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GSM/EDGE Standards Evolution (up to Rel'4)

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The standardisation work of GSM-based systems has its roots in the 1980s, when a standardisation body 'Groupe Special Mobile' (GSM)[1] was created within the Conference Europeenne des Postes et Telecommunications (CEPT), whose task was to develop a unique digital radio communication system for Europe, at 900 MHz.

Since the early days of GSM development, the system has experienced extensive modifications in several steps to fulfil the increasing demand from the operators and cellular users. The main part of the basic GSM system development during the last decade until spring 2000 has been conducted in European Telecommunications Standards Institute (ETSI) Special Mobile Group (SMG) and its technical sub-committees, as well as in T1P1, which had the responsibility of the PCS1900 MHz specifications in the United States.

Currently, the further evolution of the GSM-based systems is handled under the 3rd Generation Partnership Project (3GPP), which is a joint effort of several standardisation organisations around the world to define a global third-generation UMTS (universal mobile communication system) cellular system. The main components of this system are the UMTS terrestrial radio access network (UTRAN) based on wideband code division multiple access (WCDMA) radio technology and GSM/EDGE radio access network (GERAN) based on global system for mobile communications (GSM)/enhanced data rates for global evolution (EDGE) radio technology.

In the following sections, an introduction of the evolution steps of the GSM specifications is presented.

[1] The original GSM acronym is 'Groupe Special Mobile'. This changed afterwards to Global System for Mobile communication, which is the current official acronym.

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1.1 Standardisation of GSM—Phased Approach

The need for continuous development of the GSM specifications was anticipated at the beginning of the specification work, which was, as a consequence, split into two phases. This phased approach was defined to make sure that specifications supported a consistent set of features and services for multi-vendor operation of the GSM products, both on the terminal and the network sides.

The GSM Phase 1 work included most common services to enable as fast as possible deployment of GSM in operating networks still providing a clear technological advance compared with existing analogue networks. The main features in Phase 1 include support for basic telephony, emergency calls, 300 to 9600 kbps data service, ciphering and authentication as well as supplementary services like call forwarding/barring. Also, short message service (SMS) was also included at this early phase of the specification work, although its commercial success came much later. From a radio performance point of view, frequency hopping, power control and discontinuous transmission were terminal mandatory from GSM Phase 1.

While the GSM Phase 1 networks were being built, GSM Phase 2 was being specified in ETSI SMG. The GSM Phase 2 specifications were frozen in October 1995, and they included a mechanism for cross-phase compatibility and error handling to enable evolution of the specifications. Many technical improvements were also introduced and included several new supplementary services like line identification services, call waiting, call hold, advice of charge and multi-party call. In the speech area, a half-rate channel mode codec was introduced to complement the already specified speech codec for GSM full-rate channel mode. In the data side, the main improvement included group 3 fax service support.

The first two phases of GSM provided a solid basis for the GSM system evolution towards the third-generation (3G) system requirements and items, which were, at the beginning of the work, better known as Phase 2+ items. In the core network (CN) side, the evolution led to the introduction of general packet radio system (GPRS) network architecture, especially designed for Internet connectivity. Similarly, the radio access network experienced significant enhancements in both packet and circuit modes to provide higher bit rates and better network capacity, in both packet and circuit-switched modes. From the users' perspective, in addition to better speech quality, this evolution consisted of significant enhancements to the capabilities to provide speech and data services. The following list gives an overview of the different work item categories:

- new bearer services and data-related improvements like high-speed circuit-switched (multislot) data (HSCSD), 14.4-kbps (single-slot) data, general packet radio service (GPRS) and enhanced data rates for global evolution (EDGE);
- speech-related items like enhanced full-rate (EFR) speech codec, adaptive multi-rate codec (AMR) with both narrow and wideband options, and tandem free operation (TFO) of speech codecs;
- mobile station (MS) positioning–related items like cell identity and timing advance, uplink time of arrival (TOA) and enhanced observed time difference (E-OTD) methods based on measurements within the cellular network, and assisted GPS method based on the GPS (global positioning system) technology;

- frequency band–related items like GSM400, 700, and 850 MHz, unrestricted GSM multi-band operation, release independent support of frequency bands;
- messaging-related items like SMS concatenation, extension to alphabet, SMS interworking extensions, forwarding of SMSs and particularly multimedia messaging (MMS);
- new supplementary services like call deflection, calling-name presentation, explicit call transfer, user-to-user signalling, completion of calls to busy subscriber and new barring services;
- billing-related work items like payphone services, provision for hot billing and support for home area priority;
- service platforms like subscriber identity module (SIM) application toolkit and customised application for mobile network enhanced logic (CAMEL).

Although the phasing is used also when developing 3G standards, grouping into 'Releases' (later in this book referred to as Rel') is a more practical tool, currently used in the standardisation to manage changes between different specification versions and introduction of new features. New releases are normally produced every year, and they contain a set of all specifications having added functionality introduced to it compared with previous releases as a result of ongoing standardisation work [1].

1.1.1 GSM/TDMA Convergence through EDGE

Standardisation of EDGE was the main cornerstone for integrating the two major time division multiple access (TDMA) standards, GSM specified in ETSI and IS-136 specified in Telecommunications Industry Alliance/Electronic Industries Alliance (TIA/EIA) on the same evolution path. At the moment, new frequency bands are not available in the United States, so an evolutionary approach to the new services on existing frequency bands is important for the current US operators.

TIA/EIA-136 is a result of the evolution of the US cellular networks. While GSM was digital from the beginning, TIA/EIA-136 has its roots in analogue advanced mobile phone service (AMPS EIA-553), which has been digitised in stages. The first step was the introduction of digital traffic channels (TCHs) in the IS-54 standard version. The IS-136 releases introduced digital signalling channels, improved voice quality with an enhanced voice codec and standardised dual-band operation. Like in Europe, the next wave in complementing the capabilities of TIA/EIA-136 in the United States was targeted in finding suitable 3G high data rate solution.

The start of EDGE development goes back to 1997, when ETSI conducted a feasibility study on improved data rates for GSM evolution. At the same time, the Universal Wireless Communication Consortium's (UWCC) Global TDMA Forum prepared input for TIA's ITU IMT-2000 programme. In 1998, key technical parameters of these two developments were harmonised, forming a basis for a converged TDMA standard using 200-kHz carrier and octagonal phase shift keying (8-PSK) modulation–based radio interface.

In Europe, the work continued under ETSI SMG2's EDGE work items and the first-phase EDGE standard, including enhanced circuit-switched data (ECSD) based on HSCSD, and enhanced general packet radio service (EGPRS) based on GPRS, was finalised in spring 2000. In addition, the specifications supported an optional arrangement called *EDGE Compact*, which enabled the deployment of EDGE in as narrow as 1-MHz frequency bands.

In the United States, the work continued in TIA TR45 as further development of EDGE as part of the UWC-136 IMT-2000 proposal. In November 1999, the UWC-136 proposal was approved as a radio interface specification for IMT-2000. Currently, EDGE is part of the GERAN development that is carried out concurrently in 3GPP and UWCC to ensure high synergy between GSM and TDMA systems in the future.

The latest developments in the US market from the two largest IS-136 operators, AT&T and Cingular, define the mainstream evolution path likely to be followed by most IS-136 operators. IS-136 networks will fully migrate to GSM/EDGE/WCDMA UMTS technology making use, in some cases, of the GAIT (GSM/ANSI-136 Interoperability Team) functionality to ensure a smooth migration [2]. As a result of this, EDGE Compact is not likely to ever be implemented, since the evolution timing and the roaming requirements do not make this option interesting for operators.

1.1.2 GERAN Standardisation in 3GPP

In summer 2000, the specification work of GSM radio was moved from ETSI SMG2 to 3GPP, which is a project responsible of the UMTS standards based on the evolved GSM core network. This meant that Rel'99 was the last release of the GSM/EDGE standard that was specified in ETSI SMG, and all specifications related to GSM radio access network were moved under 3GPP responsibility with new specification numbers.

The moving of the specifications was motivated by both the need for improved working procedures and technical development closer to 3GPP. The core network part of the GSM specifications was transferred to 3GPP by the time the project work was initiated, and the arrangement of standardising the GSM network and radio access parts in two different standardisation bodies made it too cumbersome to work effectively. More importantly, activities aimed at closer integration of GSM/EDGE radio and WCDMA technologies had led to a decision to adopt the 3GPP-specific Iu interface for GERAN. Thereby, this integration achieves a true multi-radio UMTS standard, which is made up of two radio technologies, WCDMA and GSM/EDGE, that can be effectively and seamlessly integrated in order to maximise the efficiency and the quality of service provided to the end users with the introduction of the coming 3G multimedia services.

1.1.2.1 3GPP Organisation

The 3GPP is a joint effort of several standardisation organisations around the world targeting to produce specifications for a global 3G cellular system. These so-called organisational partners (OPs) have the authority to develop standards in their region, and they currently include ETSI in Europe, Association of Radio Industries and Business (ARIB) in Japan, Standardisation Committee T1—Telecommunications (T1) in the United States, Telecommunication Technology Association (TTA) in Korea, China Wireless Telecommunication Standard group (CWTS) in China and Telecommunications Technology Committee (TTC) in Japan.

3GPP also have market representation partners (MRP), whose role is to make sure that the standardisation is in line with the market requirements. In practice, the MRPs work together with OPs in the Project Coordination Group (PCG) to guide the overall direction of the 3GPP work. Potential future partners can also follow the work as observers.

More information and up-to-date links about the 3GPP-related activities and organisation can be found in http://www.3gpp.org/.

Most of the technical work in 3GPP is carried out by individual members, typically 3G operators and vendors, in technical specification groups (TSGs) and their working groups (WGs). The basic 3GPP work organisation is depicted in Figure 1.1.

Currently there are five TSGs in 3GPP:

- *TSG SA*—Service and System Aspects, taking care of the overall system architecture and service capabilities, and cross TSG coordination on these areas. Issues related to security, speech and multimedia codecs, network management and charging are also handled in SA.
- *TSG CN*—Core Network, taking care of the UMTS/GSM core network protocols. This includes the handling of layer 3 call control, mobility management and session management protocols between the mobile terminal and CN, as well as inter-working with external networks.
- *TSG T*—Terminals, taking care of the aspects related to the terminal interfaces. This includes the responsibility of issues like universal subscriber identity module (USIM), terminal equipment (TE) performance, service capability protocols, end-to-end service inter-working and messaging.
- *TSG RAN*—Radio Access Network, taking care of the aspects related to UMTS terrestrial radio access network. This includes the handling of specifications for the WCDMA-based radio interface towards the WCDMA terminal, RAN-specific interfaces Iub and Iur, and CN connectivity over Iu interface.
- *TSG GERAN*—GSM/EDGE Radio Access Network, taking care of the aspects related to the GSM/EDGE-based radio interface including testing for GERAN terminals,

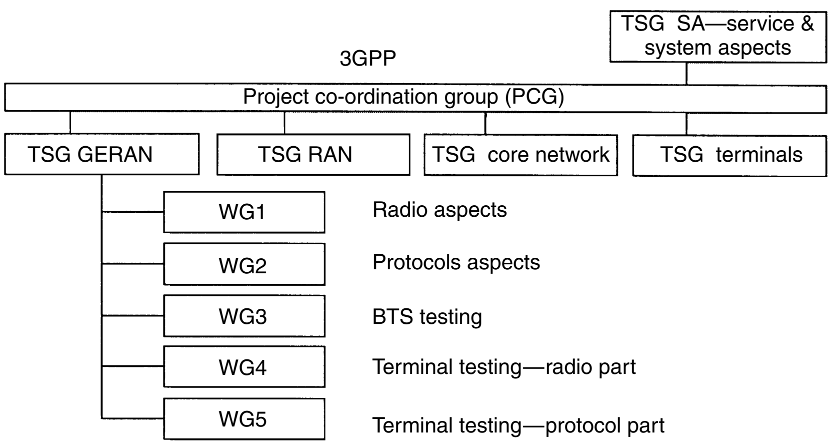


Figure 1.1. 3GPP organisation and TSG GERAN working groups

GERAN internal interfaces as well as external interfaces related to the connections towards the legacy GSM CN.

TSGs have the mandate to approve and modify technical specifications in their area. Although TSGs are reasonably independent in terms of making decisions, liaising between groups is intended to ensure the technical consistency and timely finalisation of the specifications for different releases. For TSG GERAN, this includes close cooperation, particularly with TSG SA on issues related to 3G bearer capabilities, architecture and security, and with TSG RAN on issues related to the common interfaces (Iu, Iur) and protocols (Packet Data Convergence Protocol (PDCP)) under TSG RAN responsibility, as well as overall optimisation of GSM/EDGE—WCDMA inter-operation.

While the TSGs carry the main responsibility for approval of the documents and overall work coordination, the detailed level technical work takes place in WGs. In TSG GERAN, there are currently five WGs:

- *WG1*—Radio Aspects, specifically having the responsibility over the specifications related to GSM/EDGE layer 1, radio frequency (RF), radio performance and internal Ater specification between Channel Codec Unit (CCU) and Transcoder and Rate Adapter Unit (TRAU).
- *WG2*—Protocols Aspects, specifically covering issues related to GERAN layer 2, like radio link control (RLC)/medium access control (MAC) and three radio resource (RR) specifications, A and Gb interfaces towards 2G CN and internal Abis specification between base station controller (BSC) and base transceiver station (BTS).
- *WG3*—Base Station Testing and O&M, taking care of all aspects related to conformance testing of GERAN BTSs, as well as GERAN-specific operation and maintenance issues for all GERAN nodes.
- *WG4*—Terminal Testing (Radio part), responsible for creating the specifications for the conformance testing of GERAN terminals in the area of radio interface layer 1 and RLC/MAC.
- *WG5*—Terminal Testing (Protocol part), responsible for creating the specifications for the conformance testing of GERAN terminals in the area of protocols above RLC/MAC.

1.2 Circuit-switched Services in GSM

The main drivers for the further development of the speech services and codecs are the user demand for better speech quality and the capacity gains enabling cost savings for network operators. Enhanced full-rate codec was the first significant improvement in speech quality when it was introduced in ETSI specifications in 1996 as well as in IS-641 standards in the United States during the same year. While the old GSM half-rate (HR) and full-rate (FR) codecs lacked the quality of the wireline telephony (32-kbps adaptive differential pulse code modulation (ADPCM)), EFR provided equivalent quality as a normal fixed connection even in typical error conditions. The EFR codec was jointly developed by Nokia and University of Sherbrooke and it has been widely used in operating GSM networks worldwide.

In spite of the commercial success of EFR, the codec left room for further improvements. In particular, the performance of the codec in severe radio channel error conditions could have been better. In addition to this, GSM half-rate codec was not able to provide adequate speech quality, so it was decided to continue the codec specification work with a new codec generation [3].

1.2.1 Adaptive Multi-rate Codec (AMR)

In October 1997, a new programme was initiated in ETSI, aimed at the development and standardisation of AMR for GSM system. Before the standardisation was officially started, a one-year feasibility study was carried out to validate the AMR concept. The main target of the work was to develop a codec that provided a significant improvement in error robustness, as well as capacity, over EFR.

The actual AMR codec standardisation was carried out as a competitive selection process consisting of several phases. In February 1999, ETSI approved the AMR codec standard, which was based on the codec developed in collaboration between Ericsson, Nokia and Siemens. Two months later, 3GPP adopted the AMR codec as the mandatory speech codec for the 3G WCDMA system. Some parts of the AMR codec work, such as voice activity detection (VAD) and optimised channel coding were finalised and included in the standard later, in June 1999.

The AMR codec contains a set of fixed-rate speech and channel codecs, fast in-band signalling and link adaptation. The AMR codec operates both in the full-rate (22.8 kbps) and half-rate (11.4 kbps) GSM channel modes. An important part of AMR is its ability to adapt to radio channel and traffic load conditions and select the optimum channel mode (HR or FR) and codec mode (bit rate trade-off between speech and channel coding) to deliver the best possible combination of speech quality and system capacity.

Even though the AMR codec functionality is not mandatory in Rel'98-compliant terminals, the large benefits this functionality will bring will probably drive the demand for the support of AMR codec in all terminals.

1.2.1.1 Speech and Channel Coding

The AMR speech codec utilises the algebraic code excitation linear prediction (ACELP) algorithm employed also in GSM EFR and D-AMPS EFR codecs. The codec is actually a combination of eight speech codecs with bit rates of 12.2, 10.2, 7.95, 7.4, 6.7, 5.9, 5.15 and 4.75 kbps. Table 2.1 in Chapter 2 summarises AMR speech codecs supported in the different releases. AMR narrowband (AMR NB) with Gaussian minimum shift keying (GMSK) modulation is supported already in Rel'98. Each codec mode provides a different distribution of the available bit rate between speech and channel coding. All of the codecs are defined for the full-rate channel mode, while the six lowest ones are defined also for the half-rate channel mode.

Channel coding performs error correction and bad-frame detection. The error correction in all the codec modes is based on the recursive systematic convolutional (RSC) coding with puncturing to obtain the required bit rates. Each codec mode utilises a 6-bit cyclic redundancy check (CRC) for detecting bad frames. In order to maximise the commonality with the existing GSM system, all channels use polynomials used for the previous GSM traffic channels.

1.2.1.2 In-band Signalling and Link Adaptation

In the basic AMR codec operation, shown in Figure 1.2, both the mobile station (MS) and the base transceiver station (BTS) perform channel quality estimation of the received signal. On the basis of the channel quality measurements, a codec mode command (downlink to the MS) or codec mode request (uplink to the BTS) is sent over the radio interface in in-band messages. The receiving end uses this information to choose the best codec mode for the prevailing channel condition. A codec mode indicator is also sent over the radio to indicate the current mode of operation in the sending side. The basic principle for the codec mode selection is that the mode chosen in the uplink may be different from the one used in the downlink direction, but the channel mode (HR or FR) must be the same.

The benefit of the in-band method is that it does not require a separate signalling channel for the message transfer. By sending the messages and the indicators together with the speech payload, the link adaptation operation can also be made faster, leading to improvements in the system performance.

The network controls the uplink and the downlink codec modes and channel modes. The MS must obey the codec mode command from the network, while the network may use any complementing information, in addition to codec mode request, to determine the downlink codec mode. The MS must implement all the codec modes. However, the network can support any combination of them, on the basis of the choice of the operator.

AMR also contains voice activity detection and discontinuous transmission (VAD/DTX). These are used to switch off the encoding and transmission during periods of silence, thereby reducing radio interference and extending the battery lifetime.

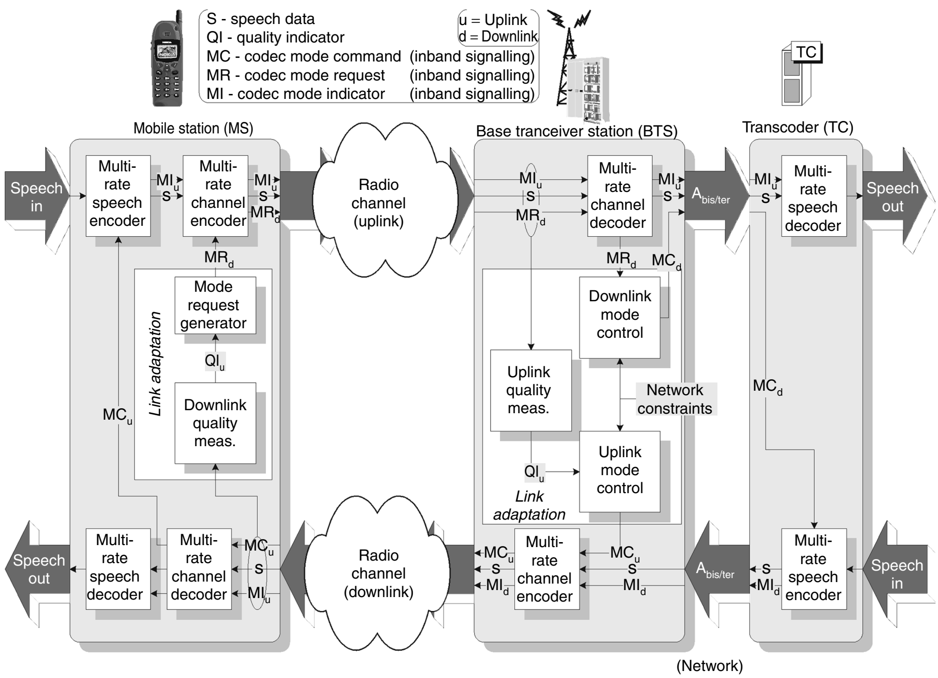


Figure 1.2. Block diagram of the AMR codec system

1.2.2 High Speech Circuit-switched Data (HSCSD)

The first-phase GSM specifications provided only basic transmission capabilities for the support of data services, with the maximum data rate in these early networks being limited to 9.6 kbps on one timeslot. HSCSD specified in Rel'96 was the first GSM Phase 2+ work item that clearly increased the achievable data rates in the GSM system. The maximum radio interface bit rate of an HSCSD configuration with 14.4-kbps channel coding is 115.2 kbps, i.e. up to eight times the bit rate on the single-slot full-rate traffic channel (TCH/F). In practice, the maximum data rate is limited to 64 kbps owing to core network and A-interface limitations.

The main benefit of the HSCSD feature compared to other data enhancements introduced later is that it is an inexpensive way to implement higher data rates in GSM networks owing to relatively small incremental modifications needed for the network equipment. Terminals, however, need to be upgraded to support multislot capabilities. The basic HSCSD terminals with relatively simple implementation (in size and cost compared to a single-slot terminal) available in the market today can receive up to four and transmit up to two timeslots and thus support data rates above 50 kbps. Figure 1.3 depicts the HSCSD network architecture based on the concept of multiple independent TCH/F channels.

In HSCSD, a new functionality is introduced at the network and terminals to provide the functions of combining and splitting the user data into separate n data streams, which will then be transferred via n channels at the radio interface, where $n = 1, 2, 3, \ldots, 8$. Once split, the data streams shall be carried by the n full-rate traffic channels, called *HSCSD channels*, as if they were independent of each other, for the purpose of data relay and radio interface L1 error control, until the point in the network where they are combined. However, logically the n full-rate traffic channels at the radio interface belong to the same HSCSD configuration, and therefore they shall be controlled as one radio link by the network for the purpose of cellular operations, e.g. handover.

For both transparent and non-transparent HSCSD connections, the call can be established with any number of TCH/F from one up to the maximum number of TCH/F, i.e. the minimum channel requirement is always one TCH/F.

If the wanted air interface user rate requirement cannot be met using a symmetric configuration, an asymmetric configuration can be chosen. The network shall, in this case, give priority to fulfilling the air interface user rate requirement in the downlink direction.

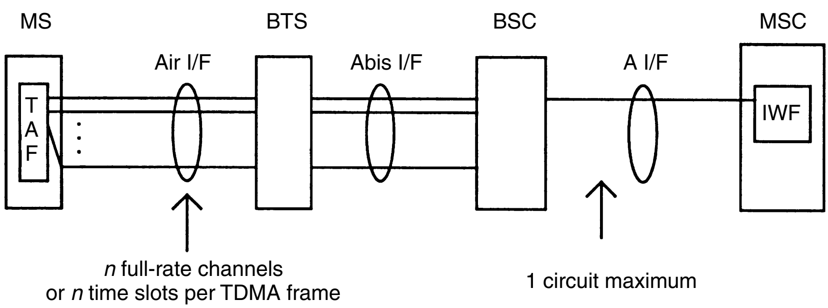


Figure 1.3. Network architecture for supporting HSCSD

The MS may request a service level upgrading or downgrading during the call, if so negotiated in the beginning of the call. This modification of channel requirements and/or wanted air interface user rate is applicable to non-transparent HSCSD connections only.

1.2.2.1 Radio Interface

Two types of HSCSD configurations exist: symmetric and asymmetric. For both types of configurations, the channels may be allocated on either consecutive or non-consecutive timeslots, taking into account the restrictions defined by the mobile station's multislot classes, described in detail in [2]. An example of the HSCSD operation with two consecutive timeslots is shown in Figure 1.4.

A symmetric HSCSD configuration consists of a co-allocated bi-directional TCH/F channel. An asymmetric HSCSD configuration consists of a co-allocated uni-directional or bi-directional TCH/F channel. A bi-directional channel is a channel on which the data are transferred in both uplink and downlink directions. On uni-directional channels for HSCSD, the data are transferred in downlink direction only. The same frequency-hopping sequence and training sequence is used for all the channels in the HSCSD configuration.

In symmetric HSCSD configuration, individual signal level and quality reporting for each HSCSD channel is applied. For an asymmetric HSCSD configuration, individual signal level and quality reporting is used for those channels. The quality measurements reported on the main channel are based on the worst quality measured among the main and the uni-directional downlink timeslots used. In both symmetric and asymmetric HSCSD configuration, the neighbouring cell measurement reports are copied on every uplink channel used. See [4] for more detail on signal level and quality reporting.

Transparent Data Transmission

In transparent data transmission, the V.110 data frames on the HSCSD channels carry data sub-stream numbers to retain the order of transmission over GSM, between the split/combine functions. Between these functions, channel internal multiframing is also used in order to increase the tolerance against inter-channel transmission delays. Depending on the location of the access point to external networks, the split/combine functionality is located at the base station sub-system (BSS) or in the IWF on the network side, and at the MS.

Non-transparent Data Transmission

Non-transparent mode of HSCSD is realised by modifying the radio link protocol (RLP) and L2R functions to support multiple parallel TCH/Fs instead of only one TCH/F

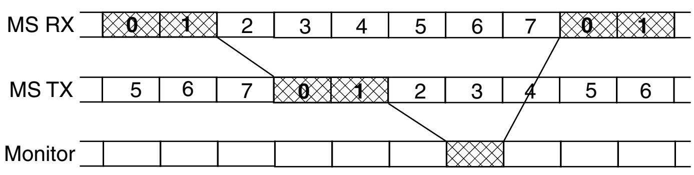


Figure 1.4. Double-slot operation in the air interface

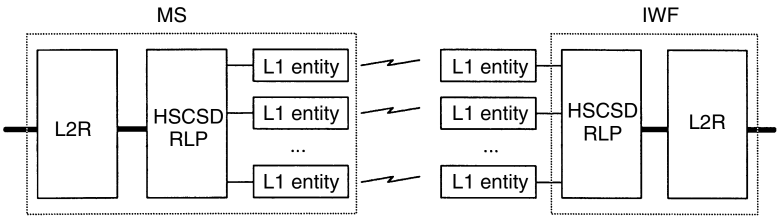


Figure 1.5. The HSCSD concept in non-transparent mode

(Figure 1.5). In addition, the RLP frame numbering is increased to accommodate the enlarged data-transmission rate.

1.3 Location Services

Different positioning methods found the basis for providing location services (LCS) in the GSM system. In general, these services can be implemented by using vendor-/operator-specific or standardised solutions. Both of these methods and their combination are likely to be used when LCS enter the operating networks.

The vendor-/operator-specific solutions utilise the information already available in the cellular network or measured by proprietary units installed for this purpose. The user location is thus estimated by combining and processing the available information with the help of tools and algorithms implemented for this purpose. In the standardised solution, the basic difference compared to the specific one is that the functionalities and capabilities required to retrieve the information for positioning are integrated into the system itself. The integrated solution affects the specifications and thus standardisation effort is needed, but benefit of this approach is that it enables multi-vendor support and provides a complete support for user privacy. In this chapter, a short overview of the history and drivers behind the standardisation of the location services is given. Chapter 4 describes in more detail the positioning methods, architecture and main technical principles behind them.

1.3.1 LCS Standardisation Process

The LCS standardisation began in ETSI SMG in 1995 after European operators asked for commercial services and separately in T1P1 in the United States after Federal Communications Commission (FCC) established the E911 requirements for the E911 service. The E911 requirement stated that by October 1, 2001, US operators must be able to locate the subscriber calling the emergency number 911.

However, several operators have applied and received waivers for the E911 requirements, and in spite of the different drivers behind the work in the United States and Europe, T1P1.5 (T1P1 is a subgroup of T1, a committee of TIA, the North American Telecommunications Industry Association) was given the responsibility to lead the overall standardisation activities in 1997. Four location methods were proposed and taken to the standardisation process:

- cell identity and timing advance;
- uplink time of arrival (TOA);

- enhanced observed time difference (E-OTD) methods based on measurements within the cellular network; as well as
- assisted GPS method based on the GPS technology.

In spring 1999, all four methods were included in the GSM specifications Rel'98 and Rel'99, including changes to GSM architecture and introduction of several new network elements for LCS purposes. Later in 2001, TOA method was excluded from the Rel'4 specifications because no commercial support for the scheme was foreseen because of the complexity in implementing the method in operating networks.

The first specification versions (Rel'98, Rel'99 and Rel'4) include the LCS support for circuit-switched connections over the A-interface towards the 2G CN only. In Rel'5, the focus is then moved towards providing the same LCS support in puncturing scheme (PS) domain connections over Gb and Iu interfaces. The main reasons for this gradual introduction of different modes were not only the regulatory requirements that, in practice, were applicable for coding scheme (CS) speech connections only but also the delays in the introduction of GPRS and its interfaces. When the penetration of GPRS users and terminals increases, LCS is expected to be more and more an attractive increment to the services provided to the GPRS subscribers in 'always connected' mode.

1.4 General Packet Radio System (GPRS)

1.4.1 Introduction of GPRS (Rel'97)

Soon after the first GSM networks became operational in the early 1990s and the use of the GSM data services started, it became evident that the circuit-switched bearer services were not particularly well suited for certain types of applications with a bursty nature. The circuit-switched connection has a long access time to the network, and the call charging is based on the connection time. In packet-switched networks, the connections do not reserve resources permanently, but make use of the common pool, which is highly efficient, in particular, for applications with a bursty nature. The GPRS system will have a very short access time to the network and the call charging could solely be based on an amount of transmitted data.

The GPRS system brings the packet-switched bearer services to the existing GSM system. In the GPRS system, a user can access the public data networks directly using their standard protocol addresses (IP, X.25), which can be activated when the MS is attached to the GPRS network. The GPRS MS can use between one and eight channels over the air interface depending on the MS capabilities, and those channels are dynamically allocated to an MS when there are packets to be sent or received. In the GPRS network, uplink and downlink channels are reserved separately, making it possible to have MSs with various uplink and downlink capabilities. The resource allocation in the GPRS network is dynamic and dependent on demand and resource availability. Packets can also be sent on idle time between speech calls. With the GPRS system, it is possible to communicate point-to-point (PTP) or point-to-multipoint (PTM); it also supports the SMS and anonymous access to the network. The theoretical maximum throughput in the GPRS system is 160 kbps per MS using all eight channels without error correction.

1.4.2 GPRS Network Architecture

In Figure 1.6, a functional view of the GPRS network is displayed. GPRS brings a few new network elements to the GSM network. The most important ones are the serving GPRS support node (SGSN) and the gateway GPRS support node (GGSN). Another important new element is the point-to-multipoint service centre (PTM-SC), which is dedicated to the PTM services in the GPRS network. Another new network element is the border gateway (BG), which is mainly needed for security reasons and is situated on the connection to the inter-PLMN backbone network. The inter-PLMN and intra-PLMN backbone networks are also new elements, both Internet protocol–based (IP-based) networks. In addition, there will be a few new gateways in the GPRS system like the charging gateway and the legal interception gateway.

While the current GSM system was originally designed with an emphasis on voice sessions, the main objective of the GPRS is to offer an access to standard data networks such as transport control protocol (TCP)/Internet protocol (IP) and X.25. These other networks consider GPRS just as a normal sub-network (Figure 1.7). A GGSN in the GPRS network behaves as a router and hides the GPRS-specific features from the external data network.

The mobile user can have either a static or a dynamic data network address, and the data traffic will always use the gateway indicated by this address (Figure 1.8). However, the home network operator could force all the traffic to use a home GGSN, for example, for security reasons.

A static address is permanently allocated for one subscriber. As it will point to a gateway of the home network, the data packets will always be routed through the home network. Figure 1.8 illustrates the case where a user is in his home network (case 1) and in a visited network (case 2).

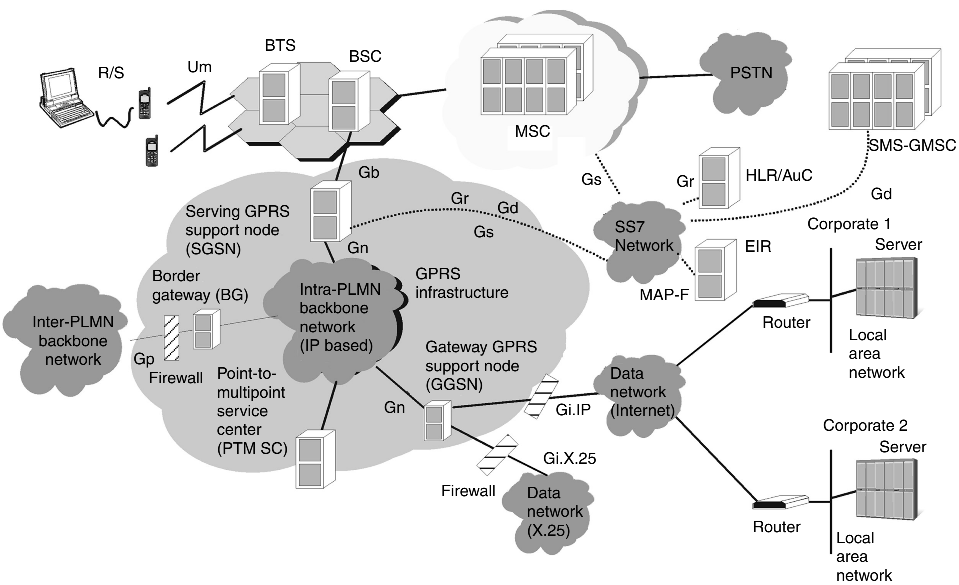


Figure 1.6. Functional view of GPRS

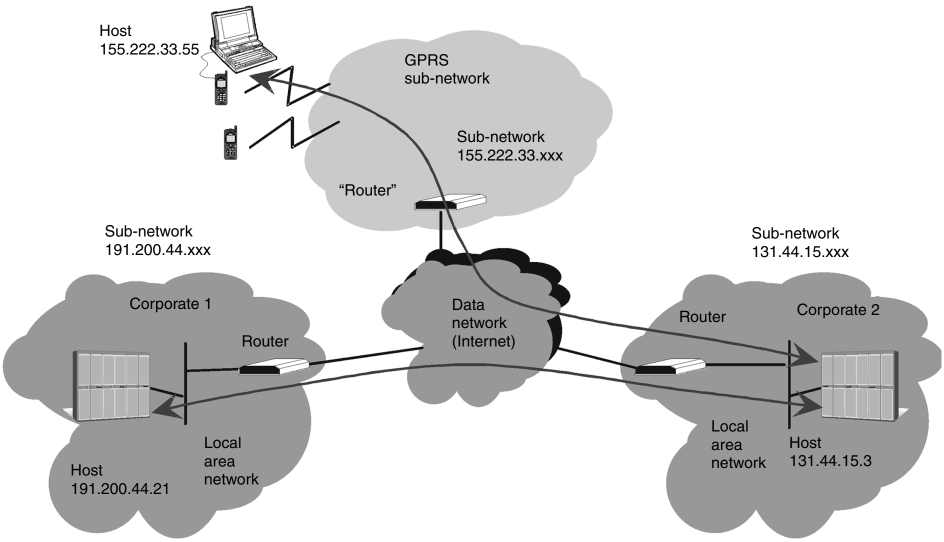


Figure 1.7. GPRS network seen by another data network

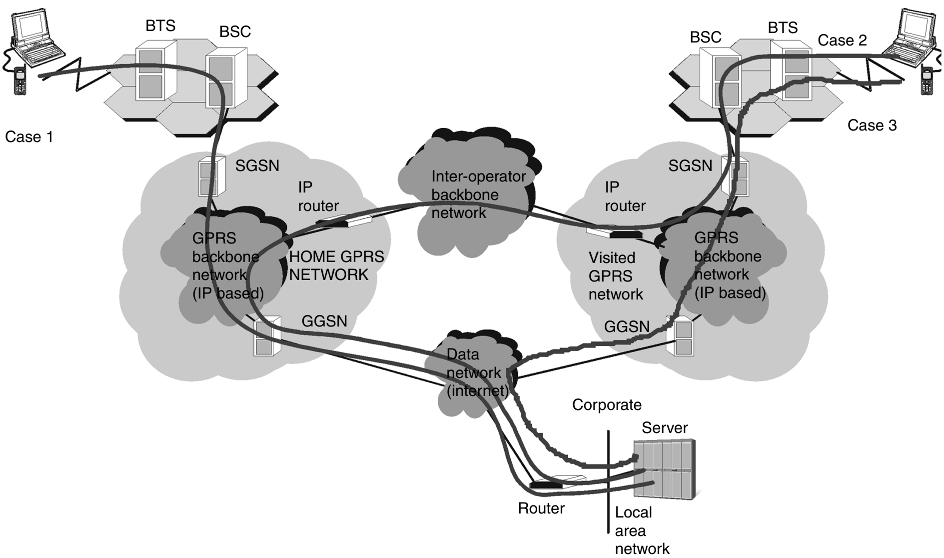


Figure 1.8. Data transfer

Each network can also have a pool of available addresses that can be allocated dynamically to the users by the GGSN. This decreases the number of addresses needed by an operator. A dynamic address is allocated to a user only for the time of a connection. To avoid routing the packet through the home network, the dynamic address can be allocated by a GGSN from the visited network (case 3 in Figure 1.8). It can also be allocated by a GGSN from the home network (case 2) to apply, for example, some specific screening.

The incoming and outgoing PTP traffic can be charged on the basis of the quantity of data transmitted, the protocol used, the length of connection, etc.

1.4.2.1 GPRS Mobiles

A GPRS MS can operate in one of three modes of operation (Table 1.1):

- *Class A mode of operation*. The MS is attached to both GPRS and other GSM services. The mobile user can make and/or receive calls on the two services simultaneously, subject to the quality of service (QoS) requirements, e.g. having a normal GSM voice call and receiving GPRS data packets at the same time.
- *Class B mode of operation*. The MS is attached to both GPRS and other GSM services, but the MS can only operate one set of services at a time. The MS in idle mode (and packet idle mode) is required to monitor paging channels (PCHs) for both circuit-switched and packet-switched services. However, the practical behaviour of the MS depends on the mode of network operation. For example, one mode of network operation is defined so that when an MS is engaged in packet data transfer, it will receive paging messages via the packet data channel (PDCH) without degradation of the packet data transfer.
- *Class C mode of operation*. The MS can only be attached to either the GSM network or the GPRS network. The selection is done manually and there are no simultaneous operations.

In GSM Phase 2, the MS uses one channel for the uplink traffic and one channel for the downlink traffic (Figure 1.9, '1-slot'). In GPRS, it is possible to have a multiple slot MS, e.g. a 2-slot MS with two uplink channels and two downlink channels (Figure 1.9, '2-slot'). The MS can have a different (asymmetric) uplink and downlink capability.

When an MS supports the use of multiple timeslots, it belongs to a multislot class as defined in [5]. Multislot class is defined as a combination of several parameters:

- The maximum number of received timeslots that the MS can use per TDMA frame.
- The maximum number of transmit timeslots that the MS can use per TDMA frame.
- Total number of uplink and downlink timeslots that can actually be used by the MS per TDMA frame.
- The time needed for the MS to perform adjacent cell signal level measurement and get ready to transmit (T_{ta}).
- The time needed for the MS to get ready to transmit (T_{tb}).

Table 1.1. GPRS mobile classes

Class A mode of operation	Class B mode of operation	Class C mode of operation
Full simultaneous use of packet and circuit mode connections	No simultaneous traffic but automatic sequential service	Alternate use or a GPRS-only MS

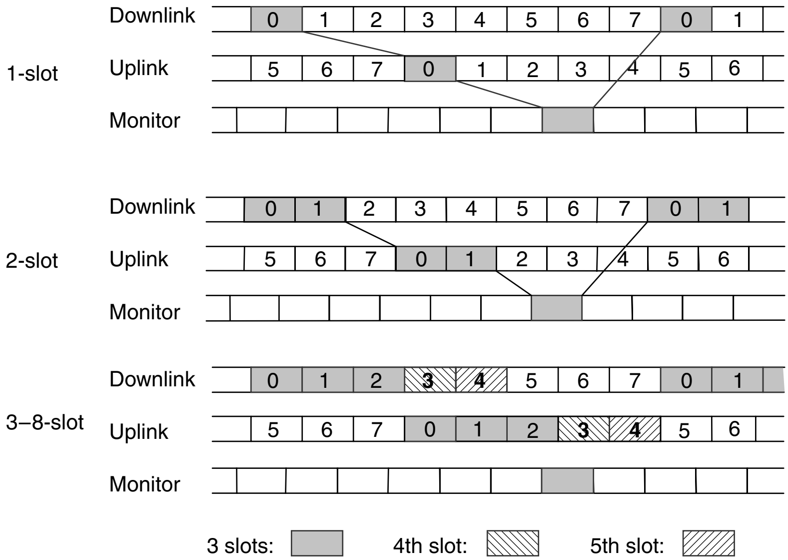


Figure 1.9. GPRS multislot capabilities

- The time needed for the MS to perform adjacent cell signal level measurement and get ready to receive (T_{ra}).
- The time needed for the MS to get ready to receive (T_{rb}).
- Capability to receive and transmit at the same time. There are two types of MS:
  - Type 1: The MS is not required to transmit and receive at the same time.
  - Type 2: The MS is required to transmit and receive at the same time.

A combination of these factors has led to a definition of 29 MS classes (in Rel'97 of GSM specifications). Most of the GPRS terminals (if not all) are type 1 owing to the simpler radio design. Table 1.2 shows the multislot classes. A complete multislot classes table (including Type 2 terminals) can be found in Annex B of [5].

1.4.2.2 Base Station Sub-system (BSS)

The BSS is upgraded with new GPRS protocols for the Gb interface (between the BSS and SGSN) and enhanced layer 2 (RLC/MAC see the section entitled *Medium access control and radio link control layer*) protocols for the air interface. The Gb interface connects the BSS and the SGSN, allowing the exchange of signalling information and user data. Both air and Gb interfaces allow many users to be multiplexed over the same physical resources. The BSS allocates resources to a user upon activity (when data are sent or received) and the data are reallocated immediately thereafter. The Gb interface link layer is based on frame relay, which is used for signalling and data transmission. The base station sub-system GPRS protocol (BSSGP) provides the radio-related QoS and routing information that is required to transmit user data between a BSS and an SGSN.

Table 1.2. Type 1 multislot classes

	Maximum number of slots			Minimum number of slots			
Multislot class	Rx	Tx	Sum	T_{ta}	T_{tb}	T_{ra}	T_{rb}
1	1	1	2	3	2	4	2
2	2	1	3	3	2	3	1
3	2	2	3	3	2	3	1
4	3	1	4	3	1	3	1
5	2	2	4	3	1	3	1
6	3	2	4	3	1	3	1
7	3	3	4	3	1	3	1
8	4	1	5	3	1	2	1
9	3	2	5	3	1	2	1
10	4	2	5	3	1	2	1
11	4	3	5	3	1	2	1
12	4	4	5	2	1	2	1

1.4.2.3 Mobile Services Switching Centre/Home Location Register (MSC/HLR)

The HLR is upgraded and contains GPRS subscription data. The HLR is accessible from the SGSN via the Gr interface and from the GGSN via the optional Gc interface. For roaming MSs, HLR is in a different public land mobile network (PLMN) than the current SGSN. All MSs use their HLR in home public land mobile network (HPLMN). The HLR is enhanced with GPRS subscriber information.

The MSC/visiting location register (VLR) can be enhanced for more efficient coordination of GPRS and non-GPRS services and functionality by implementing the Gs interface, which uses the BSSAP+ procedures, a subset of BSS application part (BSSAP) procedures. Paging for circuit-switched calls can be performed more efficiently via the SGSN. This is the case as well for the combined GPRS and non-GPRS location updates. The Gs interface also allows easy Class B mode of operation for MS design. The SMS-GMSC and SMS-IWMSC functions are enhanced for SMS over GPRS. Also, the temporary mobile subscriber identity (TMSI) allocation procedure is modified.

1.4.2.4 SGSN

The SGSN is a main component of the GPRS network, which handles, e.g. the mobility management and authentication and has register function. The SGSN is connected to the BSC and is the service access point to the GPRS network for the GPRS MS. The SGSN handles the protocol conversion from the IP used in the backbone network to the sub-network-dependent convergence protocol (SNDCP) and logical link control (LLC) protocols used between the SGSN and the MS. These protocols handle compression and ciphering. The SGSN also handles the authentication of GPRS mobiles, and when the authentication is successful, the SGSN handles the registration of an MS to the GPRS network and takes care of its mobility management. When the MS wants to send (or receive) data to (from) external networks, the SGSN relays the data between the SGSN and relevant GGSN (and vice versa).

1.4.2.5 GGSN

The GGSN is connected to the external networks like the Internet and the X.25. From the external networks' point of view, the GGSN is a router to a sub-network, because the GGSN 'hides' the GPRS infrastructure from the external networks. When the GGSN receives data addressed to a specific user, it checks if the address is active. If it is, the GGSN forwards the data to the SGSN serving the MS, but if the address is inactive, the data are discarded. The mobile-originated packets are routed to the right network by the GGSN. The GGSN tracks the MS with a precision of SGSN.

1.4.3 GPRS Interfaces and Reference Points

The GPRS system introduces new so-called G-interfaces to the GSM network architecture. It is important to understand the function of every interface and reference point because this gives an insight to the GPRS system and consequent evolution. Figure 1.10 gives a logical architecture description with the interfaces and reference points of the GSM network with GPRS.

Connections of the GPRS system to the network and switching sub-system (NSS) part of the GSM network are implemented through signalling system number 7 (SS7) network (Gc, Gd, Gf, Gr, Gs), while the other interfaces and reference points are implemented through the intra-PLMN backbone network (Gn), the inter-PLMN backbone network (Gp) or the external networks (Gi). The different interfaces that the GPRS system uses are as follows:

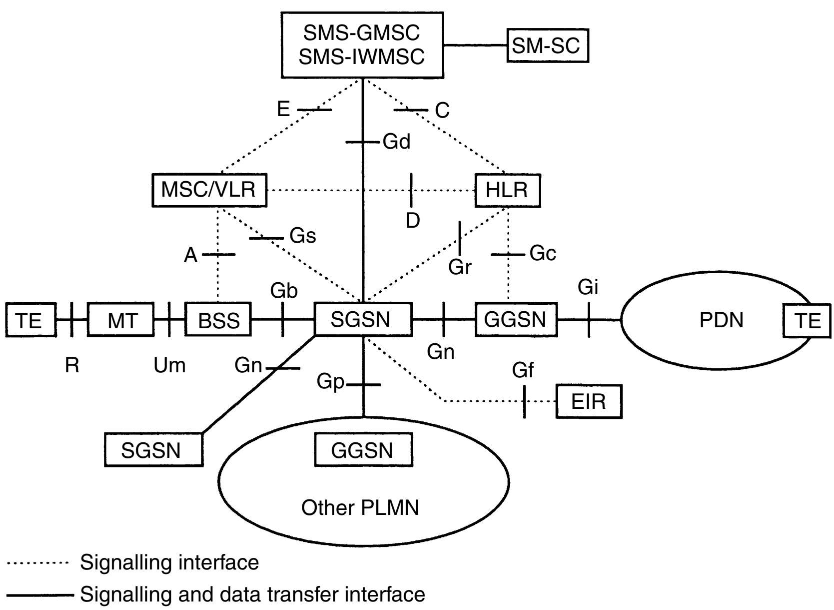


Figure 1.10. Logical architecture

- Gb between an SGSN and a BSS. The Gb interface is the carrier of the GPRS traffic and signalling between the GSM radio network (BSS) and the GPRS part. Frame relay–based network services (NSs) provide flow control for this interface.
- Gc between the GGSN and the HLR. The GGSN may request location information for network-requested context activation only via this optional interface. The standard also defines the use of a proxy GSN, which is used as a GPRS tunnelling protocol (GTP) to mobile application part (MAP) protocol converter, thus avoiding implementing MAP in GGSN.
- Gd between the SMS-GMSC and an SGSN, and between SMS-IWMSC and an SGSN. The Gd interface allows more efficient use of the SMS services.
- Gf between an SGSN and the Equipment Identity Register (EIR). The Gf gives the SGSN access to equipment information. In the EIR, the MSs are divided into three lists: black list for stolen mobiles, grey list for mobiles under observation and white list for other mobiles.
- Gn between two GSNs within the same PLMN. The Gn provides a data and signalling interface in the intra-PLMN backbone. The GTP is used in the Gn (and in the Gp) interface over the IP-based backbone network.
- Gp between two GSNs in various PLMNs. The Gp interface provides the same functionality as the Gn interface, but it also provides, with the BG and the Firewall, all the functions needed in the inter-PLMN networking, i.e. security, routing, etc.
- Gr between an SGSN and the HLR. The Gr gives the SGSN access to subscriber information in the HLR, which can be located in a different PLMN than the SGSN.
- Gs between an SGSN and an MSC. The SGSN can send location data to the MSC or receive paging requests from the MSC via this optional interface. The Gs interface will greatly improve effective use of the radio and network resources in the combined GSM/GPRS network. This interface uses BSSAP+ protocol.
- Um between a MS and the GPRS fixed network part. The Um is the access interface for the MS to the GPRS network. The MS has a radio interface to the BTS, which is the same interface used by the existing GSM network with some GPRS-specific changes.

There are two different reference points in the GPRS network. The Gi is GPRS-specific, but the R is common with the circuit-switched GSM network. The two reference points in the GPRS are as follows:

- Gi between a GGSN and an external network. The GPRS network is connected to external data networks via this interface. The GPRS system will support a variety of data networks and that is why the Gi is not a standard interface, but merely a reference point.
- R between terminal equipment and mobile termination. This reference point connects terminal equipment to mobile termination, thus allowing, e.g. a laptop-PC to transmit data over the GSM phone. The physical R interface follows, e.g. the ITU-T V.24/V.28 or the PCMCIA PC-Card standards.

1.4.4 GPRS Protocol Architecture

The GPRS system introduces a whole new set of protocols for the GSM Phase 2+ network. The inter-working between the new network elements is done with new GPRS-specific protocols. However, there are a number of existing protocols used at the lower layers of the protocol stacks, namely, TCP/user datagram protocol (UDP) IP. Figure 1.11 shows the transmission plane used in the GPRS system.

1.4.4.1 Physical Layer

The physical layer has been separated into two distinct sub-layers, the physical RF layer and the physical link layer.

The *physical RF layer* performs the modulation of the physical waveforms based on the sequence of bits received from the physical link layer. The physical RF layer also demodulates received waveforms into a sequence of bits that are transferred to the physical link layer for interpretation. The GSM physical RF layer is defined in GSM 05 series specifications, which define the following among other things:

- The carrier frequency characteristics and GSM radio channel structures [5].
- The modulation of the transmitted waveforms and the raw data rates of GSM channels [6]. Note that in the GPRS Rel'97 radio interface, only the original GSM modulation method, GMSK modulation, with the carrier bit rate of 270.833 kbps is defined. For one single timeslot, this corresponds to the gross data rate of 22.8 kbps.
- The transmitter and receiver characteristics and performance requirements [7].
- The *physical link layer* provides services for information transfer over a physical channel between the MS and the network. These functions include data unit framing, data coding and the detection and correction of physical medium transmission errors. The physical link layer operates above the physical RF layer to provide a physical channel between the MS and the network. The purpose of the physical link layer is to

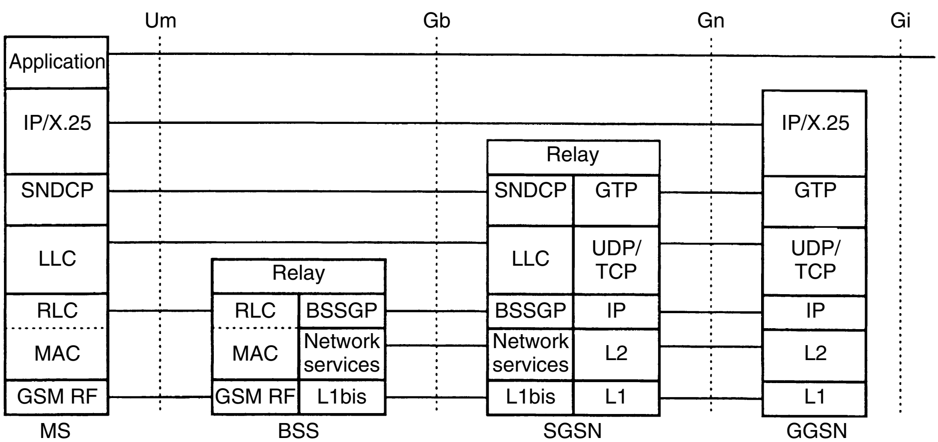


Figure 1.11. Transmission plane

convey information across the GSM radio interface, including RLC/MAC information. The physical link layer supports multiple MSs sharing a single physical channel and provides communication between MSs and the network. In addition, it provides the services necessary to maintain communications capability over the physical radio channel between the network and MSs. Radio sub-system link control procedures are specified in [4].

The physical link layer is responsible for the following:

- Forward error correction (FEC) coding, allowing the detection and correction of transmitted code words and the indication of uncorrectable code words. The coding schemes are described in the section entitled *Physical link layer*.
- Rectangular interleaving of one radio block over four bursts in consecutive TDMA frames, as specified in [8].
- Procedures for detecting physical link congestion.

The physical link layer control functions include

- synchronisation procedures, including means for determining and adjusting the MS timing advance to correct for variances in propagation delay [9],
- monitoring and evaluation procedures for radio link signal quality.
- cell selection and re-selection procedures.
- transmitter power control procedures.
- battery power saving procedures, e.g. discontinuous reception (DRX) procedures.

For more details, see the section entitled *Physical link layer*.

1.4.4.2 RLC/MAC

The RLC/MAC layer provides services for information transfer over the physical layer of the GPRS radio interface. These functions include backward error correction procedures enabled by the selective re-transmission of erroneous blocks. The RLC function offers a reliable radio link to the upper layers. The MAC function handles the channel allocation and the multiplexing, i.e. the use of physical layer functions. The RLC/MAC layer together form the open system interconnection (OSI) Layer 2 protocol for the Um interface and uses the services of the physical link layer (see the section entitled *Medium Access Control and Radio Link Control Layer*).

1.4.4.3 LLC

The logical link control layer offers a secure and reliable logical link between the MS and the SGSN to upper layers, and is independent of the lower layers. The LLC layer has two transfer modes—the acknowledged and unacknowledged. The LLC conveys signalling, SMS and SNDCP packets. See [10] for more information.

1.4.4.4 SNDCP

The sub-network-dependent convergence protocol (SNDCP) is a mapping and compression function between the network layer and lower layers. It also performs segmentation, reassembling and multiplexing. The SNDCP protocol is specified in [11].

Signalling has various sets of protocols, which are used with the existing GSM network elements. Internal signalling in the GPRS system is handled by protocols, which carry both data and signalling (LLC, GTP and BSSGP). The GPRS control plane is shown in Figure 1.12.

Figure 1.13 describes how network protocols are multiplexed. NSAPI is the network layer service access point identifier, which is used to identify the packet data protocol (PDP) context at SNDCP level. Service access point identifier (SAPI) is used to identify the points where LLC provides service to higher layers. SAPIs have different priorities. TLLI is the temporary logical link identity, which unambiguously identifies the logical link between the MS and the SGSN. The IP or the X.25 is the packet protocol offered to the subscriber by the GPRS system.

1.4.4.5 BSSGP

The primary functions of the base station sub-system GPRS protocol (BSSGP) (see [12]) include, in the downlink, the provision by an SGSN to a BSS of radio-related information used by the RLC/MAC function; in the uplink, the provision by a BSS to an SGSN of radio-related information derived from the RLC/MAC function; and the provision of functionality to enable two physically distinct nodes, an SGSN and a BSS, to operate node management control functions. See [12] for more details. The underlying network service is responsible for the transport of BSSGP packet data units (PDUs) between a BSS and an SGSN [12].

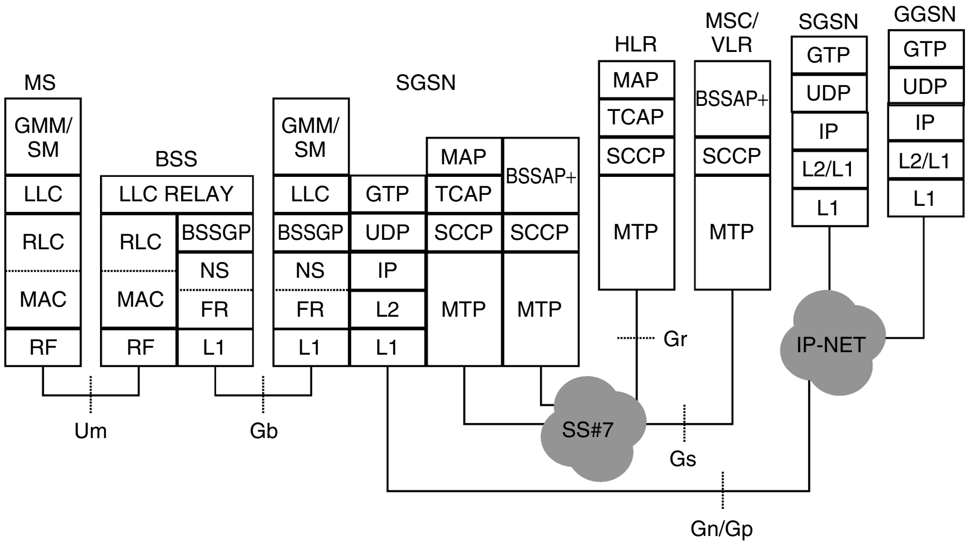


Figure 1.12. GPRS control plane

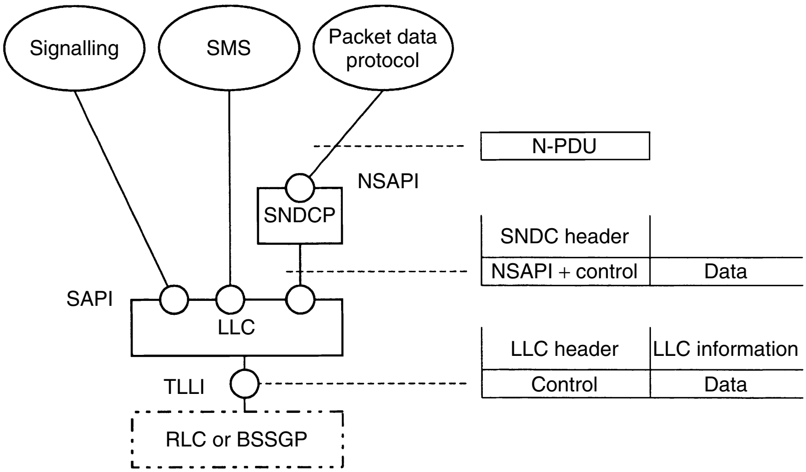


Figure 1.13. Multiplexing of network protocols

1.4.4.6 GTP

The GPRS tunnelling protocol (GTP) is used to tunnel data and signalling between the GSNs. The GTP can have proprietary extensions to allow proprietary features. The relay function relays PDP (packet data protocol) PDUs between the Gb and the Gn interfaces. For details see [13].

1.4.5 Mobility Management

The mobility management in the GPRS network is handled almost the same way as in the existing GSM system. One or more cells form a routing area, which is a subset of one location area. Every routing area is served by one SGSN. The tracking of the location of an MS depends on the mobility management state. When an MS is in a STANDBY state, the location of the MS is known on a routing area level. When the MS is in a READY state, the location of the MS is known on a cell level. Figure 1.14 shows the different mobility management states and transitions between them.

1.4.5.1 Mobility Management States

The GPRS has three various mobility management states. The IDLE state is used when the subscriber (MS) is passive (not GPRS attached). The STANDBY state is used when the subscriber has ended an active phase. An MS is in an active phase (READY state) when it is transmitting or has just been transmitting. The change between the states happens upon activity or when a timer expires. A description of the various states is given below.

IDLE State

The subscriber is not reachable by the GPRS network. The MS is only capable of receiving PTM-M data. The network elements hold no valid context for the subscriber and the subscriber is not attached to the mobility management. In order to change state, the MS has to perform a GPRS attach procedure.

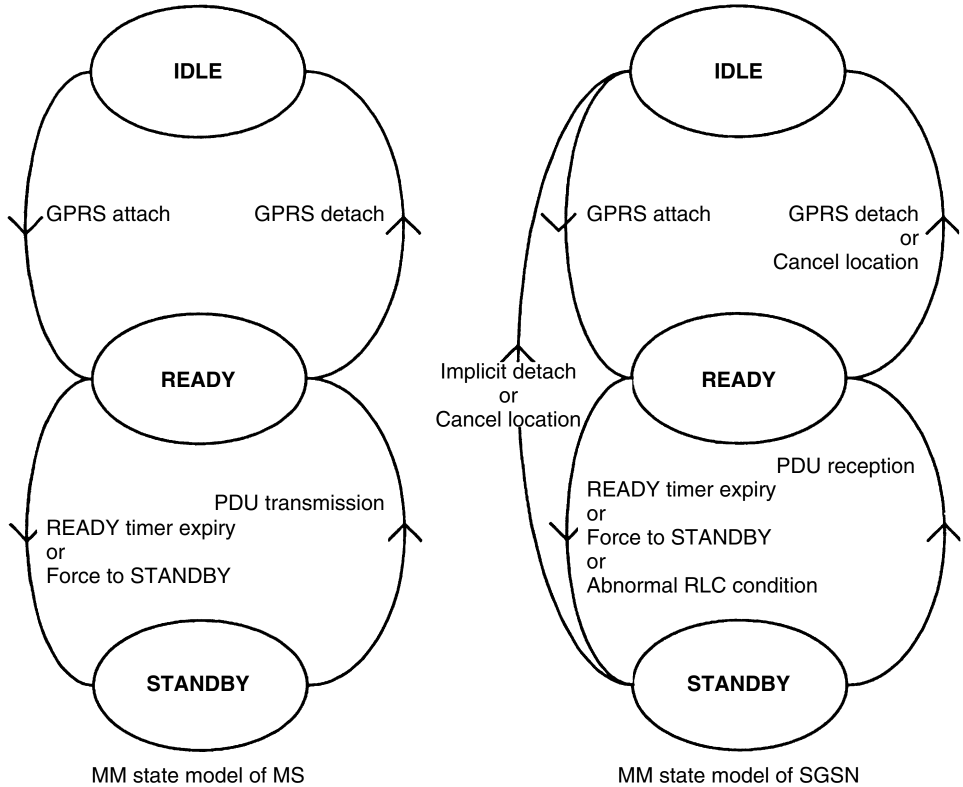


Figure 1.14. Mobility management state

STANDBY State

The subscriber is attached to the mobility management and the location of an MS is known on a routing area level. The MS is capable of receiving PTM data and pages for PTP data. The network holds a valid mobility management context for the subscriber. If the MS sends data, the MS moves to READY state. The MS or the network can initiate the GPRS detach procedure to move to IDLE state. After expiry of the MS reachable timer, the network can detach the MS. The MS can use the discontinuous reception (DRX) to save the battery.

READY State

The subscriber is attached to the mobility management and the location of an MS is known on a cell level. The MS is capable of receiving PTM and PTP data. The SGSN can send data to the MS without paging at any time and the MS can send data to the SGSN at any time. The network holds a valid mobility management context for the subscriber. If the READY timer expires, the MS moves to STANDBY state. If the MS performs a GPRS detach procedure, the MS moves to IDLE state and the mobility management context is removed. An MS in READY state does not necessarily have radio resources reserved. The MS can use the DRX to save the battery.

1.4.5.2 GPRS Attach and Detach

GPRS attach and GPRS detach are mobility management functions to establish and to terminate a connection to the GPRS network. The SGSN receives the requests and processes them. With the GPRS attach the mobile moves to READY state and the mobility management context is established, the MS is authenticated, the ciphering key is generated, a ciphered link established and the MS is allocated a temporary logical link identity. The SGSN fetches the subscriber information from the HLR. After a GPRS attach, the SGSN tracks the location of the MS. The MS can send and receive SMS, but no other data. To transfer other data, it has to first activate a PDP context.

When the subscriber wants to terminate a connection to the GPRS network, the GPRS detach is used. The GPRS detach moves the MS to IDLE state and the mobility management context is removed. The MS can also be detached from the GPRS implicitly when the mobile reachable timer expires. The GPRS detach is normally generated by the MS, but can also be generated by the network.

1.4.6 PDP Context Functions and Addresses

The packet data protocol (PDP) context functions are network level functions, which are used to bind an MS to various PDP addresses and afterwards used to unbind the MS from these addresses. The PDP context can also be modified. When an MS is attached to the network, it has to activate all the addresses it wants to use for data traffic with the external networks. Various PDP contexts have to be activated because these include address, QoS attributes, etc. After the subscriber has finished the use of activated addresses, these have to be deactivated. Also, change of address specific attributes have to be performed sometimes. The MS can use these functions when in STANDBY or READY state. The GGSN can also use these functions, but the SGSN is responsible for performing these functions. The MS can select a particular GGSN for access to certain services and also activate a PDP context anonymously, without using any identification.

1.4.6.1 Dynamic and Static PDP Address

The subscriber can use various kinds of PDP addresses. A static PDP address is permanently assigned to an MS by the HPLMN. An HPLMN dynamic PDP address is assigned to an MS by the HPLMN when an MS performs a PDP context activation. A visited public land mobile network (VPLMN) dynamic PDP address is assigned to an MS by the VPLMN when an MS performs a PDP context activation. It is indicated in the subscription if the MS can have a dynamic address or not.

1.4.6.2 PDP Context Activation

The PDP context activation can be done by the MS or by the network. The activate PDP context request is sent to the SGSN when a certain address requires activation. As an option, the request can also be made by the GGSN, if packets are received for an address without active context and the MS is GPRS attached. The request contains parameters for the context, like the TLLI, the protocol type, the address type, QoS, requested GGSN, etc.

1.4.6.3 PDP Context Modification

The PDP context can be modified by the SGSN with the modify PDP context. Only the parameters' QoS negotiated and radio priority can be changed. The SGSN sends the request to the MS, which either accepts the new QoS by sending a modify PDP context accept to the SGSN or does not accept it by sending a deactivate PDP context request for that context. In GPRS Phase 2 also, MS and GGSN can modify PDP context.

1.4.6.4 PDP Context Deactivation

The PDP context can be deactivated by the MS or by the network. Every address can be deactivated separately, but when the GPRS detach is performed, the network will automatically remove all the PDP contexts.

1.4.7 Security

The GPRS system uses GSM Phase 2–based security. These security functions include the authentication of the subscriber, the user identity confidentiality and the ciphering of the data traffic between the MS and the SGSN. The authentication of the subscriber is done the same way by the SGSN in the GPRS system as by the MSC/VLR in the Phase 2 GSM network. The TLLI is used to keep the subscriber identity confidential. The correspondence between the international mobile subscriber identity (IMSI) and the TLLI is only known by the MS and the SGSN. The ciphering function used between the MS and the SGSN is not the same as that used in the GSM Phase 2, but an optimised one for the packet-switched traffic. Security of the backbone is provided by using a private network, thus avoiding the possibility that an external *hacker* can address it. Each operator must guarantee its physical security.

1.4.8 Location Management

The location management procedures are used to handle the changing of a cell and/or a routing area, and the periodic routing area updates. If an MS stays a long time at the same place, the network has to receive an indication that the MS is still reachable. This is the reason why periodic routing area updates are made. All the MSs attached to the GPRS will perform a periodic routing area update. Only the Class B mode of operation mobiles, which are engaged in a circuit-switched communication, cannot. The MS performs a cell update when it changes cell within a routing area in READY mode. When the MS changes cell between the different routing areas, it performs a routing area update. There are two types of routing area updates, the intra-SGSN routing area update and the inter-SGSN routing area update. An SGSN can manage many routing areas and if the new routing area belongs to the management of a new SGSN, the inter-SGSN routing area update is used. If the new routing area belongs to the management of the same SGSN than the old one, the intra-SGSN routing area update is used. The old SGSN forwards user packets to the new SGSN, until it receives a cancel location from the HLR.

1.4.9 GPRS Radio Interface

This section gives an overview of the GPRS Rel'97 radio interface.

1.4.9.1 GPRS Packet Data Logical Channels

This section describes the logical channels of the GPRS radio interface. The reason for defining a new set of control channels for the GPRS packet data, which is partly parallel to that of the circuit-switched control channels, is to be able to flexibly allocate more signalling capacity for the packet data traffic without sacrificing the quality of the speech traffic.

The packet data logical channels are mapped onto the physical channels that are dedicated to packet data. The physical channel dedicated to packet data traffic is called a *packet data channel* (PDCH).

The packet data logical channels can be grouped into different categories as shown in Figure 1.15. These categories (packet common control channels, packet broadcast control channels, packet traffic channels and packet dedicated control channels) are introduced in more detail in the following list [14].

Packet Common Control Channel (PCCCH)

PCCCH comprises logical channels for common control signalling used for packet data:

- *Packet random access channel (PRACH)—uplink only.* PRACH is used by the MS to initiate uplink transfer to send data or signalling information.
- *Packet paging channel (PPCH)—downlink only.* PPCH is used to page an MS prior to downlink packet transfer.
- *Packet access grant channel (PAGCH)—downlink only.* PAGCH is used in the packet transfer establishment phase to send resource assignment to an MS prior to packet transfer.
- *Packet notification channel (PNCH)—downlink only.* PNCH is used to send a PTM-M (point to multipoint—multicast) notification to a group of MSs prior to a PTM-M packet transfer. (Note that the PTM-M service is not specified in GPRS Rel'97.)

Packet Broadcast Control Channel (PBCCH)—Downlink only

PBCCH broadcasts packet data specific system information. If PBCCH is not allocated, the packet data specific system information is broadcast on BCCH.

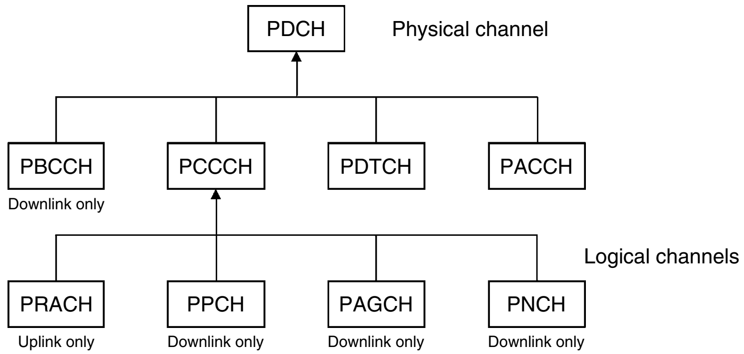


Figure 1.15. GPRS logical channels

Packet Traffic Channels

- *Packet data traffic channel (PDTCH).* PDTCH is a channel allocated for data transfer. It is temporarily dedicated to one MS. In the multislot operation, one MS may use multiple PDTCHs in parallel for individual packet transfer.

All packet data traffic channels are uni-directional, either uplink (PDTCH/U), for a mobile-originated packet transfer, or downlink (PDTCH/D), for a mobile terminated packet transfer.

Packet Dedicated Control Channels

- *Packet associated control channel (PACCH).* PACCH conveys signalling information related to a given MS. The signalling information includes, e.g. acknowledgements and power control information. PACCH also carries resource assignment and reassignment messages comprising the assignment of a capacity for PDTCH(s) and for further occurrences of PACCH. The PACCH shares resources with PDTCHs, which are currently assigned to one MS. Additionally, an MS that is currently involved in packet transfer can be paged for circuit-switched services on PACCH.
- *Packet timing advance control channel, uplink (PTCCH/U).* PTCCH/U is used to transmit random access bursts to allow estimation of the timing advance for one MS in packet transfer mode.
- *Packet timing advance control channel, downlink (PTCCH/D).* PTCCH/D is used to transmit timing advance information updates to several MSs. One PTCCH/D is paired with several PTCCH/Us.

1.4.9.2 Mapping of Packet Data Logical Channels into Physical Channels

Different packet data logical channels can occur on the same physical channel (i.e. PDCH). The sharing of the physical channel is based on blocks of four consecutive bursts, except for PTCCH. The mapping in frequency of PDCH onto the physical channel is defined in [5].

On PRACH and PTCCH/U, access bursts are used. On all other packet data logical channels, radio blocks comprising four normal bursts are used. The only exceptions are some messages on uplink PACCH that comprise four consecutive access bursts (to increase robustness). The following list shows some more details about how the packet data logical channels are mapped onto physical channels.

Packet Common Control Channel (PCCCH)

At a given time, the logical channels of the PCCCH are mapped on different physical resources than the logical channels of the CCCH. The PCCCH does not have to be allocated permanently in the cell. Whenever the PCCCH is not allocated, the CCCH shall be used to initiate a packet transfer. The PCCCH, when it exists, is mapped on one or several physical channels according to a 52-multiframe. In that case, the PCCCH, PBCCH and PDTCH share the same physical channels (PDCHs). The existence and location of the PCCCH shall be broadcast on the cell.

- *Packet random access channel (PRACH).* The PRACH is mapped on one or several physical channels. The physical channels on which the PRACH is mapped are derived by the MS from information broadcast on the PBCCH or BCCH.

  PRACH is determined by the uplink state flag (USF) being marked as free, which is broadcast continuously on the corresponding downlink. Additionally, a pre-defined fixed part of the multiframe structure for PDCH can be used as PRACH only and the information about the mapping on the physical channel is broadcast on PBCCH. During those time periods, an MS does not have to monitor the USF that is simultaneously broadcast on the downlink.
- *Packet paging channel (PPCH).* The PPCH is mapped on one or several physical channels. The exact mapping on each physical channel follows a pre-defined rule, as it is done for the paging channel (PCH).

  The physical channel on which the PPCH is mapped, as well as the rule that is followed on the physical channels, are derived by the MS from information broadcast on the PBCCH.
- *Packet access grant channel (PAGCH).* The PAGCH is mapped on one or several physical channels. The exact mapping on each physical channel follows a pre-defined rule.

  The physical channels on which the PAGCH is mapped, as well as the rule that is followed on the physical channels, are derived by the MS from information broadcast on the PBCCH.
- *Packet notification channel (PNCH).* The PNCH is mapped on one or several blocks on PCCCH. The exact mapping follows a pre-defined rule. The mapping is derived by the MS from information broadcast on the PBCCH.

Packet Broadcast Control Channel (PBCCH)

The PBCCH shall be mapped on one or several physical channels. The exact mapping on each physical channel follows a pre-defined rule, as it is done for the BCCH.

The existence of the PCCCH and, consequently, the existence of the PBCCH, is indicated on the BCCH.

Packet Timing Advance Control Channel (PTCCH)

Two defined frames of multiframe are used to carry PTCCH. The exact mapping of PTCCH/U sub-channels and PTCCH/D are defined in [5].

On PTCCH/U, access bursts are used. On PTCCH/D, four normal bursts comprising a radio block are used.

Packet Traffic Channels

- *Packet data traffic channel (PDTCH).* One PDTCH is mapped on to one physical channel. Up to eight PDTCHs, with different timeslots but with the same frequency parameters, may be allocated to one MS at the same time.

- *Packet associated control channel (PACCH).* PACCH is dynamically allocated on a block basis on the same physical channel that is carrying PDTCHs. However, one block PACCH allocation is used on the physical channel that is carrying only PCCCH when the MS is polled to acknowledge the initial assignment message.

  PACCH is of a bi-directional nature, i.e. it can dynamically be allocated both on the uplink and on the downlink, regardless of whether the corresponding PDTCH assignment is for uplink or downlink.

Different packet data logical channels can be multiplexed in one direction (either on the downlink or uplink) on the same physical channel (i.e. PDCH). The reader is advised to see details in [5]. The type of message that is indicated in the radio block header allows differentiation between the logical channels. Additionally, the MS identity allows differentiation between PDTCHs and PACCHs assigned to different MSs.

1.4.9.3 Radio Interface (Um)

Radio Resource Management Principles

A cell supporting GPRS may allocate resources on one or several physical channels in order to support the GPRS traffic [14]. Those physical channels (i.e. PDCHs), shared by the GPRS MSs, are taken from the common pool of physical channels available in the cell. The allocation of physical channels to circuit-switched services and GPRS can vary dynamically. Common control signalling required by GPRS in the initial phase of the packet transfer is conveyed on PCCCH (when allocated), or on CCCH. This allows capacity to be allocated specifically to GPRS in the cell only when a packet is to be transferred.

At least one PDCH, acting as a primary PDCH, accommodates packet common control channels that carry all the necessary control signalling for initiating packet transfer (i.e. PCCCH), whenever that signalling is not carried by the existing CCCH, as well as user data and dedicated signalling (i.e. PDTCH and PACCH). Other PDCHs, acting as secondary PDCHs, are used for user data transfer and for dedicated signalling.

The GPRS does not require permanently allocated PDCHs. The allocation of capacity for GPRS can be based on the needs for actual packet transfers ('capacity on demand' principle). The operator can as well decide to dedicate, permanently or temporarily, some physical resources (i.e. PDCHs) for GPRS traffic.

When the PDCHs are congested owing to the GPRS traffic load and more resources are available in the cell, the network can allocate more physical channels as PDCHs. However, the existence of PDCH(s) does not imply the existence of PCCCH. When no PCCCH is allocated in a cell, all GPRS attached MSs camp on the CCCH.

In response to a packet channel request sent on CCCH from the MS that wants to transmit GPRS packets, the network can assign resources on PDCH(s) for the uplink transfer. After the transfer, the MS returns to CCCH.

When PCCCH is allocated in a cell, all GPRS attached MSs camp on it. PCCCH can be allocated either as the result of the increased demand for packet data transfers or whenever there are enough available physical channels in a cell (to increase the QoS). The information about PCCCH is broadcast on BCCH. When the PCCCH capacity is inadequate, it is possible to allocate additional PCCCH resources on one or several PDCHs. If the network releases the last PCCCH, the MS performs cell re-selection.

The number of allocated PDCHs in a cell can be increased or decreased according to demand.

Multiframe Structure for PDCH

The mapping in time of the logical channels is defined by a multiframe structure. The multiframe structure for PDCH consists of 52 TDMA frames, divided into 12 blocks (of 4 frames), 2 idle frames and 2 frames used for the PTCCH according to Figure 1.16.

The mapping of logical channels onto the radio blocks is defined in this section by means of the ordered list of blocks (B0, B6, B3, B9, B1, B7, B4, B10, B2, B8, B5, B11).

The PDCH that contains PCCCH (if any) is indicated on BCCH. That PDCH is the only one that contains PBCCH blocks. On the downlink of this PDCH, the first block (B0) in the ordered list of blocks is used as PBCCH. If required, up to three more blocks on the same PDCH can be used as additional PBCCH. Any additional PDCH containing PCCCH is indicated on PBCCH.

On any PDCH with PCCCH (with or without PBCCH), up to the next 12 blocks in the ordered list of blocks are used for PAGCH, PNCH, PDTCH or PACCH in the downlink. The remaining blocks in the ordered list are used for PPCH, PAGCH, PNCH, PDTCH or PACCH in the downlink. In all cases, the actual usage of the blocks is indicated by the message type. On an uplink PDCH that contains PCCCH, all blocks in the multiframe can be used as PRACH, PDTCH or PACCH. Optionally, the first blocks in the ordered list of blocks can only be used as PRACH. The MS may choose to either ignore the USF (consider it as FREE) or use the USF to determine the PRACH in the same way as for the other blocks.

The mapping of channels on multiframes is controlled by several parameters broadcast on PBCCH. On a PDCH that does not contain PCCCH, all blocks can be used as PDTCH or PACCH. The actual usage is indicated by the message type. Two frames are used for PTCCH (see [5]) and the two idle frames as well as the PTCCH frames can be used by the MS for signal measurements and base station identification code (BSIC) identification.

An MS attached to GPRS shall not be required to monitor BCCH if a PBCCH exists. All system information relevant for GPRS and some information relevant for circuit-switched services (e.g. the access classes) shall in this case be broadcast on PBCCH. In order to facilitate the MS operation, the network is required to transmit certain types of packet system information (PSI) messages in specific multiframes and specific PBCCH blocks within the multiframes. The exact scheduling is defined in [5]. If no PCCCH is allocated, the MS camps on the CCCH and receives all system information on BCCH. In this case, any necessary GPRS-specific system information has to be broadcast on BCCH.

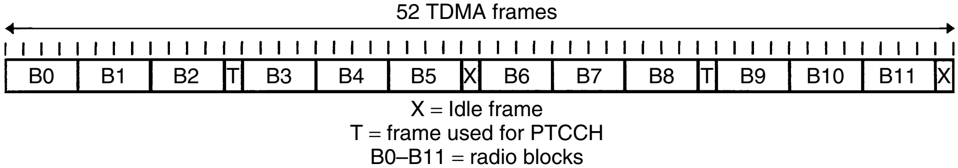


Figure 1.16. Multiframe structure for PDCH [14]

Radio Resource Operating Modes

Radio resource (RR) management procedures are characterised by two different RR operating modes [14], packet idle mode and packet transfer mode.

In *packet idle mode,* no temporary block flow (TBF) exists. Upper layers can require the transfer of a LLC PDU, which, implicitly, may trigger the establishment of TBF and transition to packet transfer mode.

In packet idle mode, the MS listens to the PBCCH and to the paging sub-channel for the paging group the MS belongs to. If PCCCH is not present in the cell, the MS listens to the BCCH and to the relevant paging sub-channels.

In *packet transfer mode*, the mobile station is allocated radio resource, providing a TBF on one or more physical channels. Continuous transfer of one or more LLC PDUs is possible. Concurrent TBFs may be established in opposite directions. Transfer of LLC PDUs in RLC acknowledged or RLC unacknowledged mode is provided.

When selecting a new cell, the MS leaves the packet transfer mode, enters the packet idle mode where it switches to the new cell, reads the system information and may then resume to packet transfer mode in the new cell.

The mobility management states are defined in [15]. Table 1.3 provides the correspondence between radio resource states and mobility management states.

Each state is protected by a timer. The timers run in the MS and the network. Packet transfer mode is guarded by RLC protocol timers.

The GPRS RR procedures and RR operating modes are defined in detail in [16].

Physical Link Layer

Different radio block structures are defined for the data transfer and for the control message transfer. The radio block consists of MAC header, RLC data block or RLC/MAC control block (Figure 1.17). It is always carried by four normal bursts. For detailed definition of radio block structure, see [17].

The MAC header (8 bits) contains control fields, which are different for uplink and downlink directions. The RLC header is of variable length and contains control fields, which are as well different for uplink and downlink directions. The RLC data field contains octets from one or more LLC packet data units (PDUs). The block check sequence (BCS) is used for error detection. The RLC/MAC control message field contains one RLC/MAC control message.

Channel Coding

Four coding schemes, CS-1 to CS-4, are defined for the packet data traffic channels. For all packet control channels (PCCHs) other than packet random access channel (PRACH)

Table 1.3. Correspondence between RR operating modes and MM states [14]

Radio resource BSS	Packet transfer mode	Measurement report reception	No state	No state
Radio resource MS	Packet transfer mode	Packet idle mode		Packet idle mode
Mobility management NSS and MS	Ready			Standby

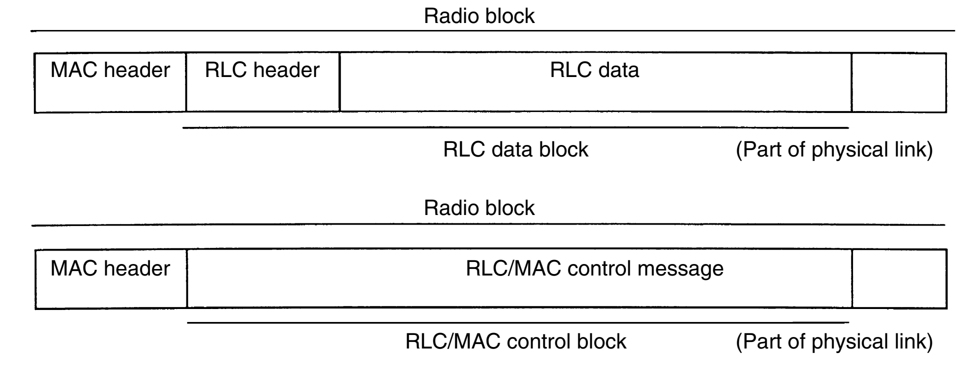


Figure 1.17. Radio block structures [14]

and packet timing advance control channel on uplink (PTCCH/U), coding scheme CS-1 is always used. For access bursts on PRACH, two coding schemes are specified. All coding schemes are mandatory for the MS. Only CS-1 is mandatory for the network.

Channel Coding for PDTCH

Four different coding schemes, CS-1 to CS-4, are defined for the radio blocks carrying RLC data blocks. The block structures of the coding schemes are shown in Figures 1.18 and 1.19.

The first step of the coding procedure is to add a BCS for error detection. For the CS-1 to CS-3, the second step consists of pre-coding USF (except for CS-1), adding four tail bits (TBs) and a half-rate convolutional coding for error correction that is punctured to give the desired coding rate. For the CS-4, there is no coding for error correction. The details of the codes are shown in Table 1.4, including the length of each field, the number of coded bits (after adding tail bits and convolutional coding), the number of punctured bits and the data rate (including the RLC header and RLC information).

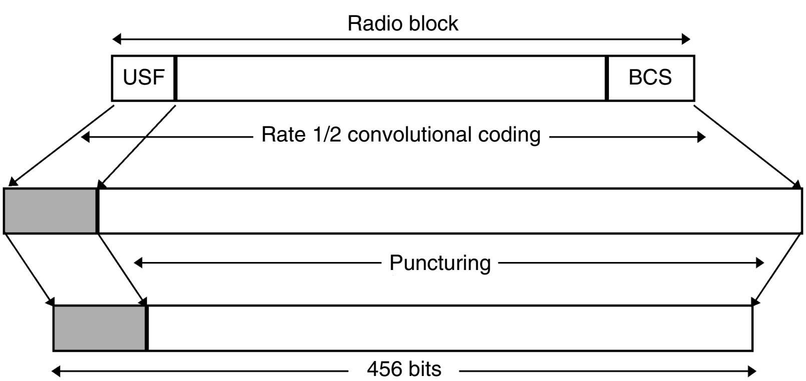


Figure 1.18. Radio block structure for CS-1 to CS-3

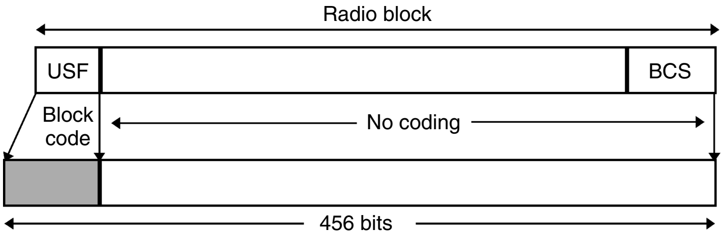


Figure 1.19. Radio block structure for CS-4 [14]

Table 1.4. Coding parameters for the coding schemes

Scheme	Code rate	USF	Pre-coded USF	Radio block excl. USF and BCS	BCS	Tail	Data rate (kbps)	Data rate kbps excl. RLC/MAC headers
CS-1	1/2	3	3	181	40	4	9.05	8
CS-2	∼2/3	3	6	268	16	4	13.4	12
CS-3	∼3/4	3	6	312	16	4	15.6	14.4
CS-4	1	3	12	428	16	–	21.4	20

The CS-1 is the same coding scheme as specified for the slow associated control channel (SACCH) in GSM 05.03. It consists of a half-rate convolutional code for FEC and a 40-bit FIRE code for the BCS. The CS-2 and CS-3 are punctured versions of the same half-rate convolutional code as CS-1. The CS-4 has no FEC. The CS-2 to CS-4 use the same 16-bit CRC for the BCS. The CRC is calculated over the whole uncoded RLC data block including the MAC header.

The USF has eight states, which are represented by a binary 3-bit field in the MAC header. For the CS-1, the whole radio block is convolutionally coded and the USF needs to be decoded as part of the data. All other coding schemes generate the same 12-bit code for the USF. The USF can be decoded either as a block code or as part of the data.

In order to simplify the decoding, the stealing bits (SB) (1-bit flag in either side of the training sequence code, as defined in [8]) of the block are used to indicate the actual coding scheme.

Channel Coding for PACCH, PBCCH, PAGCH, PPCH, PNCH and PTCCH

The channel coding for the PACCH, PBCCH, PAGCH, PPCH, PNCH and the downlink PTCCH is the same as the coding scheme CS-1. The coding scheme used for the uplink PTCCH is the same as for the PRACH.

Channel Coding for the PRACH

Two types of packet access bursts may be transmitted on the PRACH: an 8 information bits access burst or an 11 information bits access burst called the *extended packet access burst*. The reason for two different packet access bursts is that the 8-bit access burst cannot carry the additional information (USF) in the dynamic allocation mode. The mobile supports both access bursts. The channel coding for both burst formats is indicated as follows.

- *Coding of the 8 data bit packet access burst.* The channel coding used for the burst carrying the 8 data bit packet access uplink message is identical to the coding of the access burst as defined for the random access channel in [8].
- *Coding of the 11 data bit packet access burst.* The channel coding for the 11-bit access burst is the punctured version of the same coding as used for the 8-bit access burst.

Cell Re-selection

Initial PLMN selection and cell selection are done when the mobile station power is switched on. In GPRS packet, idle *cell re-selection* is performed autonomously by the MS. In GPRS packet transfer mode, the cell re-selection mechanism depends on the network operational mode. There are three network control modes: NC0, NC1 and NC2. In NC0, the MS performs the cell re-selection autonomously as in idle mode. In NC1, the MS also performs the cell re-selection autonomously but, in addition, it sends measurement reports to the network periodically. The measurement reports are analogous to the ones sent every 0.48 s in voice and CS data connections. In NC2, the MS sends periodical measurement reports and, additionally, the network sends cell re-selection commands to the MS, so the network has control of the cell re-selection process.

When the MS is attached to the circuit-switched core network (to MSC) and at the same time it is in packet transfer mode, the cell is determined by the network according to the handover procedures (handovers have precedence over GPRS cell re-selection).

In GPRS, two additional cell re-selection criteria, C31 and C32, are provided as a complement to the other GSM cell re-selection criteria. Cell re-selection criteria are specified in [4].

Also, a network controlled cell re-selection can be used in GPRS. The network can order mobile stations to send measurement reports to the network and to suspend its normal cell re-selection, and instead to accept decisions from the network. This applies to both packet idle mode and packet transfer mode. The MS measures the received RF signal strength on the BCCH frequencies of the serving cell and the neighbour cells as indicated in the BA-GPRS list and calculates the received level average (RLA) for each frequency, as defined in [4]. In addition, the MS verifies the BSIC of the cells. Only channels with the same BSIC as broadcast together with BA-GPRS on PBCCH are considered for re-selection.

The PBCCH broadcasts GPRS-specific cell re-selection parameters for serving and neighbour cells, including the BA-GPRS list. A BA-GPRS identifies the neighbour cells, including BSIC that need to be considered for the GPRS cell re-selection.

Timing Advance

The timing advance procedure is used to derive the correct value for timing advance that the MS has to use for the uplink transmission of radio blocks. The timing advance procedure comprises two parts:

- initial timing advance estimation
- continuous timing advance update.

The *initial timing advance estimation* is based on the single access burst carrying the packet channel request. The packet uplink assignment or packet downlink assignment then carries the estimated timing advance value to the MS. The MS uses this value for the uplink transmissions until the continuous timing advance update provides a new value (a few special cases for the initial timing advance are explained in [14]).

In the packet transfer mode, the MS uses the *continuous timing advance update* procedure. The continuous timing advance update procedure is carried on the PTCCH allocated to the MS. For the uplink packet transfer, within the packet uplink assignment, the MS is assigned the timing advance index (TAI) and the PTCCH. For the downlink packet transfer, within the packet downlink assignment, the MS is assigned the TAI and the PTCCH. The TAI specifies the PTCCH sub-channel used by the MS. On the uplink, the MS sends the assigned PTCCH access burst, which is used by the network to derive the timing advance.

The network analyses the received access burst and determines new timing advance values for all MSs performing the continuous timing advance update procedure on that PDCH. The new timing advance values are sent via a downlink signalling message (TA message) on PTCCH/D. The network can send timing advance information also in packet timing advance/power control and packet uplink Ack/Nack messages on PACCH. The mapping of the uplink access bursts and downlink TA messages on groups of eight 52-multiframes is shown in [14].

The BTS updates the timing advance values in the next TA message following the access burst.

Power Control Procedure

Power control is used in order to improve the spectrum efficiency and to reduce the power consumption in the MS. For the uplink, the MS follows a flexible power control algorithm, which the network can optimise through a set of parameters. It can be used for both open-loop and closed-loop power control. For the downlink, the power control is performed in the BTS. There is no need to specify the actual algorithms, but information about the downlink performance is needed. Therefore, the MSs transfer channel quality reports to the BTS. For the detailed specification of the power control see [4].

The MS shall calculate the RF output power value, P_{CH}, to be used on each individual uplink PDCH assigned to the MS [14]:

$$P_{\mathrm{CH}} = \min((\Gamma_0 - \Gamma_{\mathrm{CH}} - \alpha^*(C + 48), \mathrm{PMAX})$$

where

Γ_{CH} = an MS and channel-specific power control parameter. It is sent to the MS in any resource assigning message. The network can, at any time during a packet transfer, send new Γ_{CH} values to the MS on the downlink PACCH

Γ_0 = frequency band–dependent constant

$\alpha \in [0, 1]$ = a system parameter. Its default value is broadcast on the PBCCH. MS and channel-specific values can be sent to the MS together with Γ_{CH}

C = received signal level at the MS (see [14] for the derivation of this value)

$P\mathrm{MAX}$ = maximum allowed output power in the cell

All power values are expressed in dBm.

P_{CH} is not used to determine the output power when accessing the cell on PRACH or RACH, in which case PMAX is used.

The BTS uses constant power on those PDCH radio blocks that contain PBCCH or that may contain PPCH. This power may be lower than the output power used on BCCH. The difference is broadcast on PBCCH. On the other PDCH radio blocks, downlink power control can be used. Thus, a procedure can be implemented in the network to control the power of the downlink transmission on the basis of the channel quality reports. The network has to ensure that the output power is sufficient for the MS for which the RLC block is intended as well as the MS(s) for which the USF is intended, and that for each MS in packet transfer mode, at least one downlink RLC block per multiframe is transmitted with an output power that is sufficient for that MS, on a block monitored by that MS.

The MS has to periodically monitor the downlink Rx signal level and quality from its serving cell.

In order to derive the value of C, the MS periodically measures the received signal strength. In the packet idle mode, the MS measures the signal strength of the PCCCH or, if PCCCH does not exist, the BCCH. In the packet transfer mode, the MS measures the signal strength on BCCH. The same measurements as for cell re-selection are used. Alternatively, if indicated by a broadcast parameter, the MS measures the signal strength on one of the PDCHs where the MS receives PACCH. This method is suitable in the case in which BCCH is in another frequency band than the one used by PDCHs. It requires that constant output power is used on all downlink PDCH blocks.

MS Measurements

The MS measures the signal strength of each radio block monitored by the MS. The C value is achieved by filtering the signal strength with a running average filter. The filtering is usually continuous between the packet modes. The different filter parameters for the packet modes are broadcast on PBCCH or, if PBCCH does not exist, on BCCH. The variance of the received signal level within each block is also calculated. The filtered value SIGN_VAR is included in the channel quality report.

The channel quality is measured as the interference signal level during the idle frames of the multiframe, when the serving cell is not transmitting. In the packet transfer mode, the MS measures the interference signal strength of all eight channels (slots) on the same carrier as the assigned PDCHs. In the packet idle mode, the MS measures the interference signal strength on certain channels that are indicated on the PBCCH or, if PBCCH does not exist, on BCCH. Some of the idle frames and PTCCH frames are used for the channel quality measurements, while the others are required for BSIC identification and the timing advance procedure (see [4] for details).

The MS may not be capable of measuring all eight channels when allocated some configurations of channels. The MS has to measure as many channels as its allocation allows, taking into account its multislot capability. The interference, Γ_{CH}, is derived by filtering the measured interference in a running average filter.

In packet transfer mode, the MS transfers the eight Γ_{CH} values and the RXQUAL, SIGN_VAR and C values to the network in the channel quality report included in the PACKET DOWNLINK ACK/NACK message.

The BSS has to monitor the uplink Rx signal level and quality on each uplink PDCH, active as well as inactive. The BSS also has to measure the Rx signal level and the quality of a specific MS packet transfer.

Scheduling the MS activities during the PTCCH and idle frames. The MS uses the PTCCH and idle frames of the PDCH multiframe for the following tasks:

- Neighbour cell–level measurement and BSIC identification for cell re-selection
- Continuous timing advance procedures
- Interference measurements.

The scheduling of these tasks is not specified in detail. During the frames when the MS receives TA messages, it can also make interference measurements. Additionally, during the frames when the MS transmits access bursts, it may also be possible to make measurements on some channels.

The MS tries to schedule the BSIC identification as efficiently as possible, using the remaining PTCCH frames and the idle frames and also considering the requirements for interference measurements.

Discontinuous reception (DRX). The MS has to support discontinuous reception (DRX; sleep mode) in packet idle mode. The DRX is independent from mobility management (MM) states, ready and standby. The negotiation of the DRX parameters is per MS. The DRX parameters and the overview of DRX operation are explained in [14].

Medium Access Control and Radio Link Control Layer

The medium access control and radio link control layer operates above the physical link layer in the protocol architecture. The MAC function defines the procedures that enable multiple MSs to share a common transmission medium, which may consist of several physical channels. The RLC function defines the procedures for a selective re-transmission of unsuccessfully delivered RLC data blocks. MAC/RLC is a complex protocol, and for details the reader is referred to [17]. This section gives only an introduction of the main functions of the RLC/MAC layer.

The basic principle of data transfer is illustrated in Figure 1.20. The network protocol data units (N-PDU), which correspond to full IP packets, are compressed and segmented into the sub-network protocol data units (SN-PDU) by the sub-network-dependent convergence protocol (SNDCP). IP-packet compression is optional. The SN-PDUs are encapsulated into one or several LLC frames. The size of the data part of the LLC frames is a parameter between 140 and 1520 bytes. LLC frames are segmented into RLC data blocks. At the RLC/MAC layer, a selective automatic repeat request (ARQ) protocol (including block numbering) between the MS and the network provides re-transmission of erroneous RLC data blocks. When a complete LLC frame is successfully transferred across the RLC layer, it is forwarded to the LLC layer.

There are two important concepts that constitute the core of RLC/MAC operation: temporary block flow and temporary flow identifier.

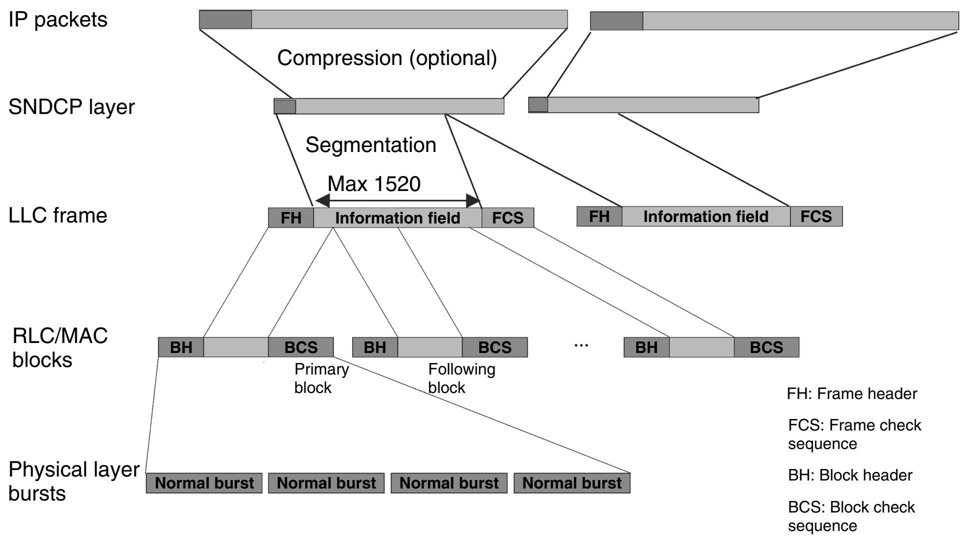


Figure 1.20. Transmission and reception data flow

Temporary Block Flow

A temporary block flow (TBF) [14] is a temporal connection between the MS and the network to support the uni-directional transfer of LLC PDUs on packet data physical channels. A TBF can use radio resources on one or more PDCHs and comprises a number of RLC/MAC blocks carrying one or more LLC PDUs. A TBF is temporary and is maintained only for the duration of the data transfer.

Temporary Flow Identity

Each TBF is assigned a temporary flow identity (TFI) [14] by the network. The assigned TFI is unique among concurrent TBFs in each direction and is used instead of the MS identity in the RLC/MAC layer. The same TFI value may be used concurrently for TBFs in opposite directions. The TFI is assigned in a resource assignment message that precedes the transfer of LLC frames belonging to one TBF to/from the MS. The same TFI is included in every RLC header belonging to a particular TBF, as well as in the control messages associated with the LLC frame transfer (e.g. acknowledgements) in order to address the peer RLC entities.

Uplink Radio Blocks Multiplexing

In uplink, the packet-switched users are multiplexed in the same PDCH by the MAC layer. There are three access modes supported by RLC/MAC protocol: dynamic allocation, extended dynamic allocation and fixed allocation.

Dynamic allocation. In order to make the multiplexing of mobile stations on the same PDCH possible, the uplink state flag (USF) is introduced. During the establishment of the uplink TBF, a USF is assigned to each mobile, which will be used by the network to indicate which terminal is allowed to transmit in the following uplink radio block.

With this method, each downlink block has to be decoded by all the MSs assigned to the PDCH in order to obtain the USF and the TFI, identifying the owner of the data or control information. In addition, even if no downlink information is available, a dummy radio block must be sent just to transmit the USF when required.

The USF is a 3-bit flag, so each PDCH can manage up to eight MSs, except on the PCCH, where the value '111' (USF = Free) indicates that the corresponding uplink radio block is reserved for the packet random access channel (PRACH) [14]. There are only 16 TA indexes for the continuous timing advance procedure to keep the MS synchronised with the TDMA frame at the BTS. If the maximum number of TBF multiplexed in UL is 7, the maximum number of TBF in DL is limited to 9 in the extreme case that the 16 TBF (7 in UL and 9 in DL) multiplexed on the same PDCH belongs to different MSs.

A special channel coding is used for the USF added to the different coding schemes that makes it quite reliable (even more than CS-1) and prevents uplink radio block losses. For multislot classes, the mobile must decode the USF in all PDCHs separately.

Whenever the MS detects an assigned USF value on an assigned PDCH, the MS shall transmit either a single RLC/MAC block or a sequence of four RLC/MAC blocks on the same PDCH. The number of RLC/MAC blocks to transmit is controlled by the USF_GRANULARITY parameter characterising the uplink TBF [14].

The fact that the MS has to receive a USF on each timeslot (TSL) introduces limitations on the maximum number of TSLs supported in the UL direction for certain MS classes. In most practical cases, with the simplest (E)GPRS radio implementation, the maximum number of TSLs in UL is limited to two. This can be seen in Figure 1.21 with a 3 + 2 MS allocation. Note that this limitation is very dependent on MS class.

Extended dynamic allocation. This procedure allows a higher number of TSLs in UL and eliminates the need of receiving USF on each TSL (which may create too many dummy blocks in DL). In this case, when a USF is received in a particular TSL, the MS is allowed to transmit on the same TSL and all the subsequent TSLs. Let us consider the allocation in Figure 1.22 where a 2 + 3 MS allocation is considered. When a USF is received on TSL 1, the MS will transmit on TSL 1 and 2. If simple dynamic allocation would be used, the MS would transmit only on TSL 1.

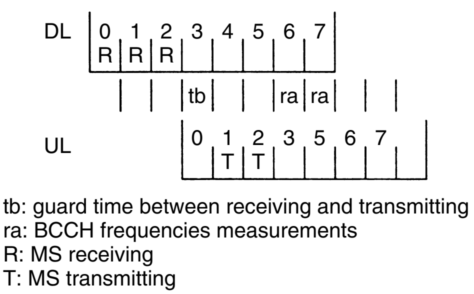


Figure 1.21. Dynamic UL allocation example

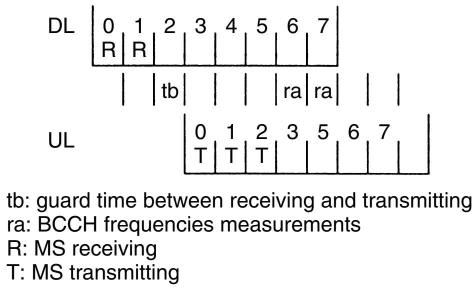


Figure 1.22. Extended dynamic UL allocation example

Fixed allocation. Using this procedure, the network communicates with the MS and UL transmission turn pattern by using a specific control message. In this case, USF is not utilised and, therefore, DL communication is point-to-point. This latter characteristic makes it more suitable for link optimisation techniques like DL power control and smart antennas. Once an MS has requested a resource assignment for the uplink, the network responds with a bitmap (ALLOCATION_BITMAP), indicating timeslots and radio blocks where the MS is allowed to transmit after a frame number contained in the parameter *TBF starting time*.

Time is slotted in *allocation periods* with a configurable duration that will be the basic time unit to assign resources for an uplink TBF. If more than one allocation is needed for the MS initial request, the network will send new bitmaps in packet ACK/NACK messages at the end of the current allocation period.

Modes for RLC/MAC Operation

In addition, there are two distinctive modes of operation of RLC/MAC layer:

- unacknowledged operation
- acknowledged operation.

Acknowledged mode for RLC/MAC operation. The transfer of RLC data blocks in the acknowledged RLC/MAC mode is controlled by a selective ARQ mechanism coupled with the numbering of the RLC data blocks within one temporary block flow. The sending side (the MS or the network) transmits blocks within a window and the receiving side sends packet uplink Ack/Nack or packet downlink Ack/Nack message when needed. Every such message acknowledges all correctly received RLC data blocks up to an indicated block sequence number (BSN). Additionally, the bitmap is used to selectively request erroneously received RLC data blocks for re-transmission.

Unacknowledged mode for RLC/MAC operation. The transfer of RLC data blocks in the unacknowledged RLC/MAC mode is controlled by the numbering of the RLC data blocks within one TBF and does not include any re-transmissions. The receiving side extracts user data from the received RLC data blocks and attempts to preserve the user information length by replacing missing RLC data blocks by dummy information bits.

Mobile-originated Packet Transfer

Uplink access. An MS initiates a packet transfer by making a packet channel request on PRACH or RACH, depending on the availability of PCCHs in the network. The network responds on PAGCH (if PCCHs are supported by the network) or access grant channel (AGCH), respectively. It is possible to use one- or two-phase packet access methods (see Figure 1.23).

In the one-phase access, the network will respond to the packet channel request by sending a packet uplink assignment message and reserving the resources on PDCH(s) for uplink transfer of a number of radio blocks. The reservation is done according to the requested resources indicated in the packet channel request. On RACH, there are only two cause values available for denoting GPRS, which can be used to request limited resources or two-phase access. On PRACH, the packet channel request may contain more adequate information about the requested resources, and, consequently, uplink resources on one or more PDCHs can be assigned by using the packet uplink assignment message.

In the two-phase access, the packet channel request is responded to with the packet uplink assignment, which reserves the uplink resources for transmitting the packet resource request. A two-phase access can be initiated by the network or an MS. The network can order the MS to send a packet resource request message by setting a parameter in packet uplink assignment message. The MS can require two-phase access in the packet channel request message. In this case, the network may order the MS to send a packet resource request or continue with a one-phase access procedure.

The packet resource request message carries the complete description of the requested resources for the uplink transfer. The MS can indicate the preferred medium access method to be used during the TBF. The network responds with the packet uplink assignment, reserving resources for the uplink transfer and defining the actual parameters for data transfer (e.g. medium access mode). If there is no response to the packet channel request within the pre-defined time period, the MS makes a retry after a random time.

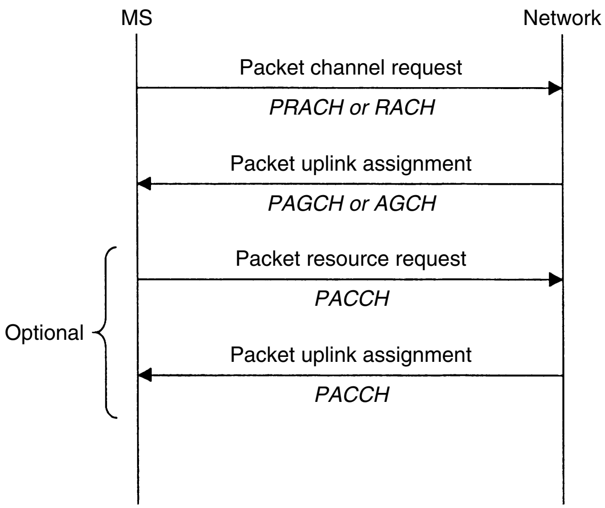


Figure 1.23. Access and allocation for the one or two-phase packet access, uplink packet transfer

On PRACH, a two-step approach is used and includes a long-term and a short-term estimation of the persistence [14]. The optimal persistence of the mobile stations is calculated at the network side.

The actual persistence values depend on

- the priority of the packet to be transmitted;
- the amount of traffic within higher priority classes;
- the amount of traffic within its own priority class.

Occasionally, more packet channel requests can be received than can be served. To handle this, a packet queuing notification is transmitted to the sender of the packet channel request. The notification includes information that the packet channel request message is correctly received and that the packet uplink assignment may be transmitted later. If the timing advance information becomes inaccurate for an MS, the network can send packet polling request to trigger the MS to send four random access bursts. This can be used to estimate the new timing advance before issuing the packet uplink assignment [14].

In case of dynamic or extended dynamic allocation, the packet uplink assignment message includes the list of PDCHs and the corresponding USF value per PDCH. A unique TFI is allocated and is thereafter included in each RLC data and control block related to that temporary block flow. The MS monitors the USFs on the allocated PDCHs and transmits radio blocks when the USF value indicates to do so. In case of fixed allocation, the packet uplink assignment message is used to communicate a detailed fixed uplink resource allocation to the MS. The fixed allocation consists of a start frame, slot assignment and block assignment bitmap representing the assigned blocks per timeslot [14]. The MS will wait until the start frame and then transmit radio blocks on those blocks indicated in the block assignment bitmap. The fixed allocation does not include the USF and the MS is free to transmit on the uplink without monitoring the downlink for the USF. If the current allocation is not sufficient, the MS may request additional resources in one of the assigned uplink blocks. A unique TFI is allocated and is thereafter included in each RLC data and control block related to that temporary block flow. Because each radio block includes an identifier (TFI), all received radio blocks are correctly associated with a particular LLC frame and a particular MS.

Contention Resolution

Contention resolution is an important part of RLC/MAC protocol operation, especially because one channel allocation can be used to transfer a number of LLC frames. There are two basic access possibilities, one-phase and two-phase access as seen in Figure 1.23. The two-phase access eliminates the possibility that two MSs can perceive the same channel allocation as their own. Basically the second phase of access will uniquely identify the MS by TLLI and that TLLI will be included in packet uplink assignment, therefore ruling out the possibility for a mistake.

Mobile Terminated Packet Transfer

Packet paging. The network initiates a packet transfer to an MS that is in the standby state by sending one or more packet paging request messages on the downlink (PPCH or PCH).

The MS responds to one packet paging request message by initiating a mobile-originated packet transfer. This mobile-originated packet transfer allows the MS to send a packet paging response message containing an arbitrary LLC frame. The message sequence described in Figure 1.24 is conveyed either on PCCCH or on CCCH. After the packet paging response is sent by the MS and received by the network, the mobility management state of the MS is ready.

The network can then assign some radio resources to the MS and perform the downlink data transfer.

Downlink packet transfer. The transmission of a packet to an MS in the ready state is initiated by the network using a packet downlink assignment message. In case there is an uplink packet transfer in progress, the packet downlink assignment message is transmitted on PACCH. Otherwise, it is transmitted in the PCCCH, in case there is one allocated in the cell. If that is not the case, it will be transmitted in the CCCH. The packet downlink assignment message includes the list of PDCH(s) that will be used for downlink transfer.

The network sends the RLC/MAC blocks belonging to one TBF on downlink on the assigned downlink channels. Multiplexing the RLC/MAC blocks addressed for different MSs on the same PDCH downlink is enabled with the TFI identifier, included in each RLC/MAC block [14].

The sending of the packet downlink Ack/Nack message is obtained through the periodical network-initiated polling of the MS. The MS sends the packet downlink Ack/Nack message in a reserved radio block, which is allocated together with polling.

Release of the resources. The release of the resources is initiated by the network by terminating the downlink transfer and polling the MS for a final packet downlink Ack/Nack message.

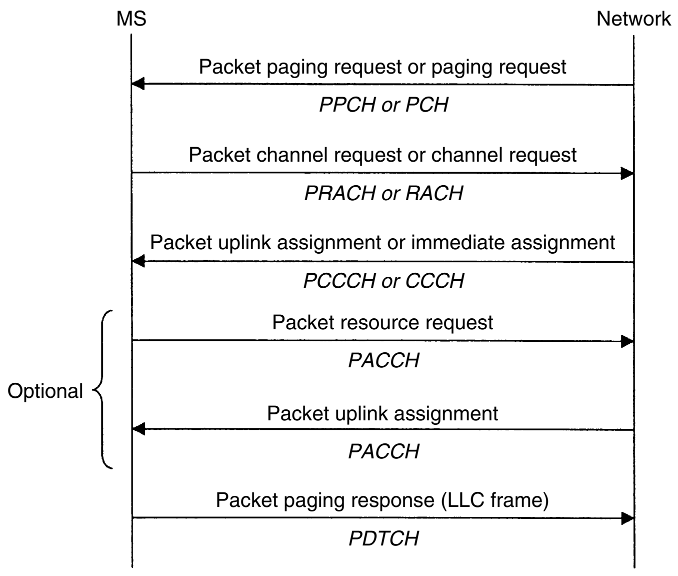


Figure 1.24. Paging message sequence for paging, downlink packet transfer

Simultaneous uplink and downlink packet transfer. During the ongoing uplink TBF, the MS continuously monitors one downlink PDCH for possible occurrences of packet downlink assignment or packet timeslot reconfigure messages on PACCH. The MS is therefore reachable for downlink packet transfers that can then be conveyed simultaneously on the PDCH(s) that respects the MS multislot capability.

If the MS wants to send packets to the network during the ongoing downlink TBF, it can be indicated in the acknowledgement that is sent from the MS. By doing so, no explicit packet channel requests have to be sent to the network. Furthermore, the network already has the knowledge of which PDCH(s) that particular MS is currently using so that the uplink resources can be assigned on the PDCH(s) that respect the MS multislot capability.

1.5 EDGE Rel'99

Enhanced data rates for GSM evolution (EDGE) is a major enhancement to the GSM data rates. GSM networks have already offered advanced data services, like circuit-switched 9.6-kbps data service and SMS, for some time. High-speed circuit-switched data (HSCSD), with multislot capability and the simultaneous introduction of 14.4-kbps per timeslot data, and GPRS are both major improvements, increasing the available data rates from 9.6 kbps up to 64 kbps (HSCSD) and 160 kbps (GPRS).

EDGE is specified in a way that will enhance the throughput per timeslot for both HSCSD and GPRS. The enhancement of HSCSD is called *ECSD* (enhanced circuit-switched data), whereas the enhancement of GPRS is called *EGPRS* (enhanced general packet radio service). In ECSD, the maximum data rate will not increase from 64 kbps because of the restrictions in the A-interface, but the data rate per timeslot will triple. Similarly, in EGPRS, the data rate per timeslot will triple and the peak throughput, with all eight timeslots in the radio interface, will reach 473 kbps.

1.5.1 8-PSK Modulation in GSM/EDGE Standard

The 'enhancement' behind tripling the data rates is the introduction of the 8-PSK (octagonal phase shift keying) modulation in addition to the existing Gaussian minimum shift keying. An 8-PSK signal is able to carry 3 bits per modulated symbol over the radio path, while a GMSK signal carries only 1 bit per symbol (Table 1.5). The carrier symbol rate (270.833 kbps) of standard GSM is kept the same for 8-PSK, and the same pulse shape as used in GMSK is applied to 8-PSK. The increase in data throughput does not come for free, the price being paid in the decreased sensitivity of the 8-PSK signal. This affects, e.g. the radio network planning, and the highest data rates can only be provided with limited coverage. The GMSK spectrum mask was the starting point for the spectrum mask of the 8-PSK signal, but along the standardisation process, the 8-PSK spectrum mask was relaxed with a few dB in the 400 kHz offset from the centre frequency [18]. This was found to be a good compromise between the linearity requirements of the 8-PSK signal and the overall radio network performance.

It was understood already at the selection phase of the new modulation method that a linear modulation would have quite different characteristics than the GMSK as a constant envelope modulation. A practical problem is how to define the receiver performance requirements for the physical channels with 8-PSK modulation, in particular, how to incorporate the phenomena resulting from the non-idealities in the transmitter and receiver. This

Table 1.5. Key physical layer parameters for GSM/EDGE

	8-PSK	GMSK
Symbol rate	270.833 kbps	270.833 kbps
Number of bits/symbol	3 bits/symbol	1 bit/symbol
Payload/burst	342 bits	114 bits
Gross rate/timeslot	68.4 kbps	22.8 kbps

problem is solved in the specification by using the concept of error vector of magnitude (EVM). The EVM effectively measures how far an observed signal sample is from the ideal signal constellation point (Figure 1.25). Thus the EVM takes into account the deviation in both I- and Q-axes, not explicitly distinguishing whether the signal impairment is due to the phase noise or the amplitude distortion. For the 8-PSK signal, [18] sets specific EVM percentage requirements for the transmitter and receiver performance.

A further peculiarity, introduced by bringing the second modulation method, 8-PSK, to GSM standard, is the need in downlink to blindly recognise the transmitted modulation method in the mobile station receiver. This is due to the characteristics of the EGPRS link quality control (LQC), where the used modulation and coding scheme (MCS) is adjusted according to the channel conditions to the most suitable one, and in downlink, no prior information is sent to the receiver but the receiver is able to find out the used MCS on the basis of (1) the blind modulation identification and (2) the decoding of the RLC/MAC header field that contains indication of the coding and puncturing scheme. The modulation identification is based on the different phase rotation characteristics in the GMSK and 8-PSK training sequences. In the GMSK training sequence, the symbol-by-symbol phase rotation is p/2, whereas in the 8-PSK training sequence, the rotation is 3p/8. Otherwise, the set of 8-PSK training sequences has identical information content (the same 26-bit sequence) as the GMSK training sequences.

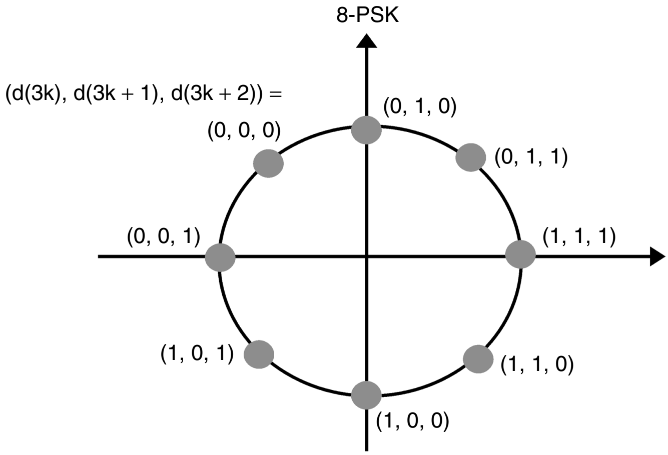


Figure 1.25. 8-PSK signal constellation

1.5.2 Enhanced General Packet Radio Service (EGPRS)

EGPRS is built on top of GPRS, which is the packet-switched data service of GSM. Examples of typical high bit rate packet services include fast file transfer, Internet service access, web browsing and remote email.

EGPRS has a major impact on the RF and the physical layer of the radio interface as well as on RLC/MAC protocol, but the changes to other protocols and protocol layers are minor. The EDGE RF specification and the definition of the burst structures are common to both EGPRS and ECSD. One large conceptual modification in EGPRS, compared with GPRS, is the link quality control which, in EGPRS, also supports incremental redundancy (type II hybrid ARQ), in addition to GPRS-type link adaptation mode (type I hybrid ARQ). The LQC includes nine different modulation and coding schemes (MCS-1–MCS-9) as well as related signalling and other procedures for link adaptation (switching between different MCSs).

Table 1.6 shows the EGPRS modulation and coding schemes (MCS) and their data throughputs. New GMSK coding schemes (MCS-1–MCS-4), different from GPRS GMSK coding schemes (CS-1–CS-4), are needed because of the incremental redundancy support. Figure 1.26 illustrates the coding and puncturing principle of the header and payload part of an MCS-9 radio block. In EGPRS radio blocks, the stealing bits indicate the header format. The RLC/MAC header is strongly coded (close to rate 1/3 code in 8-PSK modes and close to rate 1/2 code in GMSK modes) to allow reliable decoding of the header in incremental redundancy (IR) operation. There is also a separate header check sequence (HCS) for the error detection in the header part. The RLC data payload is encoded along with the final block indicator (FBI) and extension bit (E) fields. Also, the block check sequence (BCS) and tail bits (TB) are added before encoding. In the illustrated case (MCS-9), two RLC data blocks are mapped on one radio block. Note that in the case of MCS-8 and MCS-9, an RLC data block is interleaved over two bursts only. The RLC/MAC header is though interleaved over the whole radio block (i.e. four bursts).

EGPRS modulation and coding schemes are organised in families according to their RLC data block sizes. For example, the MCS-9 carries 1184 payload bits in two RLC data blocks (592 payload bits in each of them) during one EGPRS radio block of four consequent bursts. The MCS-6 carries 592 payload bits in one RLC data block and the MCS-3 carries 296 payload bits within the radio block. The number of payload bits in

Table 1.6. EGPRS modulation and coding schemes

Modulation and coding scheme	Code rate	Modulation	Data rate/ timeslot (kbps)	Family
MCS-9	1.0	8-PSK	59.2	A
MCS-8	0.92		54.4	A
MCS-7	0.76		44.8	B
MCS-6	0.49		29.6	A
MCS-5	0.37		22.4	B
MCS-4	1.0	GMSK	17.6	C
MCS-3	0.80		14.8	A
MCS-2	0.66		11.2	B
MCS-1	0.53		8.8	C

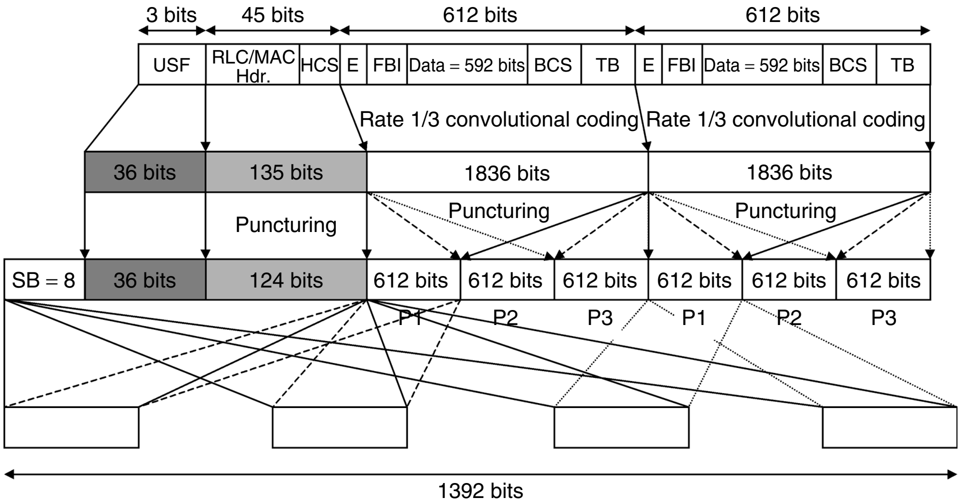


Figure 1.26. EGPRS coding and puncturing example (MCS-9: uncoded 8-PSK, two RLC blocks per 20 ms) [14]

MCS-6 and MCS-3 being sub-multiples of the higher coding schemes in the same family (family A) enables effective re-transmission of the negatively acknowledged RLC data blocks with the lower coding schemes within the family, if needed. A typical case where this functionality is useful is the sudden change in the channel conditions.

The transfer of RLC data blocks in the acknowledged RLC/MAC mode can be controlled by a selective type I ARQ mechanism or by type II hybrid ARQ (IR) mechanism, coupled with the numbering of the RLC data blocks within one temporary block flow. The sending side (the MS or the network) transmits blocks within a window and the receiving side sends packet uplink Ack/Nack or packet downlink Ack/Nack message when needed.

The initial MCS for an RLC block is selected according to the link quality. For the re-transmissions, the same or another MCS from the same family of MCSs can be selected. For example, if MCS-7 is selected for the first transmission of an RLC block, any MCS of the family B can be used for the re-transmissions. The selection of MCS is controlled by the network.

In the EGPRS type II hybrid ARQ scheme, the information is first sent with one of the initial code rates (i.e. rate 1/3 encoded data are punctured with PS 1 of the selected MCS). If the RLC data block is received in error, additional coded bits (i.e. the output of the rate 1/3 encoded data that is punctured with PS 2 of the prevailing MCS) are sent and decoded together with the already received code words until decoding succeeds. If all the code words (different punctured versions of the encoded data block) have been sent, the first code word (which is punctured with PS 1) is sent. Alternatively, it is possible to use incremental redundancy modes called *MCS-5–7* and *MCS-6–9*, in which the initial transmissions are sent with either MCS-5 or MCS-6 (respectively) and the re-transmissions are sent with MCS-7 or MCS-9 (respectively). In the EGPRS type I ARQ, the operation is similar to that of the EGPRS type II hybrid ARQ, except that the decoding of an RLC data block is solely based on the prevailing transmission (i.e. erroneous blocks are not stored). The motivation for supporting the incremental redundancy mode is the improved data throughput and the robustness for varying channel conditions and measurement errors. Incremental redundancy is defined to be mandatory for the MS and optional for the network.

Another RLC/MAC layer modification in Rel'99 due to EGPRS is the increased RLC window size. In GPRS with four-burst radio blocks (20 ms), the RLC window size of 64 is defined. With the higher coding schemes of EGPRS, MCS-7 to MCS-9, there are two RLC blocks per 20 ms radio block, which makes the RLC window size 64 too small and RLC protocol subject to stalling. Therefore, the EGPRS RLC window size was increased to 128, and a compression method was defined for the acknowledgement bitmap of the RLC blocks (see details in GSM 04.60).

1.5.2.1 Bit Error Probability (BEP) Measurements

In Rel'99, a generic enhancement to the channel quality measurements was also introduced. Earlier, the measure to be used for the channel quality indication was the RX_QUAL, which is an estimate of the pseudo bit error rate, i.e. computed by comparing the received bit sequence (before decoding) with the encoded version of the decoded bit sequence and counting the number of bit errors. It was shown in [19] that the alternative measure—bit error probability (BEP)—better reflects the channel quality in varying channel conditions—including frequency hopping and varying mobile speed. BEP is estimated burst by burst, for example, from the soft output of the receiver. The detailed realisation of the measurement itself is left for the equipment vendors, whereas the method of reporting the BEP measurements and the accuracy requirements are standardised. The measurements to be reported by the MS for the network are MEAN_BEP and CV_BEP, where the MEAN_BEP is the average of the block wise mean BEP during the reporting period and the CV_BEP is the average of the blockwise coefficient of variation (CV) (CV = Std(BEP)/Mean(BEP)) during the reporting period.

1.5.2.2 Adjustable Filtering Length for the EGPRS Channel Quality Measurements

A further standardised means of optimising the EGPRS performance is the filtering length of the LQC measurements, which can be varied [20]. The network and the MS can negotiate with each other—using the radio link control signalling—the optimum length of the filter that is used to smooth the BEP measurements that the MS sends to the network. For example, typically the filter length should be shorter for the MS with higher speed, so that the received channel quality information at the network would be as correct as possible. On the other hand, for slowly moving mobiles, the longer averaging of the measurements improves the accuracy.

1.5.3 Enhanced Circuit-switched Data (ECSD)

ECSD uses current HSCSD as a basis. The user data rates are not increased compared to HSCSD (up to 64 kbps), but these rates can be achieved with smaller numbers of timeslots and simpler MS implementation. Data rates to be provided with ECSD, although limited to 64 kbps, are still sufficient for providing various transparent and non-transparent services. ECSD enables inter-working with audio modems at higher data rates than in current GSM

Table 1.7. ECSD data rates

Coding scheme	Code rate	Modulation	Gross rate (kbps)	Radio interface rate (kbps)	User rates (kbps)
TCH/28.8 (NT/T)	0.419	8-PSK	69.2	29.0	28.8
TCH/32.0 (T)	0.462	8-PSK	69.2	32.0	32.0
TCH/43.2 (NT)	0.629	8-PSK	69.2	43.5	43.2

networks, inter-working with ISDN at various data rates and various video-based services ranging from still image transfer to videoconferencing services.

Higher data rates are defined for both transparent and non-transparent services. Radio interface rates (and user rates per timeslot) for ECSD are shown in Table 1.7. In the definition of the ECSD, only a few new traffic channels are defined. The control channels—both the common, dedicated and associated control channels—are the same as for the other circuit-switched services (like, e.g. for the 9.6 kbps circuit-switched data, HSCSD) except the fast associated control channel for the ECSD (FACCH/E), which uses GMSK modulation and is interleaved rectangularly on the four consequent TDMA frames. Further, the detection of the FACCH/E frame is based on the similar principles as the identification of the 8-PSK and GMSK modulated data blocks in EGPRS, i.e. the modulation method is blindly detected in the receiver, and if a block of four GMSK bursts is received in the middle of the 8-PSK data frames, the receiver knows that an FACCH/E frame was received.

The 28.8-kbps data rate is available for transparent and non-transparent services for both single and multislot configurations. The 32.0-kbps service is available only in multislot configuration (two-slot) and can be used for offering 64-kbps transparent service, while 43.2 kbps is available for non-transparent services only.

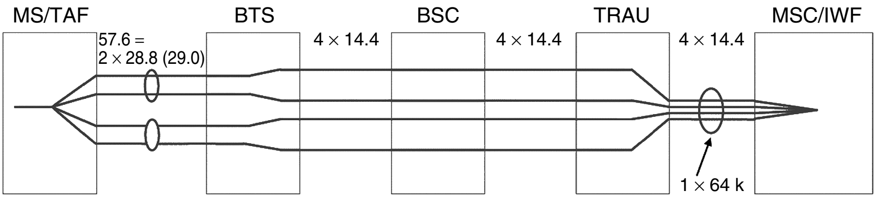
The ECSD architecture is largely based on HSCSD transmission and signalling. This ensures a minimum impact on existing specifications. The basic principle is to use the same transcoder and rate adaptation unit (TRAU) frame formats and multiple 16-kbps sub-channels in the network side. For example, 28.8-kbps service provided with one radio interface timeslot is supported with two 16-kbps sub-channels and 14.4-kbps TRAU frames on the network side.

Figure 1.27 shows the network architecture for providing (a) 57.6-kbps non-transparent service and (b) 56-/64-kbps transparent service.

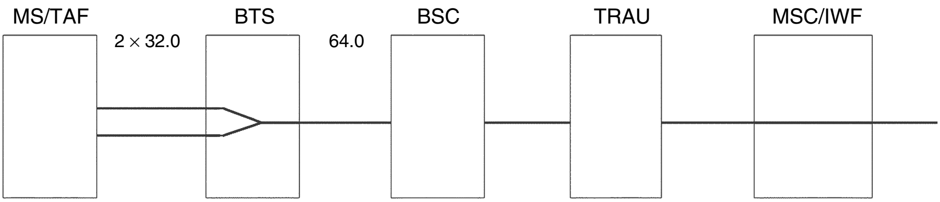
ECSD 57.6-kbps non-transparent service is provided using two timeslots of 28.8 kbps as shown in Figure 1.27(a). The same frame formats as in HSCSD are used end to end between the MS/TAF and MSC/IWF. The BTS is responsible for the de-multiplexing of data between two radio timeslots and four Abis sub-channels.

The network architecture for supporting bit transparent 56-/64-kbps service is shown in Figure 1.27(b). It is provided using two air interface timeslots of 32 kbps. The rate adaptation functions are located in the BTS.

Higher data rates can be offered with limited coverage and, therefore, the link adaptation mechanism becomes essential. Switching between 8-PSK- and GMSK-based channels is done with the standard intra-cell handover procedure. Signalling for link adaptation is based on the existing HSCSD signalling mechanism.



(a) 57.6 kbps non-transparent service



(b) 56.0/64.0 kbps transparent service

Figure 1.27. Network architecture

1.5.4 Class A Dual Transfer Mode (DTM)

With a Class A mobile station, a subscriber can simultaneously connect to the circuit-switched core network (to mobile switching centre (MSC)) and to the packet-switched core network (to SGSN), through A and Gb interfaces respectively. Until recently, the implementation of a Class A MS has been considered impractical and costly—owing to the uncoordination of the radio resources for the CS and PS connections, which implies duplicating many key parts of an MS like the transceiver. To enable practical low cost implementation of the Class A mobile, the Class A dual transfer mode (DTM) was defined. This functionality was deemed necessary in the Rel'99 specification—at the same schedule with the first UTRAN standard release (Rel'99) and with the inter-working between GSM and UMTS (e.g. inter-system handovers). This package of specifications ensures that an operator can provide the same type of service—e.g. simultaneous voice and data—in both radio access networks, GSM and UTRAN.

The DTM concept includes the following:

- *The single timeslot method.* TCH/H + PDTCH/H. In this method, AMR (adaptive multi-rate speech codec) voice frames are carried in the half-rate circuit-switched channel (TCH/H) and the user data in the half-rate packet-switched channel (PDTCH/H). The significance of the single timeslot method is that it enables implementation of the Class A DTM feature even in single timeslot MSs. Another advantage is that in case of a handover, the availability of the radio resources in the target cell can be easily guaranteed.
- *The multiple timeslot method.* TCH + one or more adjacent PDTCH. In this method, one full timeslot is reserved for the AMR voice frames and the adjacent timeslots are used for the user data transfer.

A further significant difference between the single and multiple timeslot methods is that in the former one, the PDTCH/H resource is always dedicated for the DTM user (dedicated mode of (E)GPRS MAC protocol), whereas in the latter one, the PDTCH resources can be shared (shared mode of (E)GPRS MAC protocol).

More details of Class A DTM can be found, e.g. in [21, 22].

1.5.5 EDGE Compact

Compact is a particular mode—or variant—of EGPRS, which is designed for the deployment in the narrow frequency band allocations—the minimum requirement being the deployment in less than 1 MHz of spectrum. It is known that the capacity (spectral efficiency) of the GSM network is limited by the BCCH reuse, and as available bandwidth decreases, its relative impact will be higher. This is due to the nature of the broadcast control channel concept in the GSM standard, which requires continuous transmission with constant transmit power in the beacon frequency (BCCH frequency). This causes the BCCH frequency reuse to be relatively large to ensure reliable reception of the system information in the broadcast and common control channels.

The key component of the compact mode is the new control channel structure, which is based on the discontinuous time rotating BCCH, which enables significantly lower reuse factors for the BCCH channel and thus operation in the narrow spectrum allocation. The characteristics of the compact mode are the following:

- It is a stand-alone high-speed packet data system (EGPRS variant with alternative control channel structure).
- It can be deployed in only 600 kHz of spectrum (+ guardband) by using three carriers in a 1/3 reuse pattern.
- It requires inter–base station time synchronisation.
- It uses discontinuous transmission on the control carriers and a new logical control channel combination (compact packet broadcast control channel (CPBCCH) based on a standard 52-multiframe). It uses timeslot mapping of control channels in a rotating fashion, which makes neighbour channel measurements more feasible during traffic mode.

More details on compact mode can be found in [23].

1.5.6 GPRS and EGPRS Enhancements in Rel'4

In GERAN Rel'4, a few new features are added to the GPRS and EGPRS protocols and signalling to improve the efficiency of the data transfer with certain typical traffic characteristics and the seamlessness of the service in the cell change. These features, the delayed temporary block flow release and the network-assisted cell change (NACC), are introduced briefly in the following sections.

1.5.6.1 Extended UL TBF

The early GPRS protocols are inefficient when dealing with the bursty TCP/IP traffic. The GPRS radio interface protocols are initially designed to free the unused radio resources as soon as possible. With bursty IP traffic, this may lead to the frequent set-up and release of the radio resources (more specifically TBFs), which results in an increased signalling load and inefficiency in the data transfer, as the release and set-up of the TBFs takes some time, and radio interface resources. Unnecessary TBF set-ups and releases can be avoided by delaying the release of the TBF. In DL direction during the inactive period, the connection is maintained by periodically sending dummy LLC frames in the downlink [20]. This DL enhancement was already introduced in Rel'97 specifications. Its counterpart in UL direction is called *Extended UL TBF* as was specified in Rel'4. In extended UL TBF, the UL TBF may be maintained during inactivity periods (the MS does not have more RLC information to send), and the network determines the release of the UL TBF.

1.5.6.2 Network-assisted Cell Change

The GPRS cell change was not initially designed for the services that would require seamless cell change operation, such as real-time services. Later on, these requirements gained more importance and a new type of cell change, network-assisted cell change (NACC), was defined. In the NACC, the network can send neighbour cell system information to an MS. The MS can then use this system information when making the initial access to a new cell after the cell change. This way, the MS does not need to spend some time in the new cell receiving the system information, and the typical break in the ongoing service decreases from seconds to a few hundreds of milliseconds [24].

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