Value-Added Services in GSM

Depending on the system, GSM offers several extended service features that are not supported by ISDN. In particular, these are services that support mobility and personalization directly through the system. The main ones include the following:

Device mobility, i.e., the ability to move freely within the radio coverage area of a base station during a call. Because the radio connection is switched automatically (*handover*), it is even possible to change between any number of adjacent base stations without causing an existing connection to be broken off.

Authentication of the user, i.e., establishing and checking the identity independently of the terminal used. A *Personal Identification Module* (PIM), usually in the form of a *smart card*, is used along with a *Personal Identification Number* (`PIN`).

Localization describes the automatic identification and storage of a user's location area in the network. This allows the user to be reached under his GSM telephone number irrespective of his location.

Registration means making available the services subscribed to within the framework of the contract agreed between operator and user (service profile). The costs that are incurred are charged at the same time to the user in accordance with the terms of the agreement. The subscriber's identity must be clearly established through the authentication process in order to prevent fraudulent use.

Personalization of the service profile allows the user to configure certain service features individually within the framework of the agreed service agreement. Thus, for example, the user could activate or deactivate a voice mail service (automatic answering service in the network) or restrict his own accessibility by blocking incoming calls because of the extra cost these would occur or to avoid being disturbed.

The large number of users subscribing to the GSM network underline the demand for these extended service features, so an effort is being made to support them also in the fixed network—wherever this would be technically feasible. The first step in this direction is the service referred to as *Universal Personal Telecommunication* (UPT), which is introduced below.

15.3 The UPT Service for Universal Personal Telecommunication

The UPT service has been under development since the early 1990s, and was standardized in its outline form through international recommendations [21, 24] of the ITU in 1993. The introduction of the UPT service will take place in several stages, and to date only the first (*Capability Set 1*) has been standardized [26].

In wireline telecommunications networks such as ISDN the telephone number of a user is firmly linked to the network connection of the terminal. All services of the network accessed from this network connection are charged to the user. This fixed relationship between network connection and telephone number of the user will be eliminated with the UPT service.

Identification of UPT users is carried out independently of the terminal addressing and the access points of the network. A unique UPT number allows a user to make and receive calls at any access point and from any terminal in the network.

A telecommunications network must offer different supplementary services in order to support the UPT service:

Personal mobility, which allows a UPT user to make and receive calls using different terminals in accordance with his service agreement (*UPT User's Service Profile*).

Identification of UPT users on the basis of network-independent UPT numbers.

Charging and billing, which is based on the UPT number independently of the terminal.

Uniform access and authentication functions for UPT services in different networks.

Security functions for protecting the personal data of UPT users.

Configuration functions, which are used by the UPT user and the UPT subscriber to tailor the subscribed services to meet individual requirements.

The UPT service of the first phase offers these supplementary services in a limited form, and only supports the telephone service in the analogue telephone network, in ISDN and in mobile radio networks. Other restrictions affect the scope of the security functions as well as the access and authentication functions.

These supplementary services will be available during the second phase in a less restricted form and, e.g., data services will be supported. However, this phase has not yet been standardized.

15.3.1 Existing Studies of the UPT Service

The mobile radio system UMTS (*Universal Mobile Telecommunications System*) [2, 29] is being designed to support standardized personalized worldwide user telephone numbers and supplementary services similar to those of the UPT service. Concepts for user administration in UMTS using a distributed database are being developed [28] and quantitively evaluated on the basis of models. Flat directory hierarchies used in connection with the interrogation technique *passing* are proving to be particularly favourable.

The extended service features of ISDN such as call diversion and number identification enable services similar to those of UPT to be implemented through a PC connected to the network access interface of the user. This kind of system is being developed [27] and evaluated on a simulated basis. For comparison purposes, several alternatives to UPT implementations are being simulated and studied in the Deutsche Telekom network. The implementation with a PC at the network access interface of the user is showing clear advantages in terms of waiting times for requested services. However, this does not offer a complete UPT service, because the system does not support UPT calls from non-UPT terminals.

15.3.2 Further Development of UPT

The introduction of UPT into existing telecommunications networks is a dynamic development process. The fundamental objectives mentioned in Section 15.3 are to be seen as guidelines that should be considered in any further development of UPT [30].

Figure 15.5 shows the phases projected in the development of UPT. These show a differentiation between essential and optional service features. Essential service features must be available at the outset of the corresponding development phase, whereas the offering of optional service features depends on the decisions made by the UPT provider. A certain proportion of the service features in both groups have already been standardized. The others have already been developed to a degree and are awaiting future standardization.

15.3.3 Phase 1—Scenario with Limited UPT Functionality

During the introductory phase, UPT is limiting itself to PSTN (*Public Switched Telephone Network*), ISDN (*Integrated Services Digital Network*) and possibly the *Public Mobile Telephone Service*. Only voice telephone services are being offered. UPT phase 1 does offer limited service features for security and user-friendliness. According to the projected plan, the first phase has been completed at the end of 1993. UPT phase 1 functions are already available in the IN operating tests being conducted worldwide. Phase 1 is to be carried out without any intervention in existing networks, if possible.

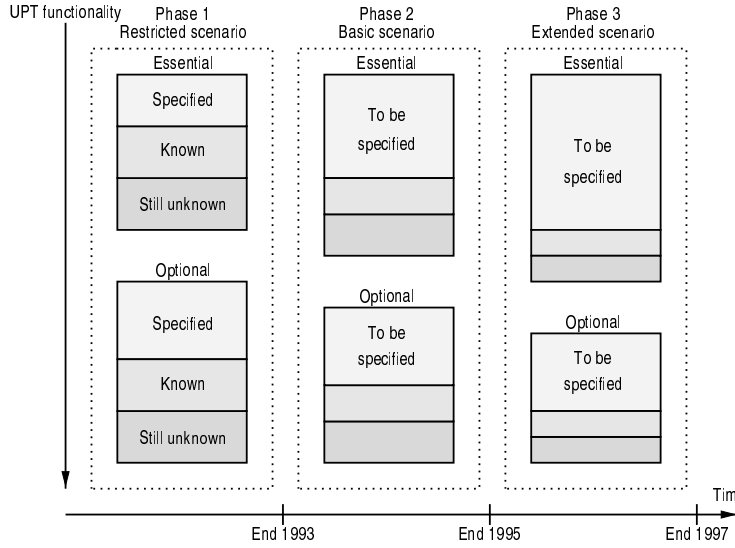


Figure 15.5: Timetable for the development and standardization of UPT service features

15.3.4 Phase 2—Scenario with UPT Basic Functionality

Phase 2 does offer UPT subscribers additional service features and supports a larger number of networks. During this phase, UPT also linked up with GSM networks and connection-oriented data networks. Smart cards and card readers offer users improved security against fraudulent use by outsiders. Phase 2, which was completed in 1995, is using the technology of intelligent networks.

15.3.5 Phase 3—Scenario with Extended UPT Functionality

Phase 3 is offering extended UPT functionality. This phase was not specified in detail, but only regarded as a target for UPT to allow a more effective integration of new technological developments and an opportunity to respond to the economic developments of UPT services. Phase 3 results were projected for late 1997, and are available since then.

15.3.6 Service Features of UPT in Phase 1 of its Introduction

As already explained, the ITU-T is considering some of the service features are being essential and others as being optional. Four service features in phase 1 are essential for the implementation of UPT [21]:

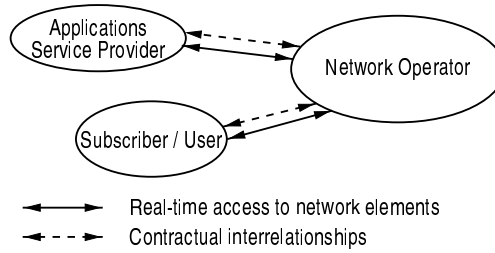


Figure 15.6: Business relationshipment between conventional telephone subscriber and system provider

- *Authentication* of a UPT user's identity protects both the UPT provider and the user from unauthorized use of the UPT service.
- *In-call registration* enables a UPT user to notify the network of the location area where he can be reached through a terminal. This registration can be restricted to a specific period of time or revoked through an explicit deregistration or new registration. It is possible for several subscribers to be registered at the same terminal.
- An *outgoing UPT call* allows a UPT user to use a terminal and have the call charged to his bill. So long as no *follow-on* service feature is available, an authentication must be carried out for each UPT call in order to prevent fraudulent use. The service feature *follow* indicates to the network that other calls are following the outcalls. New authentication is then not necessary.
- *In-call delivery* refers to the call forwarding service. The UPT subscriber makes his location known to the network beforehand through an in-call registration. The incoming call is then forwarded to this terminal. The subscriber to the network connection must be able to prevent registration and call forwarding at his terminal.

Access to UPT functions using registration is shown in Section 15.4.2.

15.4 Business Relationship between UPT Users and Providers

To be able to use the telecommunications services, users must enter into a business relationship with the provider.

Three different market participants can be identified in the conventional monopolistic telecommunications market. Figure 15.6 identifies these three business partners as:

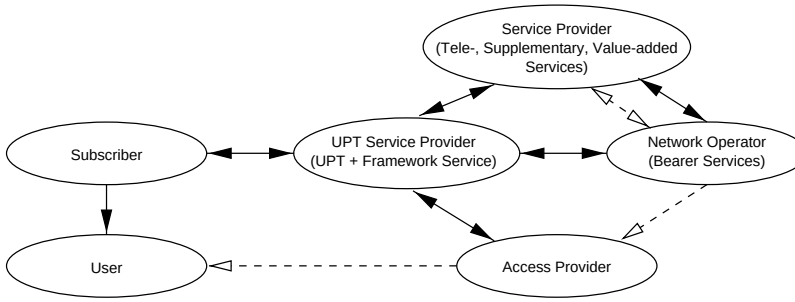


Figure 15.7: UPT business relationships

- The network operator who provides network access and the communications network
- The subscriber to the service
- The service provider of the applications who contributes to the value-added service in the network

The user has a direct business relationship with the network operator.

With UPT there is a complicated situation from the view of the user (see Figure 15.7). In addition to

- a network operator, a user and an applications service provider,

there are also

- a UPT service provider, a network access provider and a subscriber to the UPT services

who are involved in the UPT service arrangements.

A business relationship no longer exists between subscriber and network operator. Instead, a subscriber can order the UPT services from a UPT service provider. In the case of private individuals, this subscriber will be the user himself. With companies, the organization appears as the subscriber to the UPT services that it makes available to its employees through proper registration in the user service profile. Since users usually do not use their own terminals (mobility), it is important that a relationship exists with a network access provider who can provide access to the network.

A deregulation of the telecommunications market is necessary in order to enable the relationship between user and network operator to be split up. A similar situation was observed with the mobile radio networks in Europe, where users acquire mobile radio services from a service provider. The service provider is not automatically also the network operator.

15.4.1 Charging—New Concepts in the Introduction of UPT

In conventional telecommunications networks the person placing a call is charged for the duration of a successfully completed call, depending on time of day and distance between user A and user B.

With UPT, users do not have to be registered at their home locations, and instead may be linked to the network at location C. Users placing a call are not aware of the distance between home location B and visitor location C. Therefore they have no way of estimating how much a call will cost unless additional information is provided. The following scenarios are therefore possible

- User A absorbs all the costs of the call, in other words connections A–B as well as B–C.
- User A pays the cost of the connection A–B. User B pays the cost for his mobility and consequently the cost of connection B–C.
- User B absorbs all the costs.

The second solution is the one that would appear to be ideal, because the costs of the extended connection are charged to the person who is responsible for the higher cost. Other solutions are conceivable for determining the charges; however, cost transparency is important for the acceptance of new services so that users have an easy way of estimating the costs of the calls they initiate.

With UPT it may be necessary also to charge the UPT user for calls that were not switched successfully. The reason for this is the running costs and investment in processors and databases in the network. This measure could be justified at least for outgoing calls, where UPT calls have to be checked for approval.

15.4.2 Example of Registration of a UPT Subscriber

Figure 15.8 presents an example of what a communication between UPT subscriber and UPT service provider could look like with a dialogue control over the telephone.

This UPT function can also be carried out over analogue connections for terminals, e.g., using DTMF signalling.

First the user dials the access code, which can be in the form of a previously unused dialling code (in the example #0185). After he has authenticated himself by entering his own UPT number and his personal identification number, he dials the menu for registration. He then requests registration for incoming calls. Now he can notify the UPT service to which terminal the calls should be directed and for how long. This completes the access procedure to the UPT service.

In man–machine communication, especially acoustic communication, it is particularly important to allow a person a chance to return to the previously

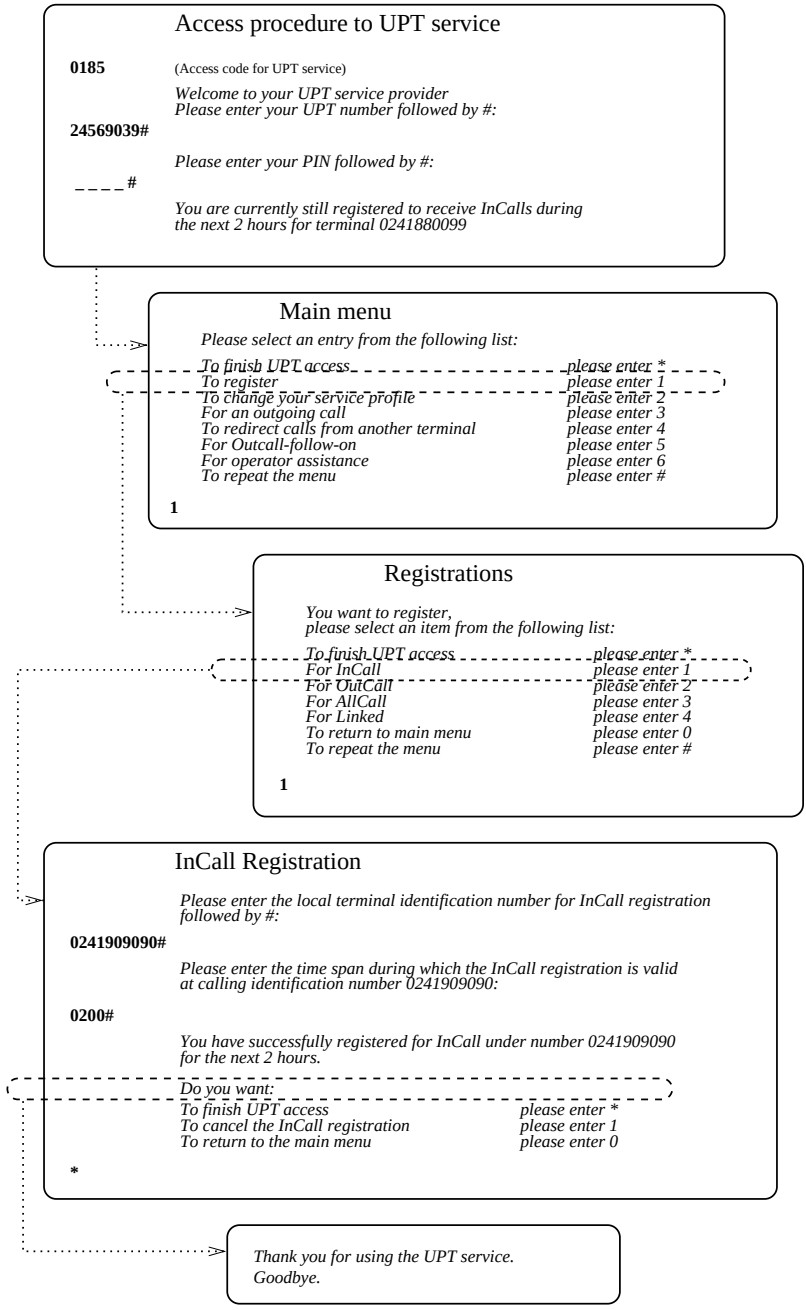


Figure 15.8: Example of an in-call registration with a UPT service provider

processed dialogue levels to enable errors to be corrected. In addition, the user should be able to have personal contact with an operator if assistance is needed.

15.4.3 Options for Authentication

For security reasons a UPT user must be authenticated before UPT resources are accessed. This can be done in different ways.

- The easiest way, which requires no intervention in terminals or the network, is authentication using a DTMF transmitter or by dialling certain numbers at a terminal. These signals are picked up by an appropriate functional unit and forwarded for evaluation. This type of authentication is regarded as weak, because it does not provide any special security. However, this approach can be implemented in every network and at every terminal, which is also a requirement of the ITU-T [21]. With this model the authentication data travels over the traffic channel, which has already been switched and puts no strain on the signalling network.
- A variation of the above option involves using an intelligent terminal for an automatic execution of the authentication. The terminal is then responsible for transmitting the DTMF signals. This option does not require signalling functions either.
- Card readers and smart cards offer a higher level of security and are easier to use. As soon as authentication is required, the card-reading device reads the inserted smart card and carries out the authentication. The signalling can be implemented over traffic channels or over signalling channels.

Thus different versions/variations, which are also able to offer different levels of user comfort, are possible, depending on the terminal intelligence or authentication security required.

15.5 UPT Service Profile

One of the elements that determines the functionality of UPT is the *Flexible Service Profile* (FSP). This is part of a database, and is used to supply the data associated with UPT users to the *Service Control Functions* (SCFs). One of the most important entries in the service profile is the terminal identification that a user has registered for incoming calls for a specific period of time. With each call, irrespective of whether it is an incoming or an outgoing call, the FSP must be interrograted for certain data relating to the call. A differentiation is made between permanent and variable entries in the structure of the FSP. Permanent entries relate to data that is fixed when the UPT identification is

established and seldom needs to be changed. Variable entries concern data with values that change owing to actions of the user, such as change of location.

Permanent entries that can be changed only by the service provider include:

- UPT number
- Identification of the home local terminal connection
- Basic services contracted by the subscriber
- Supplementary services contracted
- Number of maximum possible authentication attempts
- Type of authentication procedures
- Security options
- Barred destination numbers (e.g., police, speaking clock)

Permanent entries that can be changed by the subscriber include:

- Approved payment options (e.g.,*credit card calls*)
- Maximum ceiling on calls
- Maximum number of accesses from other terminals
- Released user functions

The following variable entries relate to service control:

- Type of active authentication
- Active option for charging
- Active security mode
- Status of supplementary services

The following variable entries are used for mobility control and supervision:

- Relevant terminal number
- Fallback terminal for incoming calls
- Relevant terminal for incoming calls
- Standard number of outgoing calls per registration
- Destination for call forwarding—absolute
- Destination for call forwarding—when terminal is busy
- Destination for call forwarding—when there is no answer
- Destination for call forwarding—if called party cannot be reached

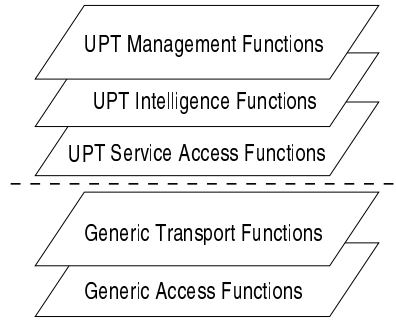


Figure 15.9: Functional grouping of UPT

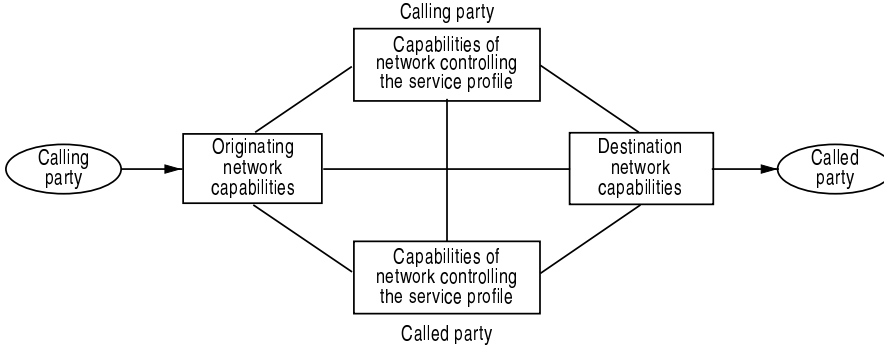


Figure 15.10: Reference model for UPT calls

15.6 Requests to UPT-Supported Networks

A model for the functional architecture of UPT was developed in I.373 [14] to enable the formulation of requests to the UPT-supporting network. Figure 15.9 illustrates this model, which consists of five layers hierarchically positioned underneath each other, which will later be related to the network. This abstraction is desirable to avoid the need of having to specifiy any specific form of network. UPT can thus be implemented in fixed as well as in mobile networks.

A UPT call can cross different networks on its route from the person placing the call to the person receiving the call. These networks have different capabilities for supporting UPT. Connections for traffic channels and for signalling exist between the different network components such as the network and service profiles. I.373 provides a reference model for UPT calls to present an overview of the dependences between the individual components (see Figure 15.10). For each UPT call a different sequence of communication between

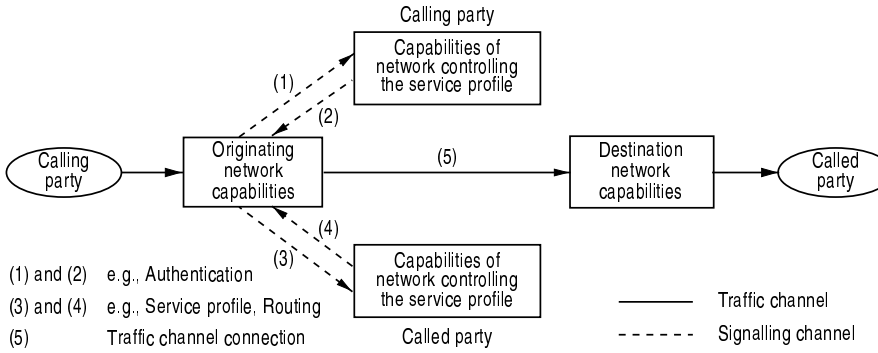


Figure 15.11: Example of a UPT call: call between two UPT users

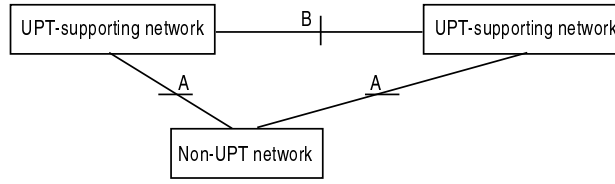


Figure 15.12: Interfaces in a heterogeneous network configuration

the network components and the service profile is conceivable. Only an example of a call between two UPT users is shown here (see Figure 15.11).

When UPT is offered in heterogeneous network configurations, conflicts can occur because networks have different capabilities for supporting UPT.

Figure 15.12 shows where interfaces occur between networks supporting UPT and non-supporting UPT. Two possibilities for handling UPT connections across network boundaries are to be considered. For networks that do not support UPT, either no UPT service or only a limited UPT service can be offered.

The same options apply to UPT connections that are to be routed over non-UPT-supporting networks. The first option is very unsatisfactory, which is why the second option should be taken into consideration even if additional costs are involved.

15.7 PSCS as a Further Development of UPT

Personal Services Communication Space (PSCS) is a service concept that was developed within the framework of the EU research programme RACE by the Mobilise project partners [5]. PSCS represents a further development of UPT phase 1, and is serving as the basis for projecting the services and functions of UPT phases 2 and 3. It enhances the UPT service in the following ways:

- User-friendliness during access
- Use of services beyond telephony, such as fax, e-mail and data services
- Easier access procedures to the service profile
- Flexible routing schemes, such as e-mail forwarding to a fax device

To achieve a certain level of user-friendliness, PSCS requires special terminals that have at least an alphanumeric display and a card reader for smart cards. Consequently, demands that UPT allow access from any terminal cannot be met, but what is gained is a far more flexible man–machine interface.

15.8 Numbering and Dialling

In telecommunications networks users are addressed by sequences of digits. By entering the telephone number of the party being called (user B), the calling party (user A) is instructing the network to set up a connection. For technical reasons, direct dialling was previously used to route a call across the network elements. On the basis of how it is structured, the telephone number can provide a direct indication of where user B is located with a direct dialling system. A telephone number therefore has a direct relationship to the location of user B, and also determines how a call is routed through the network. Because of the individual numbers that make up the telephone number, the switching centre is able to route the call to user B.

In addition to addressing users, the telephone number is used for the billing of telecommunications services. This gives the telephone number economic significance along with its technical importance.

This direct control has now been replaced by indirect dialling systems, which is why it is necessary to establish an association between telephone number and location of user B. In an indirect dialling system the routing of a call is largely separate from the telephone number. A path established from the routing tables filed in the switching centres is switched between the two users. In this case the telephone number only serves for the addressing, and does not contain any information about a subscriber's location.

However, users of telecommunications services have the knowledge to determine from the telephone number the calling area of user B, the charges and type of telecommunications services required. It is therefore important to retain this information function for the acceptance of new numbering systems.

Current and future numbering systems are explained in the following sections.

15.8.1 ISDN, PSTN

Telephone numbers in public telecommunications networks as well as in ISDN for a long time period consisted of 12 digits. Since the end of 1996, this



CC Country Code, defined in E.164
NDC National Destination Code
SN Subscriber Number

Figure 15.13: Structure of an ISDN telephone number

number has been extended to 15 digits [10]. The telephone number structure is provided in ITU Recommendation E.163 [7]; see Figure 15.13.

The country code for Germany is `49`. In their structure, national destination codes indicate the direct dialling control of the network that existed until about 1977. Like the network, they are arranged hierarchically, with

- position 1 for the central switching centre
- position 2 for the home location switching centre
- position 3 for the customer switching centre
- position 4 for the subscriber switching centre

The national destination code is different for each local network, and typically unique for a whole country. It consists of two to four digits, depending on the size of the local network.

A user within the same local network can be reached through his respective telephone number. Parties in Germany beyond one's own local network can be reached through dialling the prefix `0` followed by the national destination code and the respective telephone number. Users outside Germany can be reached through dialling the international prefix `00`, the country code and the respective telephone number. The length of the telephone number to be dialled therefore depends on the distance between the two communicating parties.

Special services can be selected by dialling special prefixes before the actual telephone number of the person being called. A `1` is reserved as the first digit for this purpose. For example, the GSM networks in Germany are accessed from the fixed network through dialling a `17x`.

Telecommunications connections are charged based on the national destination code. Knowledge of the structure of the national destination codes provides the user with an information source for determining the cost of a call. According to newer tariffing models there will be probably only two national tarif zones in the future: local and long distance. Then there will be no more advantage for the user to differentiate from the destination code where the long-distance call party is located, and the acceptance of the indirect dialling system will grow.



MCC Mobile Country Code, PLMN Country Code
MNC Mobile Network Code, PLMN Network Code
MSIN Mobile Station Identifier Number

Figure 15.14: Numbering structure in GSM

15.8.2 Public Mobile Telephone Network—GSM

The numbering used in the public mobile telephone network is similar to that of PSTN. ITU Recs. E.212 [8] and E.213 [9] provide the relevant structures. The GSM numbering scheme is based on these recommendations.

Figure 15.14 shows how a public mobile telephone number is structured. It consists of the respective country code, the network code and the terminal identification number. A complete integration into the ISDN/PSTN numbering plan could have been possible, but was not implemented. Instead, as mentioned above, GSM networks are differentiated from the PSTN through separate dialling codes which function as a traffic discrimination feature.

15.8.3 UPT

The ITU presents proposals for UPT numbering in Rec. E.168 [13]. In these proposals a UPT number is defined as a user's unique identification, and is dialled by a caller to reach the user. The UPT indicator is the part of the UPT number that identifies the call as a UPT call, e.g., in Germany `0700` is used as the prefix.

From the standpoint of the user, the UPT number must meet the following requirements [12]:

- A UPT number must be recognizable as such by users, so that they have the opportunity to determine that a call is a UPT call and therefore will be handled and charged in a special way.
- A UPT number must be as short as possible to minimize the number of digits to be dialled.
- It must be possible to dial the UPT number from any terminal in the PSTN. Therefore digits have already been reserved (digits `0...9` and possibly the characters `#` and `*`).
- UPT users should be able to retain their UPT numbers if they change service provider.
- In the future the UPT number should be valid in all networks, at all terminals and for the use of any service.



Figure 15.15: Subscriber-based UPT telephone number

- Any further developments and changes to the UPT numbering plan should have as little an effect as possible on existing UPT numbers.

Network operators have other requirements of UPT numbers:

- It must be easy for a network to recognize a UPT call.
- The numbering capacity must be protected even when UPT is introduced.
- UPT numbers should have no influence on routing.
- The administration of UPT numbers must be kept simple.
- UPT numbers must fit into existing numbering plans (see E.164) [11].

Recommendation E.168 differentiates between three scenarios that all comply with E.164 and therefore are aimed at meeting the requirements of both user and network operator.

15.8.3.1 Scenario 1—Subscriber-Based Concept

The subscriber-based concept gives no indication to the subscriber that a call is a UPT call (see Figure 15.15). All information concerning the UPT subscriber is administered by the *Flexible Service Profile* (FSP) at his home location. In scenario 1, `CC` remains the country code, `NDC` the national destination code and `SN` the subscriber's telephone number. `NDC` and `SN` together produce a unique telephone number for a country. Owing to the lack of other criteria, the entire number has to be evaluated and searched in a database to determine whether the call is to be recognized as a UPT call.

15.8.3.2 Scenario 2—Country-Based Concept

In the country-based concept, `CC` retains its significance as the country code (see Figure 15.16). Unlike scenario 1, however, `NDC` identifies a call as a UPT call. This applies to the party making the call as well as to the network. Special codes can be used to select UPT services or a UPT service provider instead of the local network. This concept has already be applied in the integration of GSM mobile radio networks. For example, the first NDC digit can select the UPT service while the other digits would be responsible for the selection of the UPT service provider.

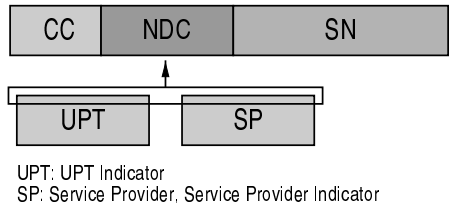


Figure 15.16: Country-based UPT telephone number

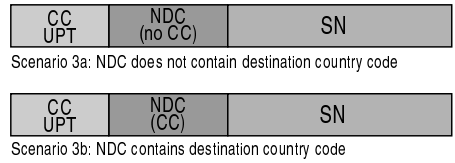


Figure 15.17: Country code-based concept for UPT telephone numbers

15.8.3.3 Scenario 3—Country Code-Based Concept

The country code-based concept indicates `CC` as the UPT indicator (see Figure 15.17). There are two possible options for the NDC field:

- `NDC` without country code or
- `NDC` with country code

The first option (scenario a in Figure 15.17) does not give the user the possibility to establish the destination country of his call. The caller therefore has no way of estimating the cost of the call on the basis of the telephone number. In this case other sources of information must be used to obtain some indication of cost. Furthermore, with this numbering scheme the telephone number is distributed globally, whereas situation b in Figure 15.17 allows for a national administration of the telephone numbers.

15.8.3.4 Relationship Between UPT and Other Network Numbers

Figure 15.18 indicates which ITU recommendations could have an influence on possible UPT numbering scenarios[13].

The UPT number should provide the network with sufficient information for using databases to establish a subscriber's network number.

15.8.3.5 Responsibility for Number Allocation

The different scenarios produce different requirements for the administration of the numbers. Table 15.1 indicates who has responsibility for the allocation of numbers in each case. Because central allocation appears less flexible than decentral allocation, scenario 2 would be the preferred one.

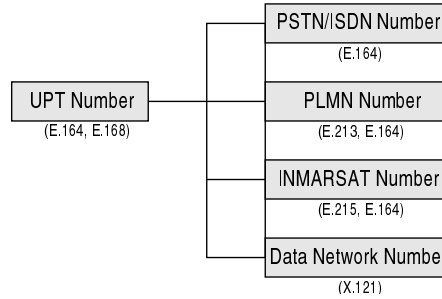


Figure 15.18: Relationship between UPT and network numbers

Table 15.1: Responsibility for number allocation

Scenario	Country code	National dest. code	Subscriber telephone no.
1	ITU	National	National
2	ITU	National	National
3	ITU	(a) ITU	ITU
		(b) ITU	National

15.8.3.6 Effect of Numbering on UPT Subscribers

The numbering used in UPT has a major effect on acceptance by potential UPT users. One just has to imagine being allocated a 15-digit combination of numbers without any discernible structure as a personal telephone number for UPT. For example, whereas until now user A has been able to reach eight different users in the same local network using a six-digit telephone number and five different users in other local networks but with possibly the identical national destination code, this same user will now have to remember 13 different 15-digit UPT telephone numbers. This situation surely will not contribute towards the success of UPT.

Because a call has to be recognizable as a UPT call to the calling party, scenario 1 from E.168 would not appear to be practical. With scenario 3 a user would always have to dial the complete 15-digit telephone number in order to reach a UPT subscriber. This solution does not seem user-friendly either. In comparison, scenario 2 offers the possibility of omitting the country code with calls within a country (after all, only about 3 % of all calls made in Germany are international calls). Thus each country could create a numbering plan tailored to its own needs. However, there has to be an agreement on how the same UPT indication numbers are used.

According to the scenarios discussed, the UPT number would also have to be changed if there is a change in UPT service provider because the UPT indicator for the service provider would be different. There is a strong commitment to remaining with a provider because of the costs involved in making

a number change. This bond to the provider can be broken if the *Personal User Identity* (PUI) can be switched as the indirect address between UPT number and the *Flexible Service Profile* of the party called. The PUI is contained in a database, which allocates a UPT number to the address of the FSP. This allows the UPT number to be retained if there is a change in the FSP's service provider or home location area. It then suffices for the address of the FSP to be changed in the PUI. The UPT number thereby has the significance of a true, possibly permanently valid, personal telephone number. For subscribers who have never changed their FSPs (this probably applies to most subscribers) the PUI would be a one-to-one duplication of the UPT number to the FSP address.

Although interrogation of a PUI database introduces some access delay, there are advantages to this approach that increase the acceptance of UPT:

- Users are assigned a true personal telephone number, which in turn is also easy to remember.
- The network operator is able to change the internal processes in his UPT network without having to change the UPT number. This allows him to change the structure and the location of his FSP databases without any problem.

The disadvantage when a UPT subscriber changes from provider A to provider B is that the PUI from provider A has to route every call to provider B. In this case other database distributions have to be taken into account:

1. A global database stores the PUI for all UPT subscribers.
2. A national database stores the PUI for all UPT subscribers whose first service provider was based in that country.
3. A database in the country of the party making the call holds the PUI.
4. As described above, the UPT provider holds the PUI for the subscribers who first used UPT with him.

All the possibilities have advantages and disadvantages. The second solution seems the most practical. Mobility between countries is not as great as between service providers. Every country or alliance of countries (e.g., the EU or ETSI) can therefore take responsibility for its own allocation of personal telephone numbers. The third solution creates problems in terms of data consistency, whereas the fourth solution saddles UPT provider A with tasks from UPT provider B. This leaves the former managing data on subscribers who have not been his customers for a long time.

From the standpoint of user-friendliness, priority is given to the concept in which a PUI guarantees the personal telephone number. The time delay associated with the additional database access can be alleviated through caching strategies.

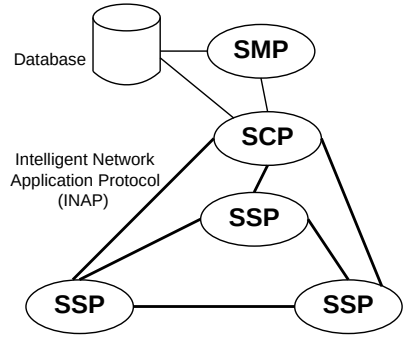


Figure 15.19: The principles of the functioning of an intelligent network

15.9 Intelligent Networks and Their Value-Added Services

Now whenever a new service is installed in a telecommunications network, the control programs in each switching centre have to be updated, which is an enormous expense in terms of time and cost, and also involves operative risks that are difficult to assess.

Because of the growing emphasis on the ability to introduce services quickly and flexibly in the future, efforts were begun in the USA and in Europe in the late 1980s to develop the concept of *Intelligent networks*, IN [1]. This is not a new telecommunications network, but instead an architecture designed for the further development of existing networks and new developments of future networks.

With INs certain network services that extend beyond pure switching and transmission functions will be concentrated in autonomous network nodes. This will allow new services to be implemented centrally rather than from each switching node, thereby accelerating the process of introducing these services into the network.

The concept of intelligent networks is being standardized in international recommendations of the ITU in several *capability sets* (CS) [22]. So far only the recommendations of the first capability set (CS-1) have been passed. This section provides an overview of intelligent networks, with special emphasis on their role in the implementation of UPT services.

15.9.1 The Functional Principle of an Intelligent Network

An important advantage of intelligent networks compared with a conventional network architecture is in the separation of switching and service-provisioning functions. Figure 15.19 clarifies this principle.

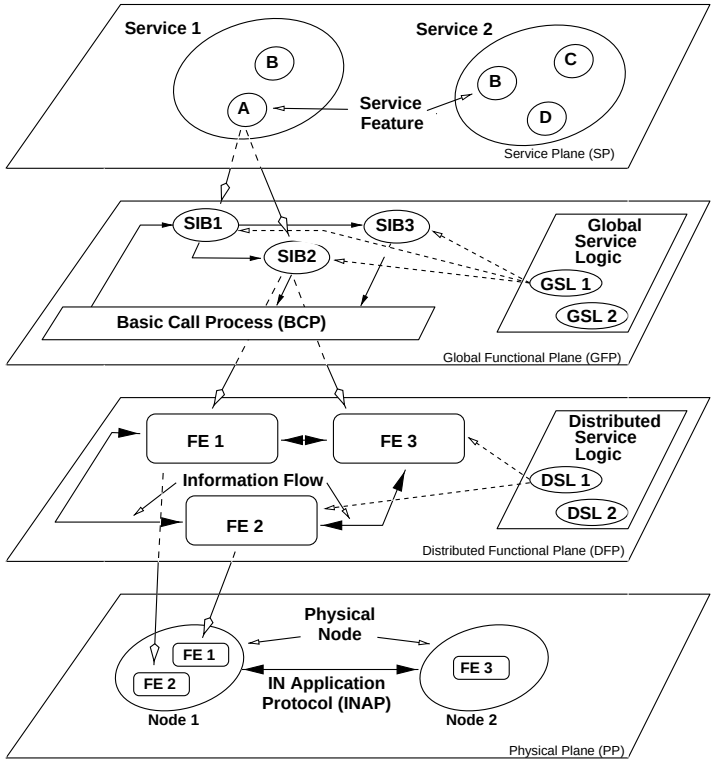


Figure 15.20: Layer model for description of services in intelligent networks

IN calls are recognized at digital *service switching points* (SSP) according to service codes (e.g., `0800` for a no-charge call in the network of Deutsche Telekom AG). The Intelligent Network Application Protocol (INAP) is used to instruct the *service control point* (SCP) of the IN call. The SCP controls the execution of the IN service. The *service management point* (SMP) is responsible for setting up, changing, administering and monitoring the IN services. All necessary parameters of the IN services are stored in the database attached to the SMP, which the SCP also accesses when providing a service.

15.9.2 Description of Services in Intelligent Networks

IN services are described at four different planes in a *conceptual model* of the intelligent network IN conceptual model (INCM) (see Figure 15.20); these planes are interrelated with each one dealing with a different aspect of IN services.

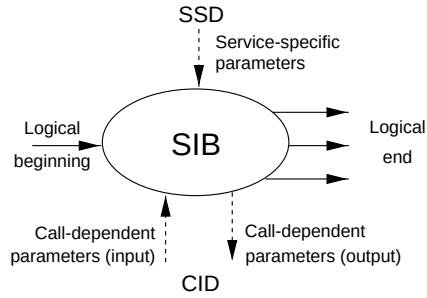


Figure 15.21: Structure of a service-independent building block (SIB)

Service plane (SP) At the service plane a service is described solely from the standpoint of the user on the basis of different *service features* (SF) and through usage scenarios [16].

Global functional plane (GFP) The global functional plane breaks down a service into *service-independent building blocks* (SIB), which are used in sequence to provide some service [16], based on the description in the service plane. Service support data (SSD) and call-instance data (CID) as well as a logical beginning and a logical end to the execution are defined for configuration of a SIB module (see Figure 15.21). The operational control process and the initialization of the service-independent and call-instance data is carried out by the *global service logic* (GSL).

A special role is played by the service-independent module BCP (*basic call process*, or basic call sequence), which functions as the central interface to the switching and transmission functions.

Distributed functional plane (DFP) A service-independent building block (SIB) is implemented by one or several *functional entities* (FE) in the distributed functional plane [17]. The interaction between two functional entities takes place through information flow (IF). Figure 15.22 shows the functional entities of the distributed functional plane and how they are linked.

Functional entities are divided into those that provide and those that manage IN services. The functions for providing IN services are shown in grey in Figure 15.22, and comprise the following:

Call control agent functions (CCAF) The call control agent function is the interface between the user and the switching functions of the network.

Call control functions (CCF) The call control function is responsible for controlling and setting up connections. The CCF also provides a mechanism for detecting so-called *trigger points* (TP) for IN services.

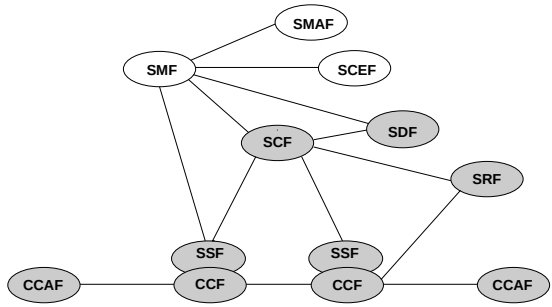


Figure 15.22: Functional entities in the distributed functional plane

Service switching functions (SSF) The service switching function enables cooperation of the call control functions of a network (CCF) with the service control function (SCF).

Service control functions (SCF) All functions used for providing an IN service in the network are carried out as service control functions. To do so, the SCF has access to the data that is supplied by the SDF.

Service data functions (SDF) This function stores all data that is required to provide, execute and bill the IN services.

Specialized resource functions (SRF) A specialized resource function provides resources, such as functions for voice input and output, that can be used by the user to interact with the network.

The other functional entities serve in the management of the IN services, in other words in providing and managing the services, and consist of the following:

Service creation environment functions (SCEF) This function can be used to create new IN services or to change existing ones.

Service management functions (SMF) The service management function provides the functions necessary to activate services from the SCEF for availability in the network.

Service management access functions (SMAF) This access function provides the administrator of IN services with access to the SMF.

Physical plane (PP) The functional entities of the distributed functional planes are allocated to the nodes of the intelligent network at the physical plane [18]. A sample model of a physical node is illustrated in Figure 15.19. Protocols implement the information flow between different nodes. The Intelligent Network Application Protocol (INAP) [15, 19] is specifically discussed in this context.

Table 15.2: Distribution of INAP signals over functional components

Information flow			ITU	ETSI
SCF	→	SSF	27	19
SSF	→	SCF	23	7
SCF	→	SDF	3	
SDF	→	SCF	2	
SCF	→	SRF	2	
SRF	→	SCF	1	

15.9.3 The Intelligent Network Applications Protocol

In the capability set 1 (CS-1) the Intelligent Network Application Protocol (INAP) enables interaction between the following functional entities of the distributed functional plane:

- Service switching function (SSF)
- Service control function (SCF)
- Service data function (SDF)
- Specialized resource function (SRF)

The *Remote Operations Service Element* (ROSE) of the OSI reference model is used in the *Intelligent Network Applications Protocol* (INAP) to transfer the protocol data units (PDU), thus defining the INAP as a client-server system. The protocol data units of the ROSE application protocol are in turn transmitted through messages in the *transaction capability application part* (TCAP). The transmission is carried out by the *Application Service Element* (ASE) of the application part for transaction processing (TCAP). The IN service in turn accesses the remote operations of the INAP using an ASE.

Abstract Syntax Notation No. 1, (ASN-1) [6, 23] is used in defining the protocol data units of the INAP. The macro definition `OPERATION` indicates the name of each remote operation as well as optionally the invoking parameters (`ARGUMENT`), return values (`RESULT`), error messages (`ERRORS`) along with linked operations (`LINKED`) (see Figure 15.23).

Reference [19] defines 58 remote operations for capability set 1 (CS-1) of an intelligent network. Only 34 remote operations have been accepted by ETSI in [3, 4] for Europe-wide standardization. Table 15.2 shows that most requests for remote operations take place between SCF and SSF.

15.9.4 UPT in the IN Layer Model

The functional architecture of UPT was presented in Section 15.6. The functional layers shown there can be assigned to the IN functional units in an

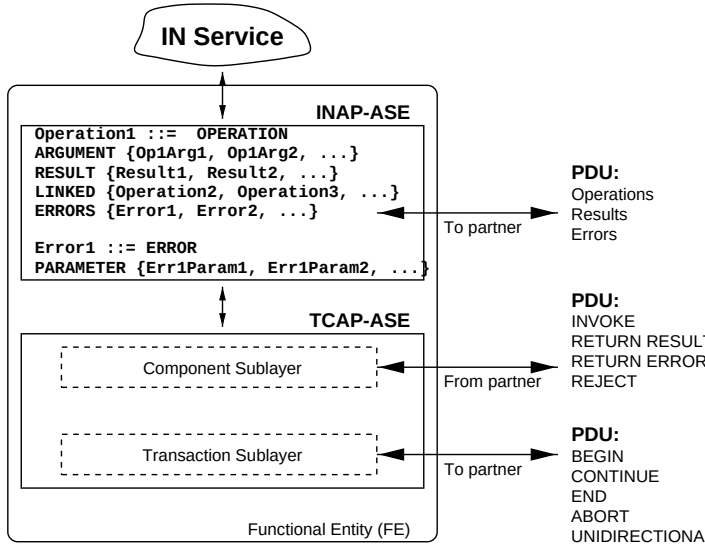


Figure 15.23: Definition of the INAP protocol data units with ASN-1 and transmission using the application service element TCAP

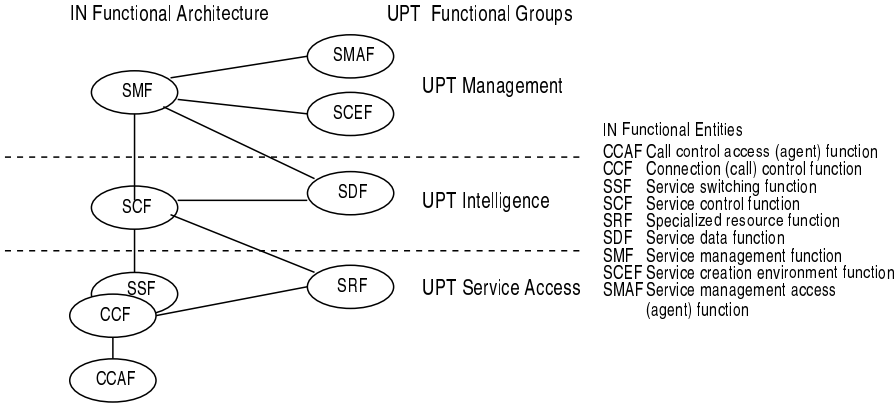


Figure 15.24: Mapping of UPT functional groups to the IN layer model

IN-based solution. Figure 15.24 shows that intelligent networks offer a good basis for the introduction of UPT.

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16
The Future is Wireless

16.1 A Typical Day in the Year 2000

Joe Mobilus is a manager with a haulage firm in a city 100 km away from his home town; his wife Sue is a nurse at the local hospital. Their son Tom and daughter Ann are still at school.

06:30 The central alarm system wakes Joe and Sue in line with their weekly schedule.

07:00 Joe uses his TETRA terminal to access his firm's central computer to see if there have been any changes to his diary.

07:15 Joe leaves his house and using his cellular phone in the car calls the traffic service to find out about current traffic conditions.

07:30 Tom, who is finishing school this year and wants to become a stockbroker, calls up a stock market service on the Internet to have the share prices from New York conveyed over satellite. Then, together with his sister, he leaves to go to school. The Mobilus household is linked to the PSTN/ISDN over an RLL system.

08:00 Sue's cordless DECT telephone rings while Sue is in the bath. Her mother informs her that she has to postpone the joint dinner that was planned for that evening.

08:10 Sue informs Joe accordingly, over a pager or the GSM short-message service.

08:15 Joe's car breaks down as he is driving to work. He activates the automatic emergency service for his car to call for help with a puncture. To do so, he uses a TETRA service or GSM-SMS.

08:20 Because Joe is not sure whether he will make it to the office on time, he sends his secretary an e-mail from his mobile data terminal with the message that she should change his 9 o'clock appointment to a later time. This message is served by GPRS, a non-transparent GSM data service.

10:00 Joe has arrived in his office, where he receives a message that a client's warehouse flooded because of heavy rain during the night and that the goods being delivered should be diverted to a different address.

10:05 The firm's TETRA system enables Joe to make a group call to the vehicles that are just on their way to the warehouse to notify them of the new address.

11:00 A customer order requires Joe to spend a long time working closely with Mr. Jones, another employee of the firm, who has his workstation on another floor of the building. To save time, Mr. Jones has his workstation transferred next to Joe's. This can be done with no problems, because the company has installed a HIPERLAN.

13:00 Sue, who has been working in the hospital for four hours, is just looking after the patient in room 8 when she is paged over ERMES. The ERMES terminal indicates that she is urgently needed in casualty admissions.

14:00 Joe has to talk to several colleagues jointly at the branch in Russia about a decision concerning a cooperation agreement with a Russian haulage company. He is able to reach them easily over the satellite radio system IRIDIUM, but video conferencing is unfortunately not available.

15:00 Tom is called at home by a school friend. Because the friend was ill, he was unable to attend classes today. He asks Tom to send him the homework that was handed out in maths class today. Using the fax machine connected to the DECT system, Tom is able to send the information that was requested.

17:00 Ann, who is out shopping with her friend Jane, finds a nice shirt that she wants to give her father for his birthday. Because she is not sure which size to take, she goes to a pedestrian zone to call up her mother using her DECT handset over a Telepoint service.

18:00 On the way home, Mr. Mobilus is informed over his cellular phone that he has to visit a customer in Rome the next day. He is able to book his air tickets and hotel reservations through a travel service from his car.

19:30 After an evening meal with his family, Joe arranges for the data he needs from his firm for his appointment in Rome to be displayed on his GPRS/GSM data terminal so he can have it printed out.

16.2 Wireless Communication in the Year 2005

Instead of having many different mobile radio systems and the associated terminals, each modern user will have a software radio that, using the software-defined configuration of the mobile terminal, adapts to the grade of service

of the radio interface, depending on the service and the location, and largely makes decisions independently.

Because of the flexibility that this allows the user in selecting services, the mobile network operators have entered into a price war and are offering confusing, *particularly reasonable* rates supposedly restricted to certain user groups. Modern users have a program installed within their terminal that always selects the best rate specific to the current receive situation and to the service to ensure that the software radio is switched to the corresponding mode and the most favourable mobile radio service is being used.

UMTS-based PLMNs have been established, but are still available only in highly crowded areas. The main mobile radio service in the large areas is still served by second-generation systems like GSM.

There is no longer a restriction to narrowband services in hot communications traffic spots (like airports, sports arenas, fairs, etc.), because even broadband (ATM- and IP-based) services are now available as wireless services at broadband terminals. However, the costs for these services are still so high that only business users can take advantage of them and afford the appropriate terminals.

Since the narrowband telecommunications services of the fixed network (telephony and data) have already been widely introduced over the Internet and have bearable real-time characteristics, the packet-based GPRS speech transmission is being extensively used for mobile calls because it is less expensive than the standard GSM telephone service. The GPRS has also been adopted by the UMTS.

All mobile radio services have fallen dramatically in cost of usage and are at almost the same cost levels as the corresponding fixed network services, which are much lower in prize than in the late 1990s.

16.3 Closing Remarks

Our society is being characterized by increasing individual mobility and the growing need for an exchange of information.

To be successful in domestic and international markets in the future, companies and their employees inevitably have to be contactable at all times, wherever they are and with the greatest confidentiality, and also have to have access to the most up-to-date information.

The more mobile employees of a company are, the more difficult it is for them to be reached and to have access to important information early enough unless they use mobile radio services. Because of the variety of application areas they are used in, mobile radio systems are also becoming increasingly important for private use. Thus newest-generation mobile radio systems, which have established themselves in the commercial world, will become increasingly cheaper through mass production, and therefore will appeal to an ever-growing group of customers.

The advantages of mobile radio systems and the evident trend towards universal systems in the future, which enable parallel transmission of voice, text, data and images over one connection, thus allowing multimedia information exchange based on UMTS and IMT 2000, will lead to a rapid increase in the number of mobile radio users.

Third-generation systems will combine radio-based and fixed networks into a standard architecture. This kind of network will consist of public and private terrestrial components, as well as ground or space components from radio-based networks combined in a way that a subscriber can use a standard terminal, the so-called *Personal Communicator*, to take advantage of all the services offered by all networks available today and those that will be operating in the next few years.

There is no conclusive response to the question repeatedly being discussed in the media about possible repercussions to health arising from cellular telephone use. Intensive studies are being conducted today on the biological effects of electromagnetic radiation in the mobile radio area in order to define meaningful limit values for acceptable radiation levels.

Because they are almost always physically separate, fixed radio facilities for mobile telephone systems are only playing a minor role as potential sources of danger due to electrosmog. Mobile equipment with integrated transmitting antennas that radiate high-frequency waves near the head warrant more concern. Most of the absorbed energy is converted into heat, and increases the temperature of the body. In tests carried out on animals and simulated human heads it has been determined that the metabolism, the nervous system and behaviour can be affected even with an additional temperature rise of 1 ℃.

The World Health Organization and the national offices for radiation protection of many countries therefore recommend threshold values for the specific absorption and maximum power output (EIRP) of transmitting facilities that are not to be exceeded, along with guidelines on the minimum permitted distance between equipment or its antennas and a person's body. Besides the thermal effect of electromagnetic radiation, non-thermal effects can also occur. These have been investigated for more than a decade, but there are still no definitive statements about possible impacts of mobile radio waves on humans.

The results of further scientific studies will influence the development and suitable protective measures as well as the acceptance of mobile radio systems. There is sufficient experience with human exposure to sources of danger, e.g., from cars and with electrical power.

Even if future research results point to a calculable danger from electrosmog due to handset use, this will hardly have any impact on acceptance. The danger-to-use consideration of individual mobile radio users will probably be similar to what happened with the use of cars and electricity.

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Appendix A
Queuing and Loss Systems

Queuing systems model real systems and enable the calculation of their performance parameters.

A.1 The Queuing System M/M/n-∞

This system models the interarrival times T_A from randomly received jobs as well as the random service times T_B of the jobs, each through a negative exponential (*Markov*, M) distribution.

The n servers are assumed to be identical, and free servers can be occupied in any sequence with the arrival of a request; when all servers are occupied, the job waits in one of the s queuing positions in the queue until it is serviced. Waiting calls are serviced in any sequence, which is not allowed to be based on the servicing time (see Figure A.1).

The system is directly suitable for modelling mobile radio systems with a large number of independent users who occupy one of several channels available in a cell for their communication. Some base station systems have a queuing option for calls while waiting for the the availability of one out of several trunked channels.

A.1.1 State Process as Special Birth-and-Death Process

State Space and Transitions

A state ($= 0, 1, 2, \ldots \infty$) defines the number of calls in a system (those being processed and those queuing). The state transition rates correspond to the arrival rate λ or the state-dependent servicing rate (see Figure A.2).

The traffic offered, which is

$$A = \frac{\lambda}{\epsilon} \tag{A.1}$$

is used to calculate the state probability for x calls in the system:

$$p_x = P(X = x) = \begin{cases} P_0 \left(\dfrac{A^x}{x!} \right) & \text{for } 0 \le x \le n \\ \\ P_0 \left(\dfrac{A^n}{n!} \right) \left(\dfrac{A}{n} \right)^{(x-n)} & \text{for } x > n \end{cases} \tag{A.2}$$

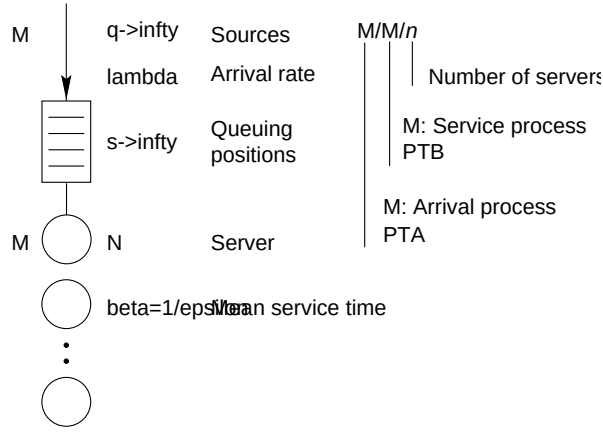


Figure A.1: Queuing system M/M/n-∞

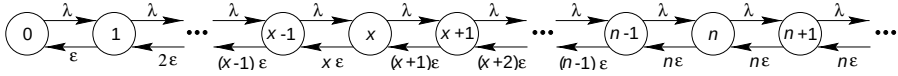


Figure A.2: State diagram of M/M/n-∞ queuing system

$$P_0 = \left[\sum_{i=0}^{n-1} \frac{A^i}{i!} + \frac{A^n}{n!} \sum_{i=0}^{\infty} \left(\frac{A}{n}\right)^i\right]^{-1} = \left[\sum_{i=0}^{n-1} \frac{A^i}{i!} + \frac{nA^n}{n!(n-A)}\right]^{-1} \tag{A.3}$$

In pure queuing systems ($s \to \infty$):

- The sum in Equation A.3 only converges if the stationary condition $A < n$ is fulfilled.
- The traffic carried (also called the traffic value) Y equals the traffic offered A if $A \leq n$; otherwise the system will become unstable and the queue increases without a limit.

The *utilization level* is given by

$$\rho = \frac{A}{n} \tag{A.4}$$

A.1.2 Characteristic Performance Parameters

Queuing probability p_w

The probability of an arriving call having to queue is

$$p_w = \frac{p_n}{1-\rho} \tag{A.5}$$

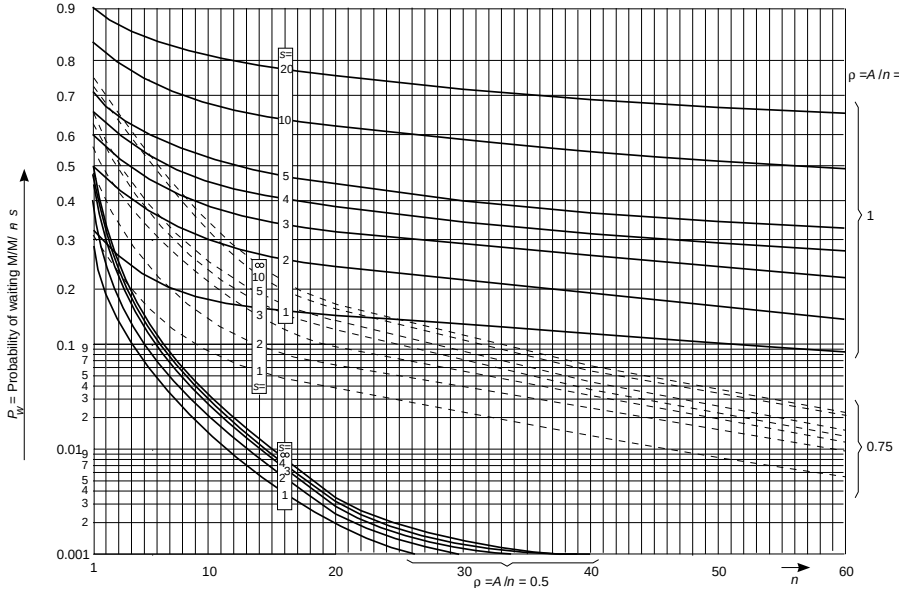


Figure A.3: Queuing probability in an M/M/n-s queuing system

Erlang queuing formula

$$p_w = P_0 \frac{A^n}{n!(1-\rho)} = \frac{\frac{A^n}{n!(1-\rho)}}{\sum_{i=0}^{n-1} \frac{A^i}{i!} + \frac{A^n}{n!(1-\rho)}} \tag{A.6}$$

Figure A.3 shows the queuing probability p_w as a function of the number of channels n, with the utilization level ρ as a parameter (see Equation A.4) for the models with $s = 1, 2, \ldots \infty$ queuing positions. $s = 0$ results in a loss system (see Section A.2).

Traffic value Y

Traffic value Y indicates the average number of servers occupied simultaneously.

$$Y = \sum_{x=0}^{n-1} x p_x + n \sum_{x=n}^{\infty} p_x = A \tag{A.7}$$

In pure queuing systems the offered traffic is the same as the carried traffic.

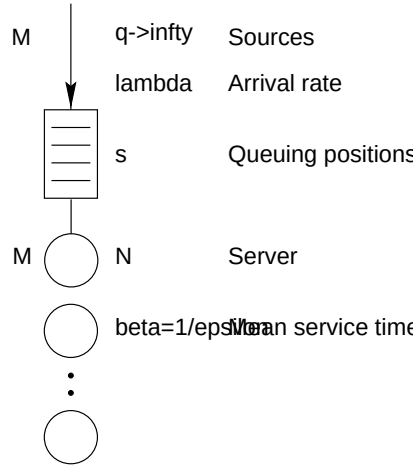


Figure A.4: Queuing-loss system M/M/$n-s$

Queuing load (average queue length)

L_Q is used to indicate the average number of queuing positions occupied. It is given by

$$L_Q = \sum_{x=n}^{\infty}(x-n)p_x = p_n \frac{\rho}{(1-\rho)^2} \tag{A.8}$$

Average queuing time W and W^-

- $W = E[T_W]$: average queuing time of all arriving calls:

$$W = \frac{L_Q}{\lambda} \tag{A.9}$$

- $W^- = E[T_W | T_W > 0]$: average waiting time for queuing calls only:

$$W^- = \frac{L_Q}{p_w \lambda} \tag{A.10}$$

The approximation of a stationary boundary, $A \to n$, makes $W^- \to \infty$.

A.2 The Queuing-Loss System M/M/n-s

Here the number of queuing positions is limited to $s < \infty$. If the traffic exceeds the number of servers available plus queuing positions then the calls above that number will not be processed (see Figure A.4). $s = 0$ equates to a pure-loss system.

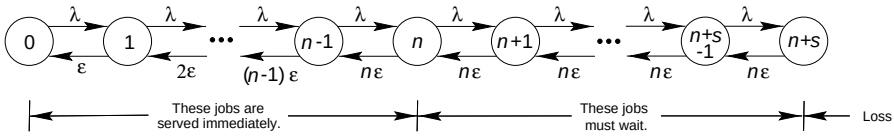


Figure A.5: State diagram of an M/M/$n - s$ queuing-loss system

A.2.1 State Process as a Special Birth-and-Death Process

State space and transitions

The state number ($= 0, 1, 2, \ldots, n + s$) of calls in a system (serviced and queuing) is

$$P(X = x) = \begin{cases} P_0 \left(\dfrac{A^x}{x!} \right) & \text{for } 0 \le x \le n \\ P_0 \left(\dfrac{A^n}{n!} \right) \left(\dfrac{A}{n} \right)^{(x-n)} & \text{for } n+1 \le x \le n+s \end{cases} \tag{A.11}$$

$$P_0 = \left(\sum_{i=0}^{n} \frac{A^i}{i!} + \sum_{i=n+1}^{n+s} \frac{A^i}{n!} \frac{1}{n^{i-n}} \right)^{-1} \tag{A.12}$$

See Figure A.5.

A.2.2 Characteristic Values

Loss probability

Calls are lost if no further ones can be accepted; thus the state probability for the state $n + s$ corresponds to the loss probability p_l:

$$p_l = P_{n+s} \tag{A.13}$$

Figure A.6 shows the loss probability p_l as a function of the number n of available parallel channels for different levels of utilization ρ of the channels and s queuing positions of the queuing-loss system. Figure A.7 shows the loss probability p_l as a function of the number n of channels for a pure-loss system, with the utilization level ρ as a curve parameter. The queuing probability is illustrated in Figure A.3.

Traffic value Y

$$Y = \sum_{k=1}^{n} k p_k + \sum_{k=n+1}^{n+s} n p_k \tag{A.14}$$

$$Y = A(1 - p_l) \tag{A.15}$$

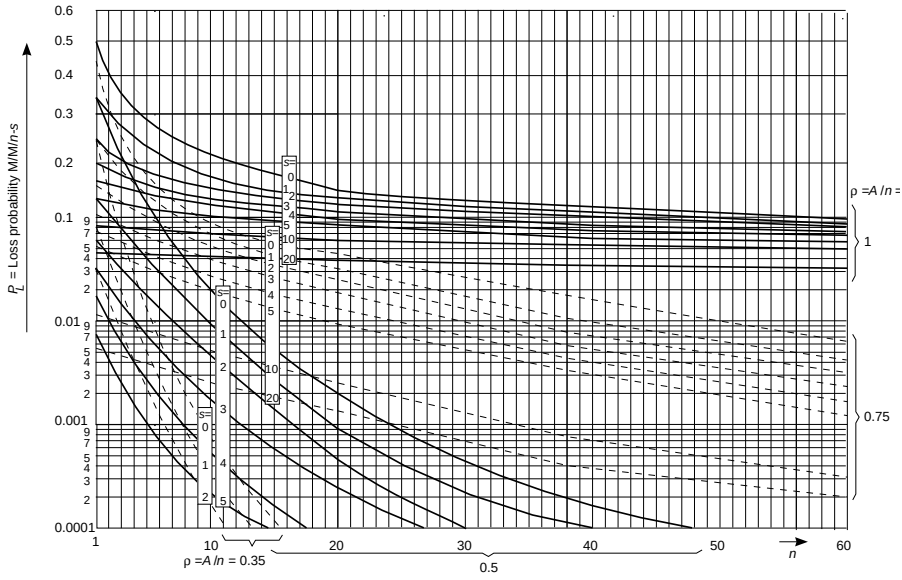


Figure A.6: M/M/n-s queuing-loss system

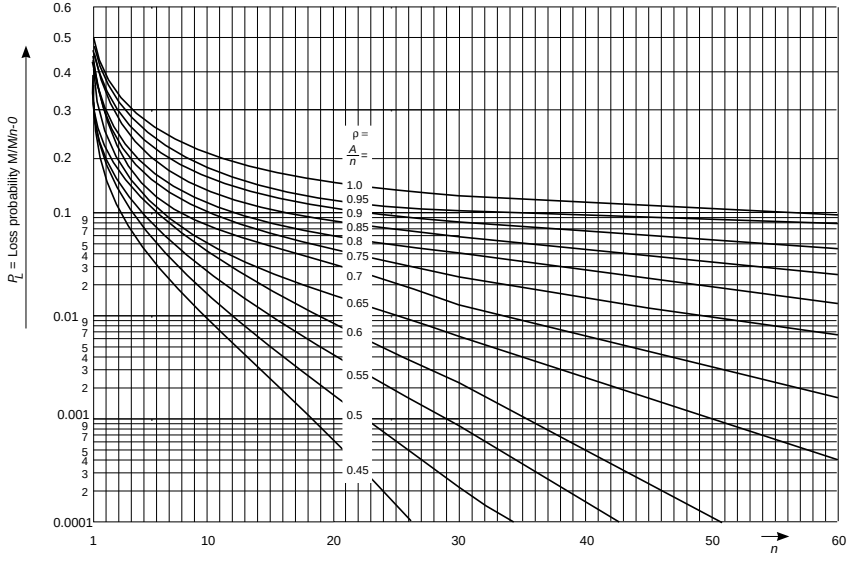


Figure A.7: M/M/n loss system

Queuing load L_Q

$$L_Q = \sum_{k=n+1}^{n+s} (k-n)p_k \tag{A.16}$$

Average Queuing Time Q and Q^-

- The average queuing time for all arriving calls is

$$W = \frac{L_Q}{\lambda} \tag{A.17}$$

- The average queuing time of the queuing calls only is

$$W^- = \frac{L_Q}{p_w \lambda} \tag{A.18}$$

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Appendix B
Standards and Recommendations

As a result of the rapid developments in communications and the variety of communications systems developed by different manufacturers all over the world, agreements on international, regional and national standards or recommendations are essential. According to the ISO, the definitions of a standard and a recommendation are as follows:

> **Standard:** "*A technical specification or other document available to the public, drawn up with the cooperation and consensus or general approval of all interests affected by it, based on the consolidated results of science, technology and experience, aimed at the promotion of optimum community benefits and approved by a body recognized on the national, regional or international level.*"
>
> **Recommendation:** "*A binding document which contains legislative, regulatory or administrative rules and which is adopted and published by an authority legally vested with the necessary power.*"

Standardization (or the drawing up of recommendations) in the area of telecommunications is defining specifications within a task area to allow as many manufacturers as possible to develop their products accordingly. This results in the elimination of trade barriers, thereby resulting in an international harmonization of telecommunications infrastructure. Other objectives being pursued in the mobile radio area include noise-free message transfer, optimal use of frequency spectrum and minimization of electromagnetic incompatibility between radio services.

On the other hand, product development is restricted owing to standards and the resultant compliance with certain regulations. Another disadvantage is the long time span between the standardization of a new system and its commercial introduction. An effort is being made to consider future technological innovation in standards in order to minimize the problem.

B.1 International Standards Organizations

The most important international standards organizations in the telecommunications area include the following:

- ISO: International Standardization Organization

- ITU: International Telecommunications Union
- IEC: International Electrotechnical Commission
- INTELSAT/INMARSAT: International Telecommunications Satellite Organization/International Maritime Satellite Organization

B.1.1 ISO

The ISO was founded in 1946 as an institution under the umbrella of UNESCO, and therefore falls within the scope of the United Nations. Its members for the most part include national standards organizations, which meet at a general assembly every three years.

The ISO has very extensive responsibilities. Its standardization work is divided into *Technical Committees* (TC) according to the different technical fields, which, depending on the scope of the work involved, are subdivided into *Subcommittees* (SC) and then into *Working Groups* (WG) and other groups.

The actual work itself is carried out by 100 000 volunteers in WGs all over the world. Many of these volunteers are delegated to the ISO by their employers whose products are being standardized. Others who are involved include government officials who are promoting one of their country's standards for international acceptance. Scientific experts are also in many of the WGs.

In general, the ISO's involvement in telecommunications is more from the standpoint of message transfer between computer systems. For example, some of the standards that were developed for *Local Area Networks* (LAN) are also applicable for voice transmission.

B.1.2 ITU

The International Telecommunications Union (ITU), with its headquarters in Geneva, was founded in 1865 and is one of the oldest international organizations in existence. As a subsidiary organization of the UN, it is active in four different task areas today:

- International allocation and registration of transmit and receive frequencies
- Coordination of efforts in the elimination of interference in radio traffic
- Coordination of the development of telecommunications systems
- Preparation of agreements on performance guarantees and tariffs

The tasks, rights and duties of the member states of the ITU are regulated in the *International Telecommunication Convention* (ITC), the international telecommunications contract. Every three to four years, the proposals that have been drawn up are presented to the general assembly, and with a high

enough quorum are published as recommendations. Internationally these recommendations have the weight of standards. This approach has now changed, and recommendations are now introduced on CD-ROM.

The *World [Administrative] Radio Conference* (W[A]RC) was convened by the ITU to revise the *Radio Regulations* to meet current needs.

Until the end of 1992, the standardization work of the ITU was carried out by five committees based in Geneva [1] and entrusted with the following areas of responsibility:

- CCITT: Consultative Committee for International Telephone and Telegraph
- CCIR: Consultative Committee for International Radiocommunication
- IFRB: International (Radio) Frequency Registration Board
- BDT: Telecommunications Development Bureau
- General Secretariat

B.1.2.1 New ITU Structure

As the result of a reorganization in 1993, the ITU now consists of four committees [4]:

- *Telecommunications standardization sector* (ITU-T), which is involved in the standardization activities that were previously carried out by the CCITT and partially by the CCIR.
- *Radio communications sector* (ITU-R), which carries out the remaining standardization activities of the CCIR along with the previous work of the IFRB.
- *Development sector* (ITU-D), which has taken over the functions of the BDT.
- *ITU General Secretariat*, which supports the activities of these committees.

B.1.2.2 CCITT

This international organization was under the control of the national PTTs (*Post, Telephone and Telegraph*) until a few years ago. As a result of deregulation and privatization of the PTT in many countries, its role has increasingly been taken over by new national institutions and ministries.

The CCITT produces and publishes interfaces, services and protocols as technical recommendations for telephony, telegraphy and data communications.

For example, the CCITT produced the recommendation for transmission over large distances in open data networks. It also developed standards for cooperation between mobile land, see and air communications systems, as well as standards for linking these different mobile telephone services with the international telegraph and public telephone network.

Another aspect of CCITT's standardization work is focused on integrated systems for the simultaneous transmission of voice and data (ISDN). Voice transmission quality, switching technology and tariff issues relating to mobile radio and telephone are other activity areas.

Depending on the subject area, the work of the CCITT is carried out by the following *Study Groups* [3]:

- SG I Service Definitions
- SG II Network Operations
- SG III Tariff Principles
- SG IV Maintenance
- SG V Safety, Protection and EMC
- SG VI Outside Plant
- SG VII Dedicated Networks
- SG VIII Terminal for Telematic Services
- SG IX Telegraphs
- SG X Software
- SG XI ISDN, Network Switching and Signalling
- SG XII Transmission and Performance
- SG XV Transmission Systems and Equipment
- SG XVII Data Transmission
- SG XVIII ISDN and Digital Communications

The CCITT acts in a consulting capacity to the ITU.

B.1.2.3 CCIR

Like the CCITT, the CCIR is a consultative committee of the ITU in the area of radio technology; its principles and regulations have developed over the decades, and have gradually been adapted to cover new aspects, applications, situations and requirements.

The recommendations of CCIR mainly relate to the planning and coordination of radio communications and radio services, the technical characteristics of systems, and the effective and efficient use of frequency spectrum. The CCIR has drawn up criteria for the prevention of noisy interference, and is working together with the IRFB to ensure that these are applied and adhered to.

The *Interim Working Party* (IWP) was founded through a resolution of CCIR for the purpose of designing standards for future mobile radio systems.

The work of CCIR is divided into study groups, which deal with the following technical areas:

- SG 1 Spectrum Management Techniques
- SG 4 Fixed Satellite Services
- SG 5 Radio Wave Propagation in Non-Ionized Media
- SG 6 Radio Wave Propagation in Ionized Media
- SG 7 Science Services
- SG 8 Mobile, Radio Determination and Amateur Services
- SG 9 Fixed Services
- SG 10 Broadcasting Services—Sound
- SG 11 Broadcasting Services—Television
- SG 12 Inter-Service Sharing and Compatibility

The work is further divided into individual groups, which help to alleviate the responsibilities of the study groups. The most important group for wireless communication—the SG 8, which deals with issues relating to mobile and radio communication—is divided into the following work groups:

- Land Mobile Services
- Maritime Mobile Services
- Aeronautical Mobile Services
- Amateur Services
- Maritime Satellite Services
- Land Mobile Satellite Services
- Future Global Maritime Distress and Safety System (FGMDSS)

The *Commission Mixte CCIR/CCITT pour les Transmissions Televisuelles et Sonares* (CMTT) [2] deals with the interests of both organizations CCIR/CCITT.

B.1.2.4 International Frequency Registration Board, IFRB

After clarification of all technical issues in accordance with CCIR regulations, the IFRB administers all radio transmission of commercial, military and scientific origin, including amateur radio, which is carried on earth or over satellite. The IFRB regulates the allocation of frequencies worldwide and specifies the maximum number of radio channels in each frequency spectrum where interference could occur. Notification of the IFBR is required if a frequency:

- could cause damaging interference to the services of other administrations
- is to be used for international radio connections
- is to be internationally recognized

After it has been checked by the IFRB (mainly because of the possibility of interference), the frequency is entered in the *International Frequency Master File*.

Based on the IFBR frequency allocation, the world was divided into three regions:

- Region 1: Europe, Middle East, former USSR and Africa
- Region 2: Greenland, North and South America
- Region 3: Far East, Australia and New Zealand

WARC (*World Administrative Radio Conference*)—recently abbreviated to WRC—is a periodic conference at which applications specifying the use of frequency spectrum are deliberated and formally agreed upon by organizations from the three regions.

B.1.2.5 BDT

The *Board of Directors for Telecommunications* (BDT), founded in 1989, has the same status as CCITT and CCIR. Its task is to guarantee technical cooperation and to raise funds for financing the development of telecommunications networks in less industrialized countries.

B.1.2.6 General Secretariat

This committee is responsible for administration, finance, publications and technical recommendations.

B.1.2.7 Cooperation Between ISO and CCITT

There are areas such as *Open Systems Interconnection* (OSI) where there is a considerable overlap in the work carried out by the ISO and CCITT. This previously led to a race between the two organizations, which almost resulted in standardization efforts keeping pace with technical progress. The *Joint Technical Committee* (JTC) was founded to enable cooperation between the two bodies.

B.1.3 IEC

The IEC has been in existence since 1906 and, like the ISO and the ITU, is a UN institution. It develops standards in the electric and electronic component area as well as for operational safety and for the environmental conditions of products.

Its suborganization, the *Committee International Special Perturbations Radio* (CISPR), studies radio interference. In 1988, the ISO and the IEC jointly formed the *Joint Technical Committee* (JTCI), which is drawing up standards for information technology.

B.1.4 INTELSAT/INMARSAT

These organizations are responsible for the development, procurement and management of international satellite services. They are developing their own satellite radio standards.

The largest operator of a satellite communications system is INTELSAT with 200 earth stations in over 100 countries. It was set up in 1964 to deal with the many technical and administrative problems associated with worldwide satellite communications systems [6]. Starting with 14 countries, the membership number has been increasing continuously. By joining INTELSAT, a member state undertakes to seek approval before operating its own satellites.

Founded in 1980, the INMARSAT organization operates a global communications satellite system for maritime radio traffic, and offers telephone and telegraph services along with data transfer between coastal ground stations and ships and production platforms, which are always equipped with mobile transmitting and receiving stations [5]. All services offered by national PTTs are accessible over the INMARSAT network. In a sense, INMARSAT could be regarded as an extension of the PTT organizations in the field of satellite communication.

B.1.5 ATM Forum

The ATM Forum is a honorary international organization based in Mountain View, California, USA, which has the aim of promoting the use of Asyn-

chronous Transfer Mode (ATM) products and services. It promotes the interoperability of products, particularly through the specification of interfaces.

The Forum maintains close contacts with other industrial associations. It consists of a worldwide technical committee (TC), three marketing committees in North America, Asia and Europe, and the *Enterprise Network Roundtable* on the Internet, which allows end users to exchange information.

B.2 European Standards Organizations

The decision of the EU states within the framework of the Treaties of Rome (1983) to introduce the common European market in 1992, along with the emphasis on the important political, social and economic role to be played by telecommunications, has produced major changes in the European communications market since 1987 as a result of the publication of the so-called *Green Paper*.

Moreover, the EU states are directing, controlling and financing a large number of research programmes that, in addition to furthering their technical objectives, are also promoting cooperations between companies from different countries. The most important projects in the telecommunications area include: *Advanced Communication Technologies and Services* (ACTS), *Cooperation in the Field of Scientific and Technical Research* (COST) and *European Strategic Programme for Information Technology* (ESPRIT). Aside from a purely technical orientation, these programmes promote cooperation between the manufacturers of different countries, thereby playing an important role in the standardization process.

B.2.1 CEN/CENELEC

The *Comité Européene de Normalisation* (CEN) is the equivalent of the ISO in Europe, with the difference that the standards that it develops are binding.

Its sister organization, the *Comité Européene des Normalisation Electrotechniques* (CENELEC), develops standards in the electrical engineering area, and is therefore the European equivalent of the IEC. The standards that it develops with the label

- EN must be published and applied as national standards
- ENV can be applied voluntarily

B.2.2 CEPT

The *European Conference for Posts and Telecommunications* (CEPT), the members of which are the European PTTs (*Posts, Telephone and Telegraph*) and the public telephone network operators, was responsible for drawing up standards in the telephone, telegraph and data network area until 1988. Many

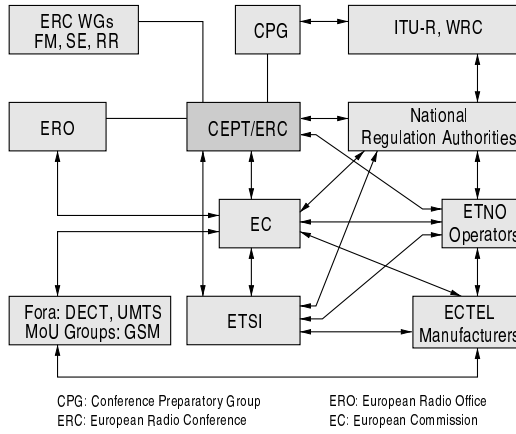


Figure B.1: European radio frequency planning coordinated by CEPT/ERC

of the recommendations it developed were taken over by the CCITT. A large proportion of CEPT's tasks were transferred to ETSI (*European Telecommunications Standards Institute*) when it was founded in 1988. CEPT is still involved in strategy and planning.

Three CEPT committees play a major role in the development of European standards:

Technical Recommendations Applications Committee (TRAC): This committee passes recommendations as *Normes Européennes de Télécommunication* (NET), which are binding for all PTT members.

Comité des Coordinations des Radiocommunications (CR): The CR's task is the development of stategies geared towards optimizing use of the frequency band and allocation of frequencies to the different services. The group CEPT/ERC (*European Radio Conference*) is responsible for the organizational aspects, and handles coordination between CEPT participants, with ETSI and on an international basis (see Figure B.1). CEPT/ERC corresponds to the international group ITU-T/WRC (*World Radio Conference*). According to Figure B.2, the *European Radio Office*, which has been part of CEPT since 1991, oversees the ongoing activities relating to frequency regulation and works together with the corresponding national organizations, e.g., RegTP in Germany.

Service and Facilities (SF) Committee: The SF Committee specifies service capabilities.

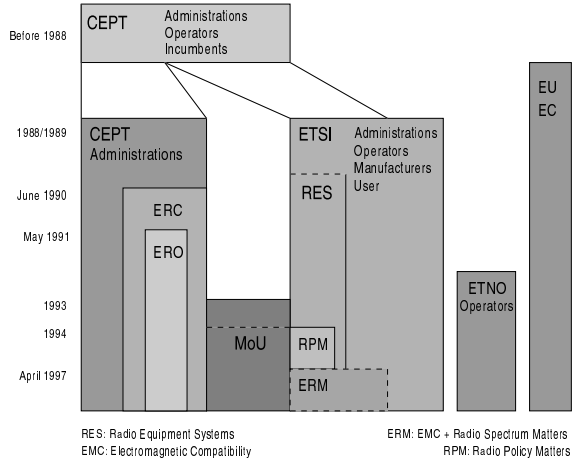


Figure B.2: CEPT, ERC and ETSI: Historical development

B.2.3 ETSI

The role of the *European Telecommunications Standards Institute* (ETSI), which was founded in 1988 in accordance with the requirements of the EU *Green Paper*, is similar to that of the CCITT, but with the main objective of developing telecommunications standards for Europe. ETSI membership is not only restricted to EU states but is open to all countries on the European continent.

ETSI recently began to include five categories of members with equal status, namely:

- National standards institutions and administrations
- Public network operators
- Manufacturers
- User groups
- Private service providers, research institutes and consultants

ETSI has thus also extended its membership to manufacturers, which means that standards in the future will be developed in close cooperation between network operators and manufacturing firms.

Another new development at ETSI is that project teams are being formed with experts secunded and paid by the member organizations and responsible for the design of standards.

To detect future trends in the market and in telecommunications, ETSI has set up *Special Strategic Review Committees of Senior Experts*, which study the ETSI programme and the need for new standards in certain areas.

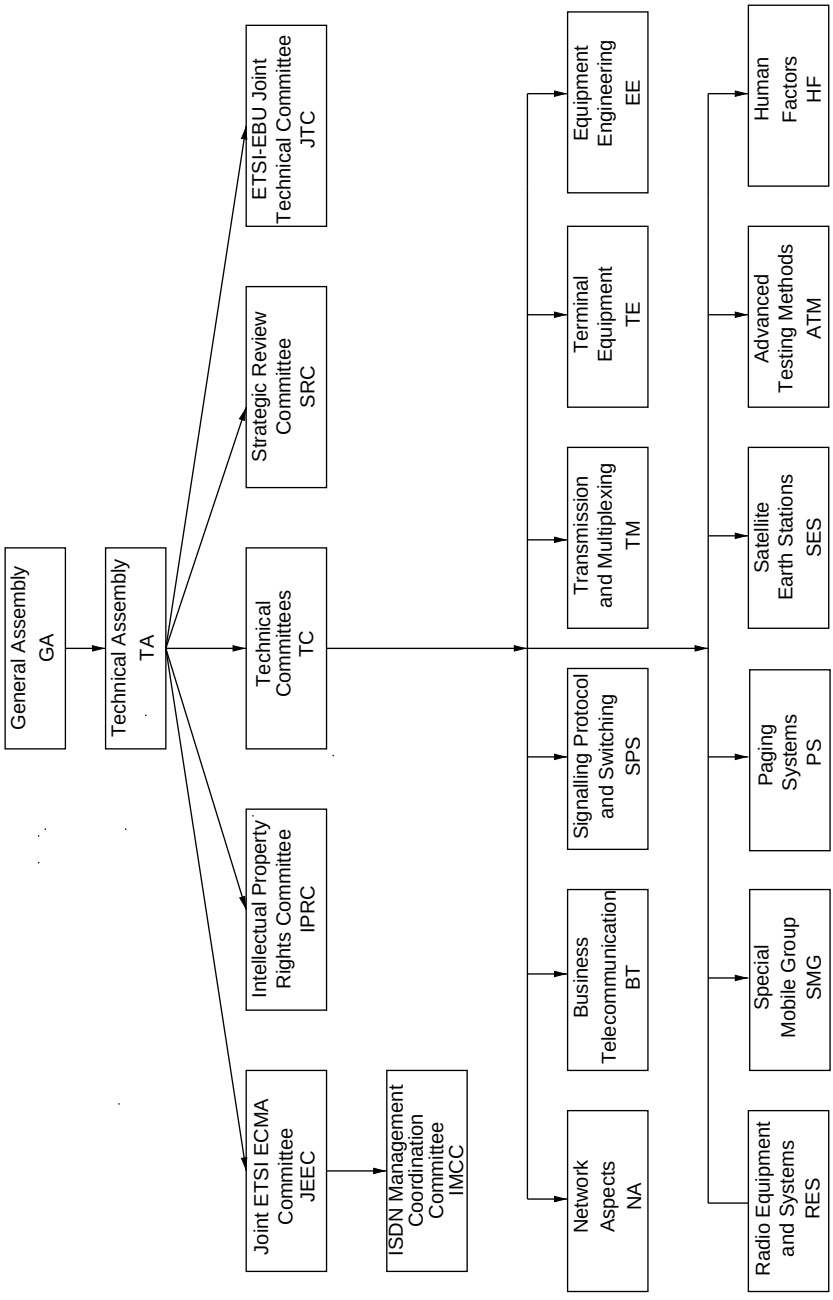


Figure B.3: ETSI

The former ETSI structure is illustrated in Figure B.3. The highest-ranking ETSI committee, the *General Assembly (GA)*, determines ETSI policies, elects new members and produces the budget. The administrative work is carried out by a secretariat based in Sophia-Antipolis, France, under the supervision of a director. The secretariat also oversees the work of the project teams (PT).

The *Technical Assembly* (TA) passes *European Telecommunications Standards* (ETS), which take effect with a majority vote. The TA appoints the *Technical Committees* (TC) and their chairmen. The standardization work is carried out by 12 TCs, which are divided into *Sub-Technical Committees* (STC). The Technical Subcommittees (STC) and the Project Teams (PT), which work under the direction of the respective Technical Committees (TC) (see Figure B.3), are listed in Table B.1.

Table B.1: Structure of the Technical Committees

Technical Committees			
NA	STC	NA 1	User Interfaces Services and Charging
		NA 2	Numbering, Addressing, Routing and Interworking
		NA 4	Network Architecture Operations, Maintenance Principles and Performance
		NA 5	Broadband Networks
		NA 6	Intelligent Networks
		NA 7	Universal Personal Telecommunications
BT	STC	BT 1	Private Networking Aspects
		BT 2	Business Telecommunications Network Performance
	PT	PT 27	PTN Attendant Services
		PT 43	PTM Mobility
		PT 18V	PSTN Access PABX
		PT 22V	ONP Leased Lines
SPS	STC	SPS 1	Network Interconnection and Signalling
		SPS 2	Signalling Network and Mobility Applications
		SPS 3	Digital Switching
		SPS 5	Customer Access to the ISDN
	PT	PT 21V	DSS 1 PICS/PIXIT
TM	STC	TM 1	Transmission Equipment Fibres and Cables
		TM 2	Transmission Networks Management, Performance and Protection
		TM 3	Architecture, Functional Requirements and Interfaces of Transmission Networks
		TM 4	Radio Relay Systems
		TM 5	Coding, Speech Processing and Associated Network Issues

Continued

Table B.1: Structure of the Technical Committees (continued)

Technical Committees			
TE	STC	TE 1	Videotex Systems
		TE 2	Text Communication Systems
		TE 3	MHS
		TE 4	Voice Terminals
		TE 5	General Terminal Access Requirements
		TE 6	Directory Systems
		TE 7	Lower Layer Terminal Protocols
		TE 8	Functional Standards
		TE 9	Card Terminal
		TE 10	Audio-Visual Management (AVM)
	PT	PT 34	Conformance Testing for Videophony
		PT 8V	Intelligent Caros
		PT 40	ISDN Terminals for ODA
		PT 17V	PSTN Access
		PT 19V	ISDN Programming Communications Interface
EE	STC	EE 1	Environment Conditions
		EE 2	Power Supply
		EE 3	Mechanical Structure
		EE 4	Electromagnetical Compatibility
HF	STC	HF 1	Telecommunication Services
		HF 2	People with Special Needs
		HF 3	Usability Evaluation
	PT	PT 16V	User Procedures for Videophony
		PT 36	Human Factor Guidelines for ISDN Equipment Design
RES	STC	RES 1	Maritime Mobile
		RES 2	Land Mobile
		RES 3	DECT
		RES 5	TFTS
		RES 6	Digital Trunking Systems (TETRA)
		RES 7	DSRR
		RES 8	Low Power Devices
		RES 9	EMC
		RES 10	Wireless LANs
	PT	PT 10	DECT
		PT 19	TFTS
		PT 29	Trunked Mobile Systems
		PT 41	Radio LANs
SMG	STC	SMG 1	Services and Facilities

Continued

Table B.1: Structure of the Technical Committees (continued)

Technical Committees			
		SMG 2	Radio Aspects
		SMG 3	Network Aspects
		SMG 4	Data and Telematic Services
		SMG 5	UMTS
		SMG 6	Network Management Aspects
	PT	PT 12	Pan European Cellular Digital Radio Systems
		PT 12V	DCS1800 (PCN)
PS	STC	PS 2	ERMES Radio Aspects
		PS 3	ERMES Network Aspects
SES	STC	SES 1	General Systems Requirements
		SES 2	RF and IF Equipment
		SES 3	Interconnections to Terrestrial Networks Control and Monitoring Functions
		SES 4	TV and Sound Programmes Equipment
		SES 5	E.S for Mobile Services
	PT	PT 42	VSAT. ISDN
		PT 15V	Mobile Earth Stations for LMSS and RDSS
		PT 23V	News Gathering Earth Stations
ATM	STC	ATM 1	Conformance Testing Methodologies
		ATM 2	Conformance Testing Environment
	PT	PT 31	TTCN SDL
		PT 37	SDL Guide
		PT 38	Test Specification Handbook
		PT 39	Conformance Testing Consulting Group

ETSI has assigned a high priority to the mobile radio area. It has provided the specifications for systems such as GSM, DCS 1800, DECT, ERMES, TETRA and TFTS. The following committees are important in the mobile radio area:

- RES: Radio Equipment and Systems
- SMG: Special Mobile Group
- PS: Paging Systems
- SES: Satellite Earth Stations

ETSI is drawing up three types of standards:

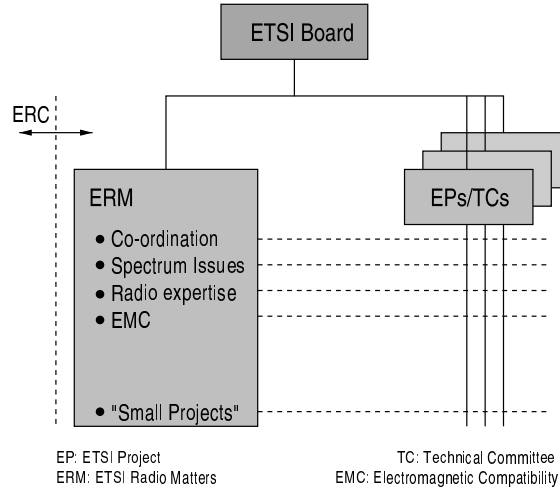


Figure B.4: Matrix organization for dealing with radio-related matters

- EN or ENV: European Norm
- ETS: European Telecommunications Standards
- NETS: Normes Européennes de Télécommunications

Owing to the overlap in the work being carried out by ETSI, CEN and CENELEC, the *Information Technology Steering Committee* (ITSTC) was set up to assume a coordinating role. The *European Radio Office* (ERO), with its headquarters in Copenhagen, Denmark, was established for the purposes of coordinating European frequency use.

Changes to the ETSI Organization

The reorganization implemented in 1997 had a major influence on the standardization process and associated documents. A brief overview of the old and new types of documents and the corresponding approval procedures is presented below. The *Radio Policy Matters* (RPM) group (see Figure B.2) has existed since 1994 and the *ETSI Radio Matters* ERM group since 1997. CEPT/ERC and ETSI work together in accordance with MoU 1994; RPM coordinates this cooperation on the ETSI side. The ERM group can be seen as a "horizontal technical committee" with responsibility for the coordination between different mobile radio systems. In future every mobile radio system will be dealt with in an ETSI project (EP) (see Figure B.4). Future cooperation between CEN/ERC and ETSI/ERM will be organized as shown in Figure B.5.

The labeling of the documents has changed as follows:

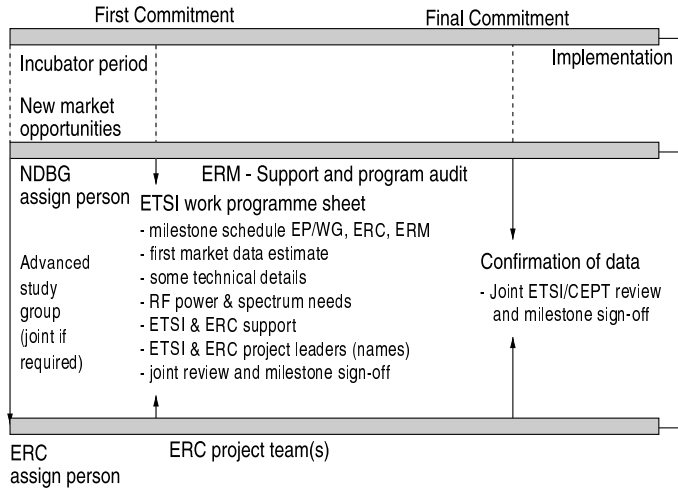


Figure B.5: Joint CEPT/ERC-ETSI implementation process

Old

ETSI Technical Report, ETR This provides an overview of the status of technology and/or requirements for a specific system.

ETSI Technical Specification, ETS This is the actual standard. It was also produced by STC and then presented to the *Public Enquiry* that incorporated the teams from the different countries who were involved in the standardization process.

New

Technical Report, TR This is similar to the reports previously prepared by ETR. It was released by the respective committee, and in that sense is more of a working paper.

Technical Specification, TS This is comparable to the previous ETS, but is not a standard because it is only released by the responsible committee.

ETSI Standard, ETS: This is a new type of paper comparable in content to the previous ETS. This is a de facto standard, which is ratified through an abbreviated procedure. According to the procedure, ETSI members are apprised of the standard (by a single contact person) over the Internet, and have 60 days in which to give their vote over the Internet. The experts from the different countries who were involved in the standardization work are not included.

ETSI Guide, EG The content is the same as that of TR, and the confirmation procedure is the same as with ETS.

Euro Norm, EN This is the de jure standard for which the national organizations are included in the ratification. The enquiry (and voting) procedure has also been shortened and now takes either 17 or 23 days.

B.2.4 ECMA

The *European Computer Manufacturers Associations* (ECMA) was established in 1960. Its members include approximately 20 leading computer manufacturers who on a volunteer basis develop standards in the data communications area, in particular *Open System Interconnection* (OSI), and the recently developed signalling system QSIG for private networks.

The most important ECMA committee is the *Communications, Networks and Interconnection* (TC32) Committee, which is divided into four working groups:

- TG11: Computer-Supported Telecommunications Applications
- TG12: PTN Managment
- TG13: PTN Networking
- TG14: PTN Signalling

The *Joint ETSI ECMA Committee* (JEEC) was founded in 1991 to coordinate standardization activities because of the overlapping of ETSI and ECMA in the area of private network standards. ETSI's work is directed towards the development of standards for interactive connections between public and private networks, whereas ECMA develops the standards for private networks. Information on ongoing work is exchanged between the two organizations. Furthermore, an opportunity exists for members of one organization to work with members of the other. The standards completed by ECMA are presented to ETSI and voted on for approval as an ETS.

B.2.5 EBU

The *European Broadcasting Union* (EBU), which is based in Geneva, is an association of European television and broadcasting corporations. The technical committee of EBU works on recommendations for norms and standards, such as the satellite transmission norm for television and broadcasting.

B.2.6 EUTELSAT

The *European Telecommunications Satellite Organization* (EUTELSAT) was founded in 1982 in accordance with an international agreement between 28 member states of CEPT. As a result of an agreement signed in 1983, the regional operating company EUTELSAT was put on the same legal footing as

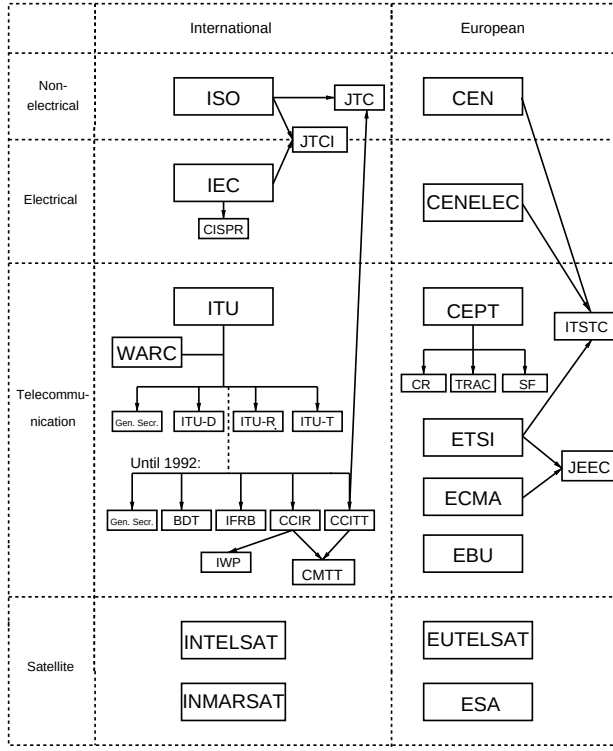


Figure B.6: Cooperation between international and European standards bodies

INTELSAT. The responsibilities of EUTELSAT include managing the *European Communication Satellites* (ECS) satellite system. EUTELSAT satellites are used to transmit telephone conversations, radiate television programmes, transfer data and provide teleconferencing.

B.2.7 ESA

Established in 1975, the *European Space Agency* (ESA) is also active in European satellite communications. The predecessors of ESA were the *European Space Research Organization* (ESRO) and the *European Launch Development Organization* (ELDO), along with the *Conferènce Européen pour des Télécommunications par Satellite* (CETS) [5].

ESA promotes cooperation between European countries in space research and technology, and develops operational space applications systems. An overview of international and European Standards organizations and their cooperations is presented in Figure B.6.

Table B.2: National standards organizations and professional associations

Country	Organization	
Germany	RegTP	Regulation Authority for Telecommunications and Post
	DIN	German Standards Institute
	VDE	Federation of German Electrical Engineers
	VDI	Federation of German Engineers
	DKE	German Commission on Electrical Engineering
Great Britain	MPT	Ministry of Post and Telecommunications
	BSI	British Standards Institute
	OFTEL	Office of Telecommunications
	BABT	British Approvals Board for Telecommunications
France	AFNOR	
USA	ANSI	American National Standards Institute
	IEEE	Institute of Electrical and Electronics Engineers
	EIA	Electronic Industries Associations
	TIA	Telecommunication Industries Associations
	SME	The Society of Manufacturing Engineering
	T1	Exchange Carriers Standards Association
Japan	TTC	The Telecommunications Technology Committee

B.3 National Standards Organizations

National standardization facilities, organizations and professional associations concern themselves mainly with the specification of national standards, cooperation with international standards bodies and the translation of international standards into the local language. The best-known facilities are listed in Table B.2.

B.4 Quasi-Standards

B.4.1 Company Standards

Despite the numerous standards organizations in the world, many companies develop their own specifications, which can sometimes become de facto standards. These de facto standards are either accepted by the international and the national organizations or become serious competition to them for a long time.

Company standards are not published, and therefore play a major role in competition. Anyone who wants to use these standards in their own systems must apply for a licence. Examples of companies that have developed their own standards include:

- Motorola: e.g., RD-LAP protocol, IRIDIUM, Altair WIN 802.11
- Ericson: Mobitex
- Qualcomm: CDMA (IS-95)

B.4.2 User Standards

In addition to companies, other organizations or institutions such as universities, the military, the police and railways also develop their own de facto standards. These standards that are designed for internal use are called user standards. If there is an interest, they also can be used by outside users in their own systems.

The standards in this category include *Mobile IP*, which is a further development of TCP/IP (*Transmission Control Protocol/Internet Protocol*) of the Internet, which was developed by the *Defense Advanced Research Projects Agency* (DARPA).

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Appendix C
International Frequency Allocations

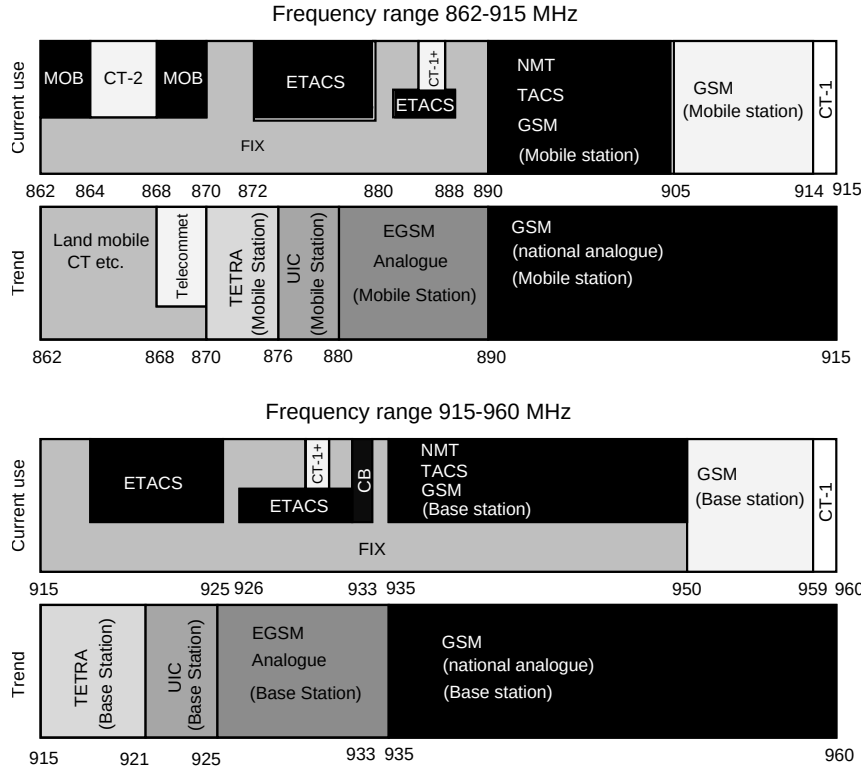


Figure C.1: Allocation of frequencies in accordance with Article 8 of the *Radio Regulations*

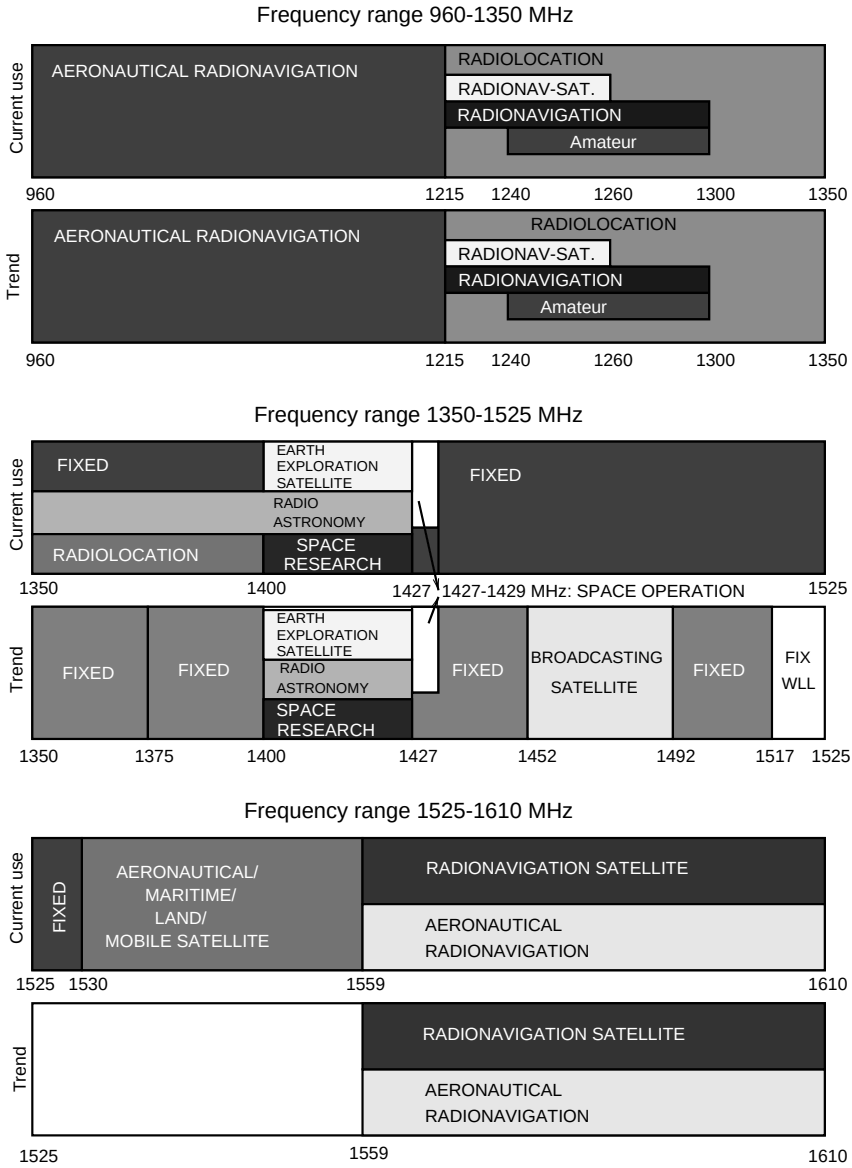


Figure C.2: Allocation of frequencies in accordance with Article 8 of the *Radio Regulations*

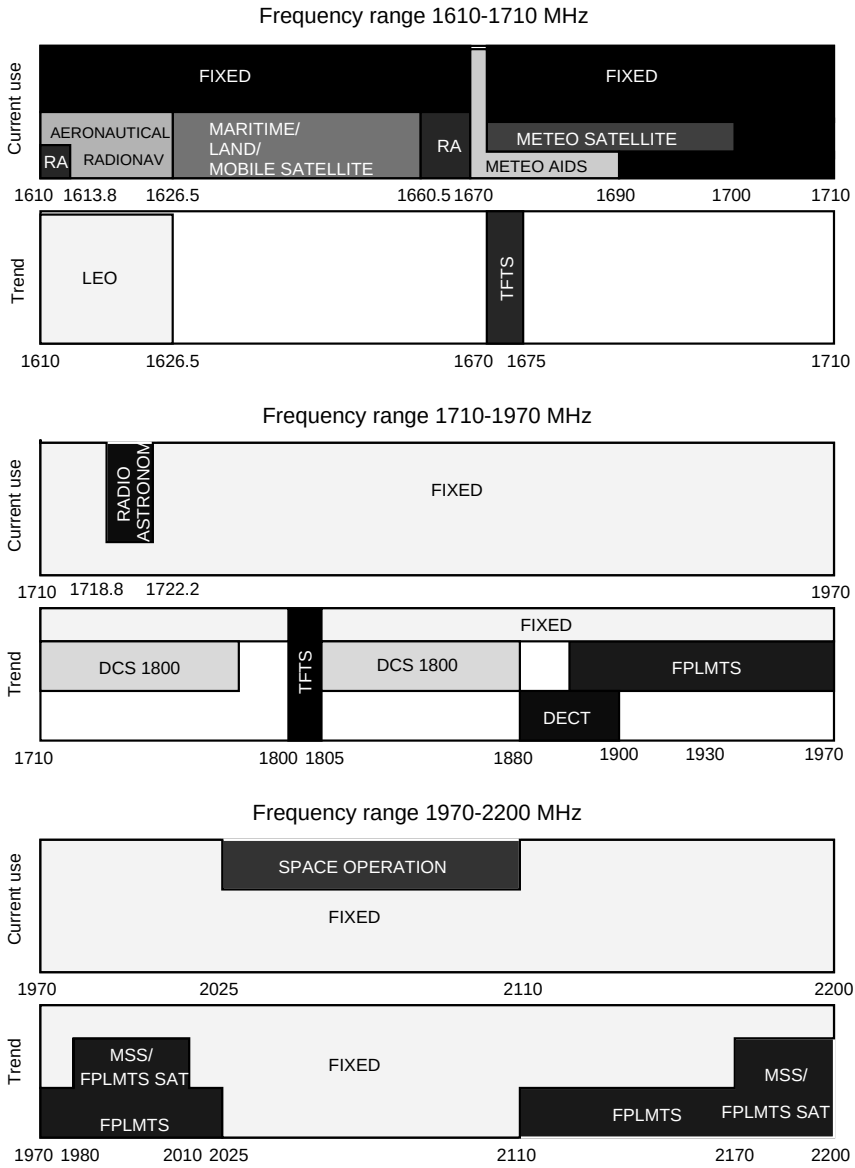


Figure C.3: Allocation of frequencies in accordance with Article 8 of the *Radio Regulations*

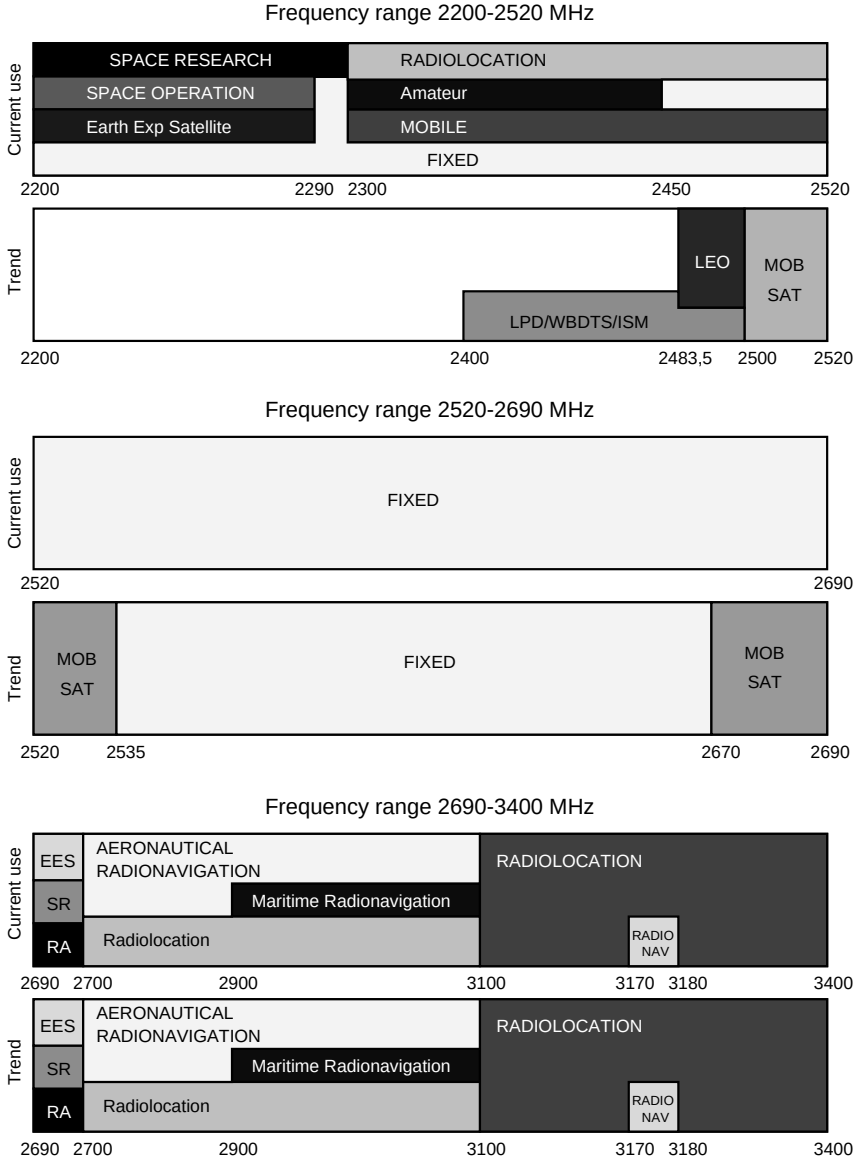


Figure C.4: Allocation of frequencies in accordance with Article 8 of the *Radio Regulations*

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Appendix D
The Frequencies of European Mobile Radio Systems

The latest data is available from the European Radio Office (ERO) under `http://www.ero.dk` (see Technical Report 25).

System	Freq. (lower band) [MHz]			Freq. (upper band) [MHz]		
Cordless telephones						
CT1	914	...	915	959	...	960
CT1+	885	...	887	930	...	932
CT2	864.1	...	868.1			
DECT	1880	...	1900			
Digital trunked radio (TETRA)						
Frequency ranges based on CEPT recommendation, from which frequencies are to be selected on a national basis.						
	380	...	400			
	410	...	430			
	450	...	470			
	870	...	888	915	...	923
Cellular mobile radio systems						
GSM	890	...	915	935	...	960
Extension band GSM	880	...	890	925	...	935
UIC (European orbits)	870	...	874	915	...	919
DCS1800	1710	...	1785	1805	...	1880
FPLMTS (UMTS)	1885	...	2010	2110	...	2200
FPLMTS satellite band	1980	...	2010	2170	...	2200
Radio paging systems						
EUROSIGNAL (D, F, CH)	87.340					
	87.365					
	87.390					
	87.415					

Continued

System	Freq. (lower band) [MHz]	Freq. (upper band) [MHz]
Cityruf (D only)	465.970	
Euromessage	466.075	in D, F, I, GB
	466.230	
ERMES	169.4 ... 169.8	
	Terrestrial flight telephone system	
TFTS	1670 ... 1675	1800 ... 1805
	LEO satellite systems	
	1610 ... 1626.5	2483.5 ... 2500
	INMARSAT systems	
Only sections of	1525 ... 1559	1626.5 ... 1660.5

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Appendix E
The GSM Standard

Table E.1 provides an overview of the GSM standard. The standards for GSM1800 appear with the abbreviation DCS (Digital Communications System). The GSM/GPRS standards are listed seperately in Table E.2.

Table E.1: The GSM standard

GSM No.	GSM Title
01.02	General Description of a GSM PLMN
01.04 REP	Vocabulary in a GSM PLMN
01.06	Service Implementation Phases and Possible Further Phases in the GSM PLMN
01.78	Requirements for the CAMEL Feature
02.01	Principles of Telecommunication Services by a GSM PLMN
02.02	Bearer Services Supported by a GSM PLMN
02.03	Teleservices Supported by a GSM PLMN
02.04	General on Supplementary Services
02.05	Simultaneous and Alternate Use of Services
02.06	Types of Mobile Stations
02.06 DCS	Types of Mobile Stations
02.07	Mobile Station Features
02.08 REP	Report: Quality of Service
02.09	Security Aspects
02.10	Provision of Telecommunications Services
02.11	Service Accessibility
02.11 DCS	Service Accessibility
02.12	Licensing
02.13	Subscription to the Services of a GSM PLMN
02.14	Service Directory
02.15	Circulation of Mobile Stations
02.16	International MS Equipment Identities
02.17	Subscriber Identity Modules, Functional Characteristics
02.20	Collection Charges
02.24	Description of Advice of Charge
02.30	Man–Machine Interface of the Mobile Station
02.40	Procedures for Call Progress Indications
02.78	CAMEL Service Definition (Stage 1)
02.81	Number Identification Supplementary Services
02.82	Call Offering Supplementary Services

Continued

Table E.1: The GSM standard (continued)

GSM No.	GSM Title
02.83	Call Completion Supplementary Services
02.84	Multi-Party Supplementary Services
02.85	Community of Interest Supplementary Services
02.86	Charging Supplementary Services
02.87	Additional Information Transfer Supplementary Services
02.88	Call Restriction Supplementary Services
03.01	Network Functions
03.02	Network Architecture
03.03	Numbering, Addressing and Identification
03.04	Signalling Requirements Relating to Routing of Calls to Mobile Subscriber
03.05	Technical Performance Objectives
03.07	Restoration Procedures
03.08	Organization of Subscriber Data
03.09	Handover Procedures
03.10	GSM PLMN Connection Types
03.11	Technical Realization of Supplementary Services—General Aspects
03.12	Location Registration Procedures
03.12 DCS	Location Registration Procedures
03.13	Discontinuous Reception (DRX) in the GSM System
03.14	Support of DTMF via the GSM System
03.20	Security-Related Network Functions [A]
03.38	Alphabets and Language-Specific Information
03.40	Technical Realization of the Short Message Service Point-to-Point
03.41	Technical Realization of Short Message Service Cell Broadcast
03.42 REP	Report: Technical Realization of Advanced Data MHS Access
03.43	Technical Realization of Videotex
03.44	Support of Teletex in a GSM PLMN
03.45	Technical Realization of Facsimile Group 3 Service—Transparent
03.46	Technical Realization of Facsimile Group 3 Service—Non-Transparent
03.48 REP	Report: GSM Short Message Service—Cell Broadcast
03.50	Transmission Planning Aspects of the Speech Service in the GSM PLMN System
03.70	Routing of Calls to/from PDNs
03.78	Customized Applications for Mobile Network Enhanced Logic (CAMEL)
03.81	Technical Realization of Line Identification Supplementary Services
03.82	Technical Realization of Call Offering Supplementary Services

Continued

Table E.1: The GSM standard (continued)

GSM No.	GSM Title
03.83	Technical Realization of Call Completion Supplementary Services
03.84	Multi-Party Supplementary Services
03.86	Technical Realization of Charging Supplementary Services
03.88	Technical Realization of Call Restriction Supplementary Services
03.90	Unstructured Supplementary Service Data (USSD—Stage 2)
04.01	MS-BSS Interface—General Aspects and Principles
04.02	GSM PLMN Access Reference Configuration
04.03	MS-BSS Interface: Channel Structures and Access Capabilities
04.04	MS-BSS Layer 1—General Requirements
04.05	MS-BSS Data Link Layer—General Aspects
04.06	MS-BSS Data Link Layer Specification
04.07	Mobile Radio Interface Signalling Layer 3—General Aspects
04.08	Mobile Radio Interface—Layer 3 Specification
04.08 DCS	Mobile Radio Interface—Layer 3 Specification
04.10	Mobile Radio Interface Layer 3—Supplementary Services Specification—General Aspects
04.11	Point-to-Point Short Message Service Support on Mobile Radio Interface
04.12	Cell Broadcast Short Message Service Support on Mobile Radio Interface
04.21	Rate Adaption on MS–BSS Interface
04.22	Radio Link Protocol for Data and Telematic Services on the MS-BSS Interface
04.80	Mobile Radio Interface Layer 3—SS Specification—Formats and Coding
04.82	Mobile Radio Interface Layer 3—Call Offering SS Specification
04.88	Mobile Radio Interface Layer 3—Call Restriction SS Specification
04.90	Unstructured Supplementary Service Data (USSD)—Stage 3
05.01	Physical Layer on the Radio Path (General Description)
05.01 DCS	Physical Layer on the Radio Path (General Description)
05.02	Multiplexing and Multiple Access on the Radio Path
05.03	Channel Coding
05.04	Modulation
05.05	Radio Transmission and Reception
05.05 DCS	Radio Transmission and Reception
05.08	Radio Subsystem Link Control
05.08 DCS	Radio Subsystem Link Control
05.10	Radio Subsystem Synchronization
06.01	Speech Processing Functions: General Description
06.10	GSM Full-Rate Speech Transcoding

Continued

Table E.1: The GSM standard (continued)

GSM No.	GSM Title
06.11	Substitution and Muting of Lost Frames for Full-Rate Speech Traffic Channels
06.12	Comfort Noise Aspects for Full-Rate Speech Traffic Channels
06.31	Discontinuous Transmission (DTX) for Full-Rate Speech Traffic Channels
06.32	Voice Activity Detection
07.01	General on Terminal Adaptation Functions for MSs
07.02	Terminal Adaptation Functions for Services Using Asynchronous Bearer Capabilities
07.03	Terminal Adaptation Functions for Services Using Synchronous Bearer Capabilities
08.01	General Aspects on the BSS–MS Interface
08.02	BSS-MS Interface—Interface Principles
08.04	BSS-MSC Layer 1 Specification
08.06	Signalling Transport Mechanism for the BSS-MSC
08.08	BSS-MSC Layer 3 Specification
08.09	Network Management Signalling Support Related to BSS
08.20	Rate Adaption on the BSS–MSC Interface
08.51	BSC-BTS Interface, General Aspects
08.52	BSC-BTS Interface Principles
08.54	BSC-TRX Layer 1: Structure of Physical Circuit
08.56	BSC-BTS Layer 2 Specification
08.58	BSC-BTS Layer 3 Specification
08.58 DCS	BSC-BTS Layer 3 Specification
08.59	BSC-BTS C
08.60	Inband Control of Remote Transcoders and Rate Adaptors
09.01	General Network Interworking Scenarios
09.02	Mobile Application Part Specification
09.02 DCS	Mobile Application Part Specification
09.03	Requirements on Interworking Between the ISDN or PSIN and the PLMN
09.04	Interworking Between the PLMN and the CSPDN
09.05	Interworking Between the PLMN and the PSPDN for PAD Access
09.06	Interworking Between the PLMN and a PSPDN/ISDN for Support of Packet Switched Data
09.07	General Requirements on Interworking Between the PLMN and the ISDN or PSTN
09.09 REP	Detailed Signalling Interworking within the PLMN and with the PSIN/ISDN
09.10	Information Element Mapping Between MS-BS/BSS-MSC Signalling Procedures and MAP

Continued

Table E.1: The GSM standard (continued)

GSM No.	GSM Title
09.10 DCS	Information Element Mapping Between MS-BS/BSS-MSC Signalling Procedures and MAP
09.11	Signalling Interworking for Supplementary Services
11.10	Mobile Station Conformity Specifications
11.11	Specification of the SIM-ME Interface
11.11 DCS	Specification of the SIM-ME Interface
11.20	The GSM Base Station System: Equipment Specification
11.30 REP	Mobile Services Switching Centre
11.31 REP	Home Location Register Specification
11.32 REP	Visitor Location Register Specification
11.40	System Simulator Specification
12.00	Objectives and Structure of Network Managment
12.01	Common Aspects of GSM Network Managment
12.02	Subscriber, Mobile Equipment and Services Data Administration
12.03	Security Management
12.04	Performance Data Measurements
12.05	Subscriber Related Event and Call Data
12.06	GSM Network Change Control
12.07	Operations and Performance Management
12.10	Maintenance Provisions for Operational Integrity of MSs
12.11	Maintenance of the Base Station System
12.13	Maintenance of the Mobile-Services Switching Centre
12.14	Maintenance of Location Registers
12.20	Network Management Procedures and Messages
12.21	Network Management Procedures and Messages on the A_{bis}-Interface

Table E.2: The GPRS standards: an overview

Stage	GSM No.	GSM Title
1	02.60	GPRS Overview
2	03.60	System Description
	03.61	PTMM Service
	03.62	PTMG Service
	03.64	Radio Architecture
3 (new)	04.60	RLC/MAC Protocol (U_m-Interface)
	04.61	PTM-M Protocol
	04.62	PTM-G Protocol
	04.64	LAPG Protocol (MS-SGSN)
	04.65	SNDCP (MS-SGSN)
	07.60	User Interworking
	08.64	BSSGP (G_b-Interface)
	09.16	Network Service (G_s-Interface)
	09.18	Layer-3 Protocol (G_s-Interface)
	09.60	GTP (G_n-Interface)
	09.61	Network Interworking (G_i-Interface)
3 (modified)	03.20	Security Aspects
	04.03–04.07	System Scheduling
	04.08	Mobility Management
	05.xx	Radio Interface
	08.06	TRAU Frame Mod.
	08.08	BSSGP at G_b
	08.20	Physical Layer at G_b
	09.02	MAP at G_r and G_d
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	12.xx	O & M Additions

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