mobile communications series

Gunnar Heine
Holger Sagkob

GPRS

Gateway to Third Generation Mobile Networks

GPRS: Gateway to Third Generation Mobile Networks

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For Natalie, Sophie, and Patrick.

Et ecce ego vobiscum sum omnibus diebus usque ad consummationem saeculi.
—Matthew 28:20

Contents

Preface

This book is the second in its series. Many readers will also know my first Global System for Mobile Communications (GSM) book, which was written during my time in the United States from 1996 to 1998. For this new book on general packet radio service (GPRS) and high speed circuit switched data (HSCSD), I have had considerable support from my old university friend, Holger Sagkob, who is responsible for the HSCSD section. For the last project, Holger was mainly involved in the review process, and at that time we decided that we would publish a book together. He is also now a father—the "Patrick" referred to in the dedication is Holger's son by his wife Andrea. My wife and I have also been creative on this front and our first daughter, Natalie, now has a little sister, Sophie. This book is dedicated to our three children and to all poor, tortured mobile communication experts.

After my return to Germany from the United States, I founded my own company, INACON GmbH, which, because of our consistency in pursuing the American maxim, "The customer is always right," and due to our numerous customers worldwide, is growing constantly.

INACON GmbH focuses on everything to do with mobile communications. Our specialities are expert training and special consulting services for the mobile communications industry. Our best customers and partners include the big names in mobile communications: Siemens, Motorola, Ericsson, Nokia, Tektronix, and Cetecom, with whom I have a special personal relationship. In this book I shall make several references to one particular INACON product: an interactive, multimedia training CD-ROM that contains information on GSM and GPRS and, in the latest versions, also enhanced data rates for GSM evolution (EDGE) and Universal Mobile Telecommunication System (UMTS).

But let us return to mobile communications. Since my last book, the world of mobile communications has taken several dramatic steps forward. We have seen the surprising demise of most of the new satellite mobile communications operators, who were not able to successfully execute terrestrial mobile communications despite such a promising start in 1992.

We are currently seeing the beginning of the Bluetooth™ era—a new variation of mobile communications for short-distance communication. Although Bluetooth is nothing new from the technological point of view, it is nonetheless an interesting new angle, since the Special Interest Group (SIG) is a consortium of manufacturers that has been formed without any involvement on the part of the standardizing bodies such as European Telecommunications Standard Institute (ETSI).

This is similar to the case of the Wireless Application Protocol (WAP) Forum. The future will tell us the extent to which these spin-offs can improve the technology of mobile communications or whether ETSI, ANSI, and especially 3GPP can continue to set the standards.

The Third Generation Partnership Project (3GPP) is also the key organization in this book. The GSM is and will continue to be the worldwide standard in mobile communications systems.

GSM, with its ingenious safety features, has still not been cracked, to use hacker language. Furthermore, GSM—and particularly those responsible for it—have proven to be flexible enough to make GSM attractive for every frequency range all over the world.

What is more important for us, however, is that GSM has been expanded from a purely channel switched system to one that also incorporates a packet switched system without introducing any restrictions on the existing features. The GPRS is the proof of this.

The greatest achievement of all is the almost unnoticed blending of an existing GSM/GPRS network into a new UMTS network. 3GPP has really landed a major success here.

I often hear from the other side of the Atlantic that GSM is too complicated, which is why North Americans sometimes refer to it as the Great Signaling Monster. The decision of AT&T Wireless, however, to favor GSM/GPRS and my experience of being reachable all over the United States on my GSM phone make it very easy for me to overlook this criticism.

Here are a few more remarks on GPRS, the main subject of this book: by using GPRS, network operators can expand their GSM networks, which are (more or less) purely speech-based, to include (relatively) rapid data highways for Internet Protocol (IP)-related services. GPRS is only the first step in this direction: using GPRS as a base, it is possible to evolve to the use of EDGE or the UMTS, with even more flexible service features and even more bandwidth. These new services are expected to increase sales, which will then be reinvested to expand even further the already existing commercial success. The problems today (mid-2001) are still the same as they have always been:

1. Which services can be offered and charged mobile via the Internet?
2. Will users take advantage of the magnificent "always-on" feature from GPRS and UMTS in the future or remain more sporadically connected to the Internet?
3. What should a telephone in the mobile Internet era look like, and what features should it have? Currently, the trend is towards a hybrid of telephone and personal digital assistant (PDA).
4. Can the network operators maintain their ruling position in this service business or will they evolve into mere bandwidth providers, like the telecom companies? How can this process be prevented should it become necessary to do so?

Although this book is clearly orientated towards technological aspects and details, benefiting from the countless questions that our customers have asked us along the way, I would like to emphasize that it is primarily the marketing questions that will determine the success of GPRS and UMTS, and indeed the whole mobile communications of the future. This success cannot be forced by the solution of technical problems alone.

This book is not aimed at presenting all the details of GSM. In our opinion this job has already been done excellently, especially in [1, 2]. We refer our readers to these works for a detailed description of GSM. Nonetheless, the first chapter here is clearly focused on normal GSM. Even here, however, important linking threads between GSM and GPRS are introduced, which is why I strongly advise the reader against skipping this chapter. Never forget that GPRS takes over the air interface in a practically unchanged state. For this reason, such important procedures as cell selection and layer 1 remain basically unchanged.

One of my experiences as an author is the annoyance at one's own mistakes and miscalculations. At the same time it is a problem for the reader, as a customer, to find one of these errors. Although this problem cannot be solved because of time pressure and my own shortcomings, it can, however, at least speed up the correction of errors. For this reason, I provide my e-mail address: gheine@inacon.de. You can turn to this address not only for pointing out any mistakes in this book, which will hopefully be very few in number, but also if you have any further questions on GSM and GPRS that have not been answered here.

One final word on the numerous and perhaps irritating abbreviations: unfortunately, we specialists tend to abbreviate entire sentences, making them totally incomprehensible for the outsider ("ciphering"). For this reason, this book contains a comprehensive index of abbreviations in an attempt to defuse this situation a little.

Gunnar Heine
INACON GmbH
Karlsruhe, Germany
2003

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1

The Basics: Principles of GSM and Influences on GPRS

1.1 The Network Architecture of GSM

As an overview, each GSM network can be subdivided into the base station subsystem (BSS) and the network switching subsystem (NSS), as well as the mobile station. Please note that the introduction of GPRS can only expand, but must not change, the existing structure as presented in Figure 1.1, since both types of application—circuit switched and packet switched—should run via the mutual GSM/GPRS network.

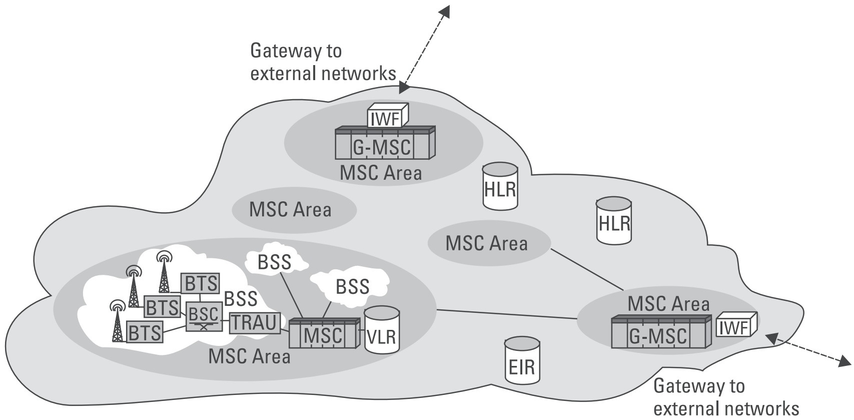


Figure 1.1 GSM network architecture.

1.1.1 The BSS

1.1.1.1 The Base Transceiver Station

The BSS consists primarily of a larger number of base transceiver stations (BTSs) that enable wireless connection of the mobile stations to the network via the U_m or air interface (Figure 1.2). Apart from transcoding rate and adaption unit (TRAU) framing, the BTS assumes all layer 1 functions in communications between the network and the mobile station. These include, amongst others, channel coding, interleaving, ciphering (only GSM, not GPRS), and burst generating. Other functions include Gaussian minimum shift keying (GMSK) modulation and demodulation, which are carried out by the base station and will be discussed in detail later.

1.1.1.2 The Base Station Controller

All BTSs of a BSS are connected to the base station controller (BSC) via the Abis interface (Figure 1.3). The BSC is, by definition, a circuit switching exchange in addition to the mobile services switching center (MSC), which will be discussed later. The BSC was basically viewed as a further exchange in order to relieve the MSC from all wireless-related tasks. These include, in particular, the evaluation of the measurement results from the BTS and mobile station during a live connection and the handover and power control adjustments resulting from this.

These regulatory functions are generally performed in their entirety by the BSC, although the GSM standard expressly allows preliminary prepara-

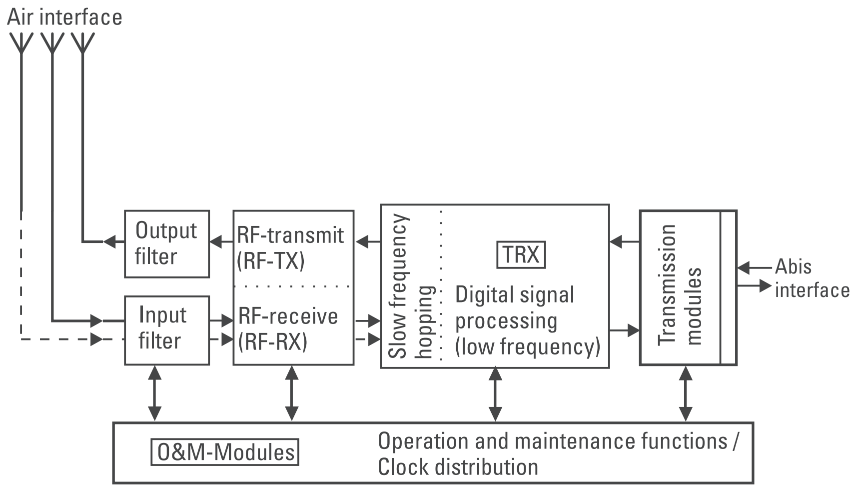


Figure 1.2 Principal schematic diagram of the base transceiver station.

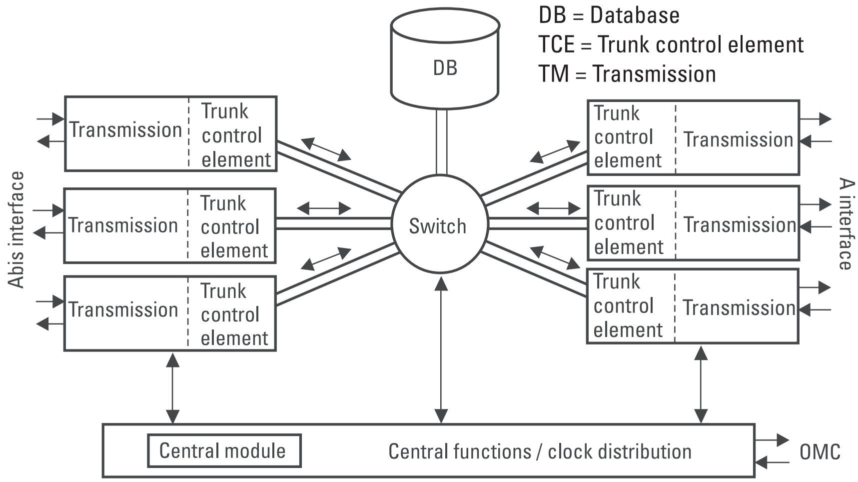


Figure 1.3 Principal circuit diagram of the BSC.

tion of the measuring results in the BTS. Additional BSC functions are the *Peer* function of the mobile station for the Radio Resource Management protocol (RR) and the resource administration on the Abis and air interface.

The BSC, as a circuit switching network element, is a considerable hindrance to packet switched services (GPRS). Its exchange functions are almost unusable for packet switched services, and the RR protocol is extremely difficult to adjust to the requirements of packet switched services. Hence, if the BSS is to be used at all for GPRS, the BSC must either be modified accordingly or a new network element or an extension of the BSC will be necessary.

1.1.1.3 The Transcoding Rate and Adaptation Unit

The TRAU is the third BSS network element. The best-known task of the TRAU is speech compression from 64 Kbps to 16 Kbps (full-rate) or 8 Kbps (half-rate). The TRAU also carries out comfort noise generation while discontinuous transmission (DTX) is in operation.

What is considerably more important for a basic understanding of signal processing within GSM is another TRAU function: the conversion of all information coming from the MSC into so-called TRAU frames. This conversion is carried out for fax, data, and speech. In other words, all payload transfer between mobile station and TRAU takes place on the basis of TRAU frames. TRAU frames have a length of 320 bits. Every 20 ms a TRAU frame is transmitted or received. Consequently, there are channels of 16 Kbps.

The number of actual payload bits will vary depending on the type of TRAU frame or application. For full-rate speech and enhanced full-rate speech, the TRAU frame will, for example, contain 260 bits of payload data, whereas the normal data TRAU frame contains 240 payload data bits (see Section 1.7.1).

As already implied, payload channels of 16 Kbps are used on account of the TRAU framing between the TRAU and the BTS, especially on the Abis interface. In other words, if more than 16 Kbps are transferred, there is a problem. This is, however, exactly what happens in data transfer via GPRS or EDGE. Most manufacturers will have to find new approaches in order to solve this problem.

Since the functions of the TRAU are specific layer 1 functions, the TRAU function should be assumed to be locally situated in the BTS. Indeed, the GSM standard permits the integration of the TRAU into the BTS. This possibility is illustrated in Figure 1.4. Most manufacturers, however, take a different course and use so-called remote TRAUs. The reason for this is the opportunity to save on connection costs. If the TRAU is installed on the MSC, then 16-Kbps channels can be used all the way from the MSC to the BTS, instead of 64-Kbps channels. In other words, a remote TRAU cuts connection costs by three-quarters. For the implementation of GPRS, the actual position of the TRAU is of some significance, since the packet switched GPRS data is fed into the existing GSM network at some point. We shall encounter this again in Chapter 2.

1.1.2 The Network Switching Subsystem

As shown in Figure 1.1, the NSS consists of one or more home location registers (HLR) with the authentication center (AuC) and optionally with

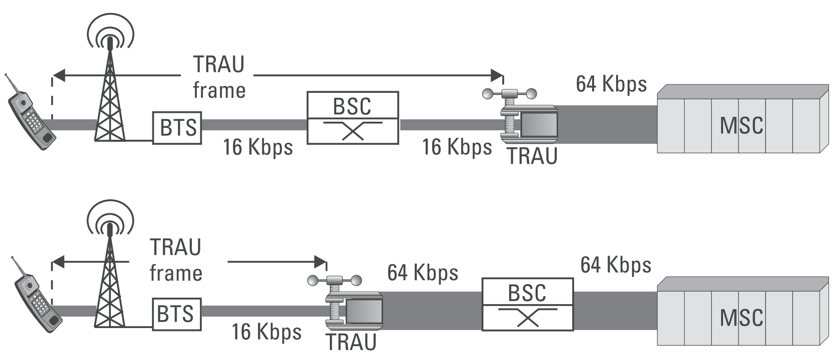


Figure 1.4 Possible location of the TRAU.

the equipment identity register (EIR) and various MSCs with a connected visitor location register (VLR). Please note that the NSS is also used by GPRS, at least in part.

1.1.2.1 The Home Location Register and the Authentication Center

The HLR is a static database in which information on hundreds of thousands of subscribers is stored. This information includes the telephone number(s) [i.e., the mobile subscriber international service directory number (MS-ISDN)] of a subscriber as well as his service characteristics and service limitations. For mobility management (MM), which is so important in GSM, the HLR holds the information as to which VLR area a subscriber is currently registered. With the introduction of GPRS, the data on individual subscribers in the HLR will be more comprehensive. This implies that for GPRS, the HLR must not only possess the information regarding the respective VLR but also that of the corresponding serving GPRS support node (SGSN). Other GPRS-specific data stored in the HLR are possible Packet Data Protocol (PDP) contexts and service characteristics and service limitations, only for GPRS this time.

The AuC, which is an integral part of the HLR, calculates the respective authentication results (SRES) and ciphering keys (Kc), using the algorithms A3 and A8 from RAND numbers (RAND = random number) and the subscriber keys Ki stored in the HLR. These processes are presented in diagrammatic form in Figures 1.5 and 1.6. Please note that the AuC predetermines up to five so-called authentication triplets (RAND, SRES, Kc) for each subscriber and puts them at the disposal of the VLR responsible, via

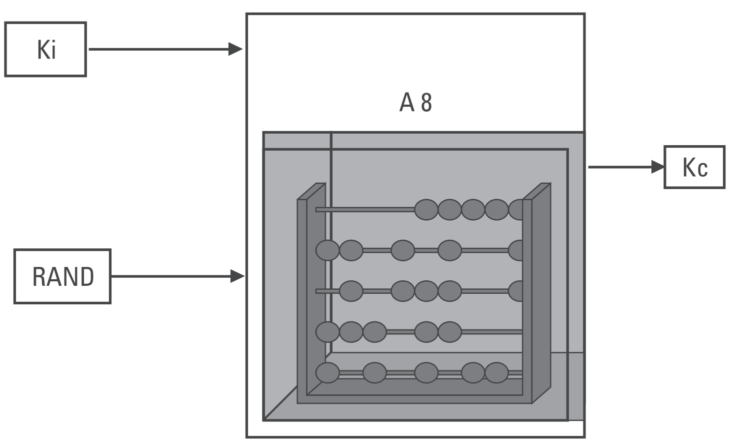


Figure 1.5 The determination of SRES from Ki and RAND.

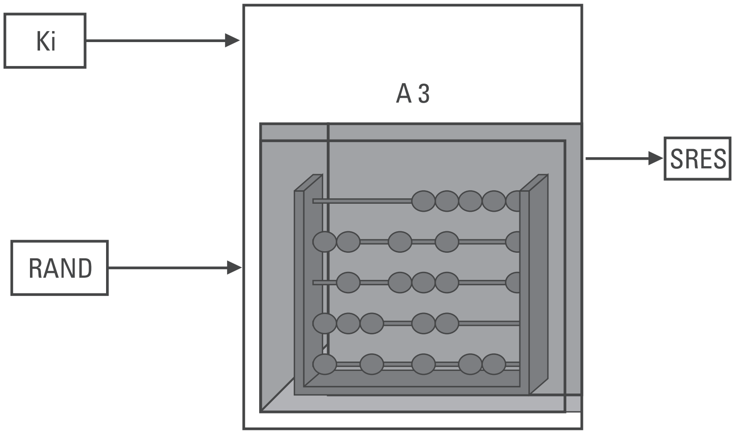


Figure 1.6 The determination of the ciphering key Kc from Ki and RAND.

the HLR, for authentication purposes. A detailed description of the processes of GSM authentication and GSM ciphering can be found in [1, 2].

The introduction of GPRS does not alter these GSM mechanisms. It should be noted, however, that in GPRS the authentication and activation of the GPRS ciphering are controlled by the SGSN. As a consequence, a mobile station can be authenticated twice, once by the VLR and once by the SGSN, each with a different RAND variable, of course. Accordingly, two different Kc values must be stored and ready for retrieval in the mobile station—one for GPRS and one for normal GSM. This poses a problem for older subscriber identity module (SIM) cards, which will be discussed in more detail later in the book together with ciphering in GPRS.

1.1.2.2 The Mobile Services Switching Center and Visitor Location Register

Before the introduction of GSM in the 1980s, MSC and VLR were conceived as two independent network elements: the MSC as a network element for all call control (CC) functions and the VLR for the greater part of the MM functions. Both protocols, CC and MM, are transparent for the BSS and are treated between the MSC and the VLR on the one hand, and by the mobile station on the other. A detailed presentation of both protocols and their functions can be found in [1, 2]. In the early 1990s, after the introduction of GSM, the physical independence of MSC and VLR disappeared, and by 1997 the MSC and VLR became the MSC/VLR. This did not, however, alter the protocol independence of MM and CC.

It is important to understand about GSM that the MSC is essentially an ISDN exchange that has been modified for use as a GSM-MSC. ISDN exchanges, however, are circuit switched. Historically, this path was taken at the end of the 1980s because GSM was supposed to be as ISDN compatible as possible. One of the problems to be solved in this context was that existing exchanges, such as the Siemens EWSD System or the Alcatel System12, could assume the RR functions, which are necessary for a mobile telephone system, only with great difficulty, at least not without drastic, and thus expensive, modifications. Therefore, an unusual way was taken with GSM and these RR functions were relocated to the BSC. As a consequence, circuit switched exchanges, which are unsuitable for a packet switched transfer process such as GPRS, are situated centrally in the form of MSCs in GSM networks. The consequences of this will be discussed in more detail in the next chapter.

The Gateway MSC and the Interworking Function

In Figure 1.1, a typical Public Land Mobile Network (PLMN) with different MSCs was presented. In total, only two of these MSCs have an interface to external networks. These special MSCs are described as gateway MSCs (G-MSCs) in GSM. The network operator has to decide whether all or only selected MSCs should have this interface function.

On the side facing away from the PLMN of a G-MSC, there is the so-called interworking function (IWF), which, amongst other things, takes care of the rate adaptation (RA) functions in connections to external data networks. For this reason, the IWF is also frequently called a modem base.

GSM supports interworking with different types of external networks such as Circuit Switched Public Data Networks (CSPDNs), Packet Switched Public Data Networks (PSPDNs), the Public Switched Telephone Network (PSTN), and Integrated Services Digital Network (ISDN).

1.1.2.3 The Equipment Identity Register

In contrast to the databases in GSM already described (i.e., the VLR and the HLR), the EIR does not administer subscriber data but the data of the mobile terminals themselves. Another difference from VLR and HLR is the fact that the EIR is an optional network element that has, for reasons of cost, only rarely been introduced by network operators.

It is important to look at the historical development of EIR. In the standardizing phase of GSM in the 1980s, mobile devices and mobile telephoning were very expensive and the danger of theft and abuse was accordingly high. By definition, GSM opens up new doors for the black market since the subscriber's identity [the international mobile subscriber identity (IMSI)] and all his data, such as the telephone number (the MS-ISDN), are separated from the identity of the device itself. In other words, theoretically, a stolen device could be used as early as the day of its theft without anyone noticing whether a different SIM card is used. To counteract this danger, two measures were taken. First, every GSM device must be given an unchangeable and unmistakable identity number [the international mobile equipment identity (IMEI)]. Second, the EIR, in which stolen or conspicuous IMEIs can be stored, was introduced. It became clear, however, on or shortly after the introduction of GSM in 1991 that the prices of GSM devices were going to fall and thus theft protection and mechanisms to counter the black market were no longer going to be of primary concern to the end user. In accordance with this, many network operators no longer placed orders for EIRs or cancelled existing orders.

The GPRS core network, however, which we will introduce later, also has interfaces to the EIR for compatibility reasons.

1.1.3 The GSM Mobile Station and the SIM

The expression, "GSM Mobile Station and the Subscriber Identity Module" is in itself incorrect, because the GSM mobile station (MS) only arises through the physical connection of GSM mobile equipment (ME) with a SIM. To put it simply, ME + SIM = MS. Nevertheless, many specialists use the term "mobile station" as a synonym for the correct term, "mobile equipment," which is why we shall not make any differentiation in the following unless it is necessary to do so.

Let us return now to the GSM mobile device, which is an essential part of GSM's success. Many characteristics of GSM are defined in terms of the mobile device:

- The cellular network configuration with relatively small cell sizes enables low transmission energy consumption on the MS side, which is why the mobile station battery can be kept small and light.
- The GMSK modulation used in GSM enables the use of low-cost power amplifiers; this is basically a simple modulation process. Production costs should also be accordingly low.
- The original GSM standard did not provide for GSM mobile stations being able to transmit and receive simultaneously. Duplex operation was not envisaged. Consequently, a duplexer on the interface between transmitting/receiving path and antenna was not necessary. This factor also reduces the complexity and costs of a GSM mobile station.

- As opposed to other network elements in the NSS and BSS, detailed requirements, for example, of the man-machine interface (MMI), were defined for the mobile station.
- The clear definition of compulsory and optional features of the GSM mobile station with regard to performance allows for hundreds of test cases. These are specially defined for mobile stations in the GSM standards (GSM 11.10). Every GSM mobile station must conform with these test cases before it is permitted to be retailed. At first glance, this restriction may seem to be a hindrance, but it proves to be most advantageous in the long run because it ensures customer confidence and reduces significantly the number of costly recall campaigns.

Despite these simplifications every GSM mobile station is a piece of top-rate technology. As illustrated in Figure 1.7, a GSM mobile station contains all layer 1 functions that can also be found in the BTS and the TRAU. Furthermore, the mobile station must support all MM and CC functions in conversation alongside the MSC and the VLR. There also have to be mechanical devices for the insertion and removal of the SIM. Finally, the different user interfaces have to be integrated. These include, in addition to the MMI, loudspeaker, microphone, and electrical and/or optical interfaces for data connections.

1.1.3.1 The GPRS Mobile Station

For the introduction of GPRS, the functions of the GSM mobile station must be diversified in many areas. Although we shall be examining this process in more detail later, the essential characteristics or differences between

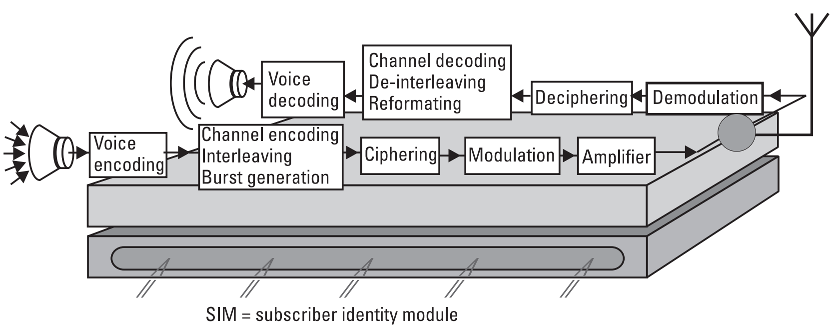


Figure 1.7 Circuit diagram of a GSM mobile station.

a GSM/GPRS mobile station as compared to a purely GSM mobile station should be outlined now:

- New protocol stack: support of new protocols in the radio resource, mobility management, and session management areas;
- Support of new channel coding processes;
- Multislot transmission: With higher multislot classes (type 2), even simultaneous transmitting and receiving is possible. Apart from the multifunctional capability itself, the increased demands on the battery capacity must also be considered;
- Data services require a new MMI. For instance, the touchscreen of GSM/GPRS-PDAs are used both as a keypad for telephoning and a display area for visual information;
- Possibly the development of GPRS-only mobile stations, which are virtually wireless Internet sockets and no longer offer any speech services at all;
- As an option, the simultaneous support of channel and packet-orientated services—for example, downloading e-mail while telephoning.

1.2 The Multiple Access Processes: SDMA, FDMA, and TDMA

As a second generation mobile communication network, GSM uses the three classical multiple access processes, space division multiple access (SDMA), frequency division multiple access (FDMA), and time division multiple access (TDMA) in parallel and simultaneously. GPRS does not alter this nor many other basic GSM processes.

1.2.1 SDMA

The entire geographical area is not supplied by a single transmitting station. The transmitting power of the individual transmitting stations is limited in order that a given frequency may be used again at a short distance away. As illustrated in Figure 1.8, however, SDMA gives rise to a cellular network structure that has both advantages and disadvantages. The most important advantages are the high reusability rate of the frequencies used and, at least as important, the considerably lower demands on the transmitting power of

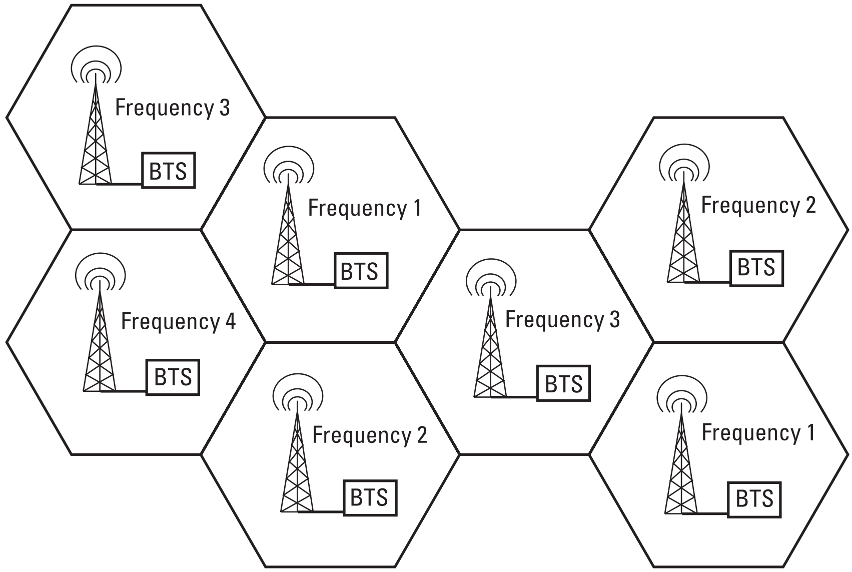


Figure 1.8 The SDMA multiple access process gives rise to cellular network architecture.

the mobile stations. Accordingly, it is possible to produce small mobile stations with low power requirements. On the other hand, the SDMA configuration automatically leads to a complex network structure, which is necessary to connect the individual transmitters to each other and to enable standard functions such as roaming and handover.

1.2.2 FDMA

Similarly to SDMA, FDMA is a multiple access process that is relatively easy to understand and in which the given frequency band is divided into individual frequency channels. Each user is allocated just one of these narrow channels. In this context it has to be considered that two frequency channels are necessary for a bidirectional connection: one for the transmission to the mobile station (downlink) and one for the opposite direction from the mobile station to the base station (uplink). In GSM, a complete frequency channel thus requires 2×200 kHz. Here, the frequency distance between the uplink and downlink frequency channels is always determined precisely and only changes for the different GSM variants (see Figure 1.9). For example, in PCS1900, the GSM variant used in the United States, this distance between the uplink and downlink channels is exactly 80 MHz, whereas in P-GSM900, it is 45 MHz. A serious disadvantage of the usual FDMA systems is the

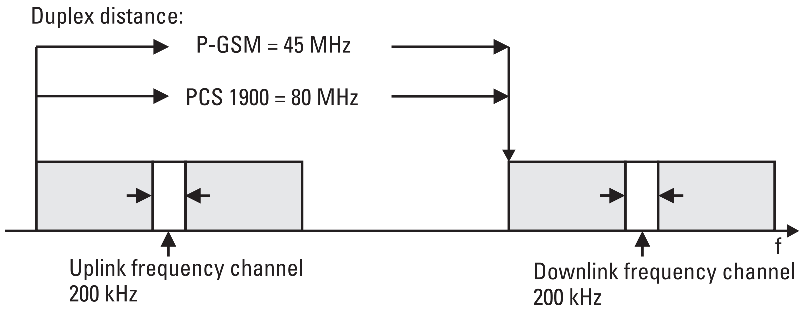


Figure 1.9 There is a set distance between the uplink and downlink channels in the application of FDMA in GSM.

necessity of so-called paired bands (i.e., two frequency bands that have to be provided at a fixed duplex distance form one another). Such systems are also described as frequency division duplex (FDD) systems. This requirement is of particular disadvantage because frequency is a rare resource, as the bidding for the UMTS licenses has clearly demonstrated to the general public. If, for example, one wishes to operate GSM in a country or a region, it first has to be determined whether the uplink and downlink frequencies are even available.

1.2.3 TDMA

For FDMA, the available frequency range is divided into individual frequency channels. When there is an active connection, a subscriber receives this frequency channel exclusively for the entire duration of the call and no one else can use this part of the spectrum.

With TDMA, a further step is taken. Each frequency channel is also subdivided temporally and each subscriber receives access rights to the frequency channel during a connection for a relatively short but repeated period of time. In a TDMA system these periodically repeated time intervals are called time slots (Figure 1.10). In order to give the impression of an uninterrupted connection, sufficient information must be transmitted in these time slots per connection.

In GSM, each frequency channel is divided into eight time slots (TS), as shown in Figure 1.11. Each time slot has a length of 576.9 μs ($\approx$ 577 μs) or 156.25 bits and is repeated every 4.615 ms. According to this definition, up to eight users in GSM can use one frequency channel almost simultaneously and independently from one another. It must be stressed again, however,

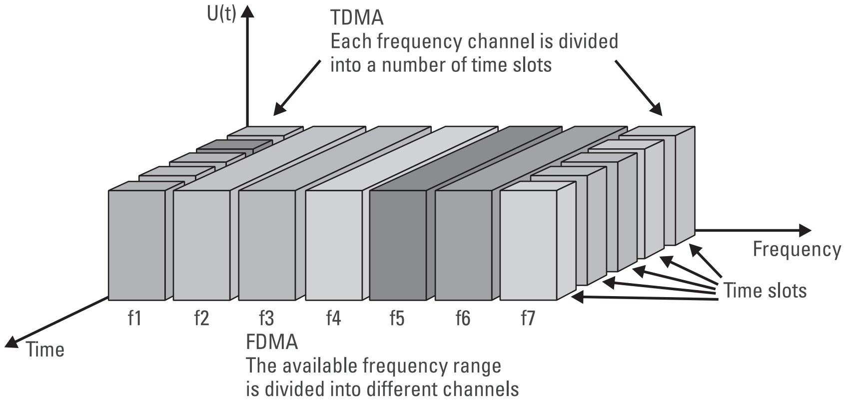


Figure 1.10 The combination of FDMA and TDMA.

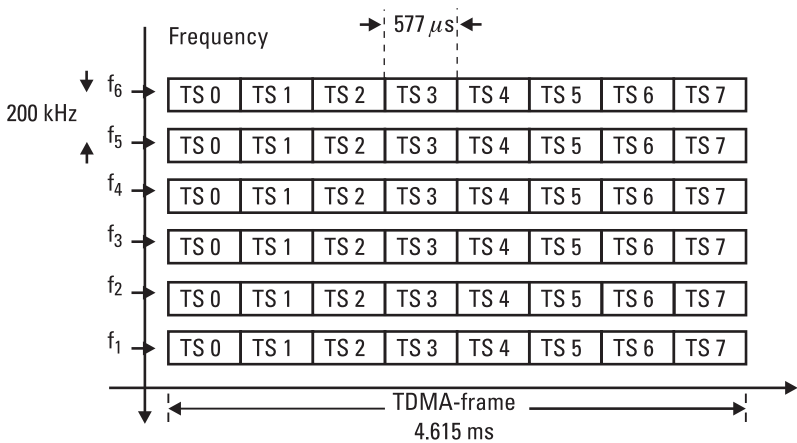


Figure 1.11 The combination of FDMA and TDMA in GSM.

that Figure 1.11 only represents one direction, but two frequencies are required for a bidirectional connection, in which the same time slot is used.

1.3 Chronological Sequence of Uplink and Downlink Transmission

In GSM, the base station always transmits three time slots before the mobile station. In other words, the transmission of a time slot in the downlink direction always takes place three time slots before the transmission of the same time slot in the uplink direction. It could also be stated thus: time slot X in the downlink direction is three time slots before time slot X in the uplink direction (Figure 1.12). A mobile station that has synchronized itself to a base station and receives information in time slot X will wait 3 time slots, or 3×156.25 bits = 468.75 bits, before sending its data to the base station. This golden rule for GSM, which is only compromised in the transmission delay problem described in Section 1.4, does not change with the introduction of GPRS.

One should also consider Figure 1.12 from the point of view of multislot transmission. If simultaneous transmission and reception on the side of the mobile stations is to be avoided, there are only a few uplink/downlink time slot combinations available that can avoid this problem. It is then actually impossible to provide an individual user with more than four time slots in the downlink direction or four in the uplink direction. This applies in particular when transmissions are to take place in the opposite direction and the mobile station has to carry out neighboring cell measurements at the same time.

1.4 Problems of Transmission Delay in TDMA Systems—Timing Advance Control

In every TDMA system, data transmission in both directions necessarily takes place in the form of impulses. In GSM, these impulses are called *bursts*. One of the main problems of TDMA systems, which must not be neglected, is the delay time that it takes to transmit a burst from transmitter to receiver. In the direction from the base station to the mobile station (downlink), there are no problems in this respect as every mobile station can receive its signal, its burst, independently from other mobile stations. In the other direction, however, [i.e., from the mobile station to the network (uplink)] there is the possibility of collisions with the bursts sent from various mobile stations. The cause of the unknown delay time is the unknown distances

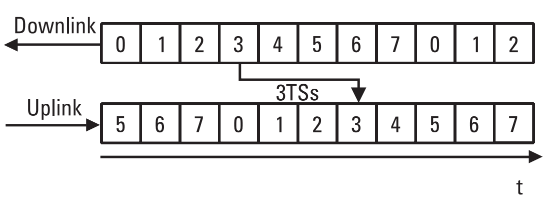


Figure 1.12 Synchronization of downlink and uplink transmission in GSM.

between the mobile stations and the base stations. Since mobile stations may also move within the network, these delay times vary during a connection. Accordingly, the various active mobile stations must constantly adjust the starting time of their transmission in order to reach their receiver window in the base station (Figure 1.13). The solution to this problem of delay time in the uplink direction is not only necessary for the beginning of a connection but is also necessary during a live connection. Otherwise, mobile stations would have to be prohibited from moving at all during an active connection. In GSM delay time control is spoken of as timing advance (TA) control.

1.4.1 Timing Advance Control When Accessing the Network

Timing advance control appears particularly difficult when accessing the network. At this point, the mobile station can be almost any distance from the bases station. In any case, this distance is unknown. One must therefore ask:

1. How does the mobile station inform the base station about its intention to access the network at this time?
2. How can the collision of the signal from the mobile station with signals from other mobile stations due to the unknown delay time be avoided?

In GSM, the mobile station uses the *access burst* for initial access to the network. This is much shorter than the *normal burst* and will thus

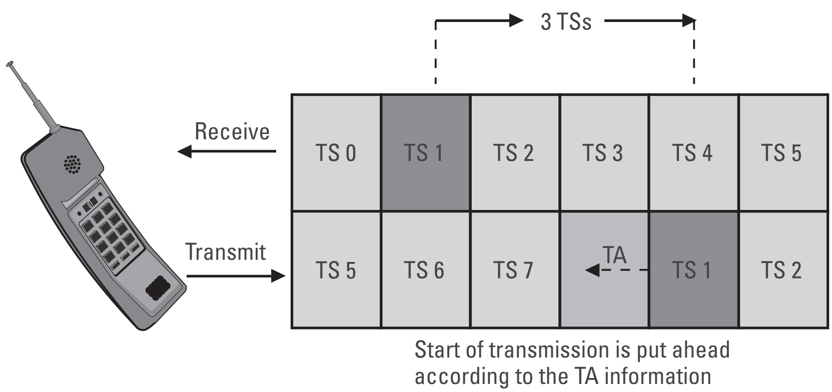


Figure 1.13 The mobile station sets its start of transmission ahead according to the timing advance.

definitely fit into the base station's receiver window, even if it has been sent from a long distance (Figure 1.14). The mobile station always assumes the timing advance (i.e., the distance from the network) to be zero when an access burst is transmitted (slotted Aloha; Chapter 2). The length of the access burst and the width of the receiver window on the BTS side are added to give the maximum radius of a base station of 35 km. According to the time of entry of the access burst at the respective receiver window, the base station estimates the distance to the mobile station and returns this value to the mobile station during the channel assignment. The mobile station then regulates its timing advance by the respective number of bits and can then, from this time on, use normal bursts. Note that in GSM, without the so-called extended cell operation, the TA value varies, depending on the distance, between 0 and 63_{dez}.

Example. The base station passes on a TA value of 26 to the mobile station. The mobile station then transmits not 486.75 bits, but 460.75 bits (486.75 – 26) to the base station. There is then a 1:1 correspondence between the TA value and the delay time between receiving and transmitting on the mobile station side.

1.4.2 Timing Advance Control During a Connection

During a connection, the base station receives a burst from the mobile station every 4.615 ms. Bursts are discussed in detail in Section 1.6.6. With the help of the training sequence code (TSC) in normal burst (Figure 1.15 and

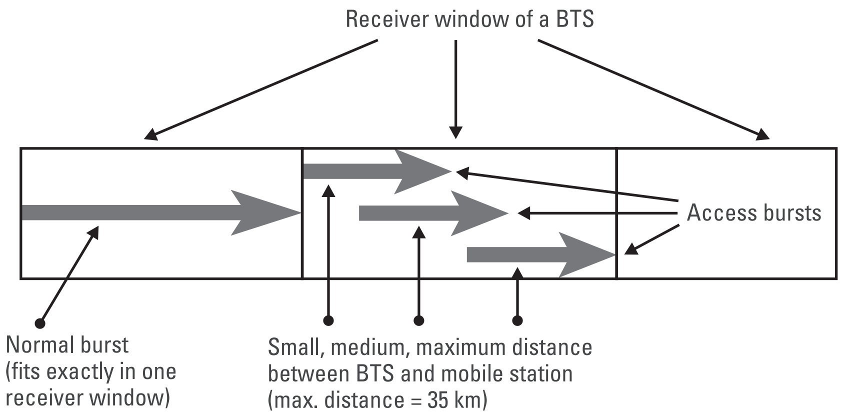


Figure 1.14 The short length of the access burst allows it to be sent from distances of up to 35 km.

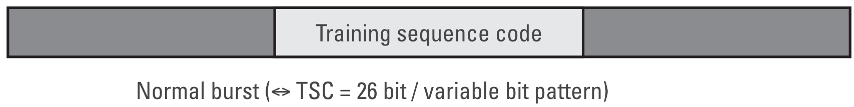


Figure 1.15 The BTS measures the bit shifting of the known training sequence code in the uplink normal burst for determining the timing advance.

Figure 1.34), familiar to both sides, the base station can use bit shifting. This arises due to varying distance within the training sequence code, for adjusting the timing advance. Note that the process presented is based on the periodic transmission of bursts in the uplink direction during an active connection. This condition also applies with an active DTX because even then, a burst is sent from the mobile station to the base station every 120 ms. The question remains: How does timing advance control work in GPRS, which does not provide for this kind of regular transmitting? We will answer this question in a later chapter; our intention at this stage is merely to point out the problem to the reader.

1.5 Frame Hierarchy and Logical Channels in GSM

As shown in Section 1.2, GSM uses TDMA as a multiple access process as well as SDMA and FDMA. Each frequency channel is subdivided into eight independent time slots. However, a further step is taken. As shown in Section 1.2, each time slot is repeated every 4.615 ms. In order to be able to deal with all tasks, the different types of logical channel are placed onto the individual time slots. In other words, each time slot is not occupied by the same logical channel type every time but is occupied by different logical channel types in sequence. This principle especially applies to the different signaling channels, and also to the traffic channels and their associated control channels (SACCH). Please refer to [1, 2] for a detailed description of the logical channel types in GSM. These are summarized in Figure 1.16.

1.5.1 The 51 Multiframe

It is important that the mobile station and the base station know when which logical channel is going to use which time slot. This necessity led to the introduction of frame hierarchy and in particular to the definition of the various types of *multiframe*. As an example, we shall take time slot 0 of the so-called Carrier 0 (C0; BCCH carrier):

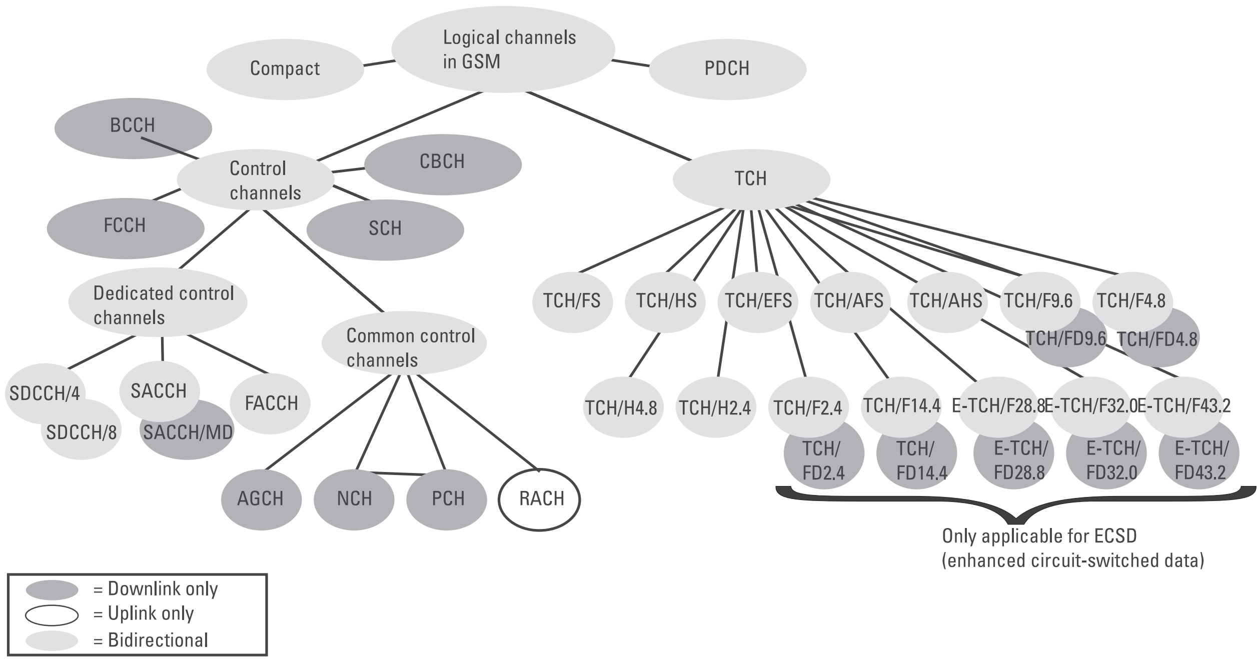


Figure 1.16 The logical channel types in GSM.

While time slot 0 of the broadcast control channel (BCCH) carrier is initially used for the frequency correction channel (FCCH), it is then used for the synchronization channel (SCH). Following this, it is then used four times consecutively for the transmission of the four segments of an item of system information (SYS_INFO). This process is illustrated in detail on the time axis in Figure 1.17.

Figure 1.17, for all its clarity and despite being an abridged version, saves little space. Usually, the "wallpaper" illustration is used, as in Figure 1.18. Figure 1.18 represents the occupation of time slot 0 on the so-called BCCH carrier over a longer period of time. Note that Figure 1.17 only displays the first six frame numbers (FN) of Figure 1.18, which shows the entire 51 multiframe. As the name suggests, each 51 multiframe consists of 51 repetitions of the same time slot.

In this context it is important to point out that Figure 1.18 only shows one possible variation of the 51 multiframe in the downlink direction, which again can only be found in time slot 0 of the BCCH carrier. The reason for this is that the FCCH and the SCH of a BTS can only be configured in this time slot. The FCCH can always be found in the position FN = X0 of a 51 multiframe, where X can assume the values of 0, 1, 2, 3, or 4. An FCCH is always followed by an SCH. The reason for these restrictions is obvious. FCCH and SCH play a decisive part in the initial cell selection and synchronization of a mobile station to a BTS. The mobile station must be able to identify where time slot 0 is situated on the time axis. Further details on the FCCH and SCH are presented in Section 1.6. We are describing this process in such detail here because the introduction of GPRS does not alter the process of cell selection up to this stage. In the further course of this cell selection, the mobile station reads the so-called BCCH or system information of a BTS. The system information communicates all cell-specific data to the surrounding mobile stations. These include information about the BTS's neighboring cells, the cell identity of a BTS, and the physical parameters for possible access to the cell. As shown in Figures 1.17 and

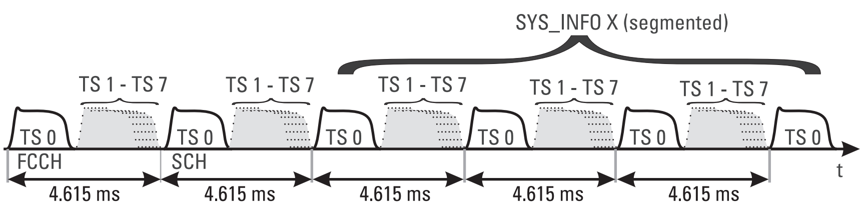


Figure 1.17 The chronological sequence of logical channels (using time slot 0 of the BCCH carrier as an example).

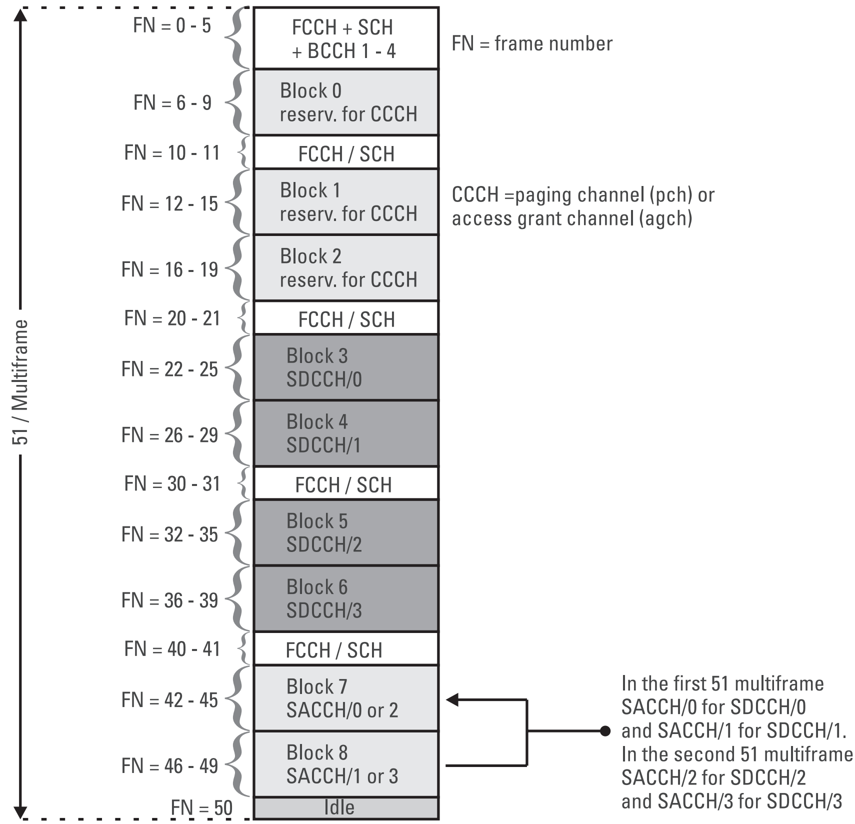


Figure 1.18 Example of a 51 multiframe with a duration of 235.38 ms in the downlink direction of time slot 0 of a BCCH carrier.

1.18, each 51 multiframe can transmit one item of BCCH/SYS information, which means that it takes a certain amount of time until a mobile station has read all the cell information. However, a different amount of BCCH/SYS information has been defined for the various GSM derivations.

Example. For P-GSM900, four BCCH/SYS info messages are defined (plus SYS_INFO5 and 6 for the SACCH), and accordingly, it takes four 51 multiframes or 4×235.38 ms $\approx$ 1s until a mobile station has read all four items of BCCH/SYS info. These circumstances mean an unavoidable delay in the synchronization of a mobile station to a BTS. This delay is increased in DCS1800 and PCS1900 by the presence of further BCCH/SYS info.

With regard to the 51 multiframe, it must be stressed that it may only be used in time slots with circuit switched control channels (SDCCH, CCCH, BCCH, SCH, FCCH), but not for circuit switched traffic channels (TCH).

1.5.1.1 The Introduction of the BCCH/SYS Info 13 for GPRS

With the introduction of GPRS, an additional item of system information, number 13, was defined. In other words, if a BTS supports GPRS, it will definitely transmit the BCCH/SYS info 13 in time slot 0 of the BCCH carrier. However, a mobile station does not need to wait for a complete 51 multiframe cycle in order to be able to decide whether or not a cell supports GPRS. The presence of the BCCH/SYS info 13 will be announced in advance in at least one of BCCH/SYS info 3, 4, 7, or 8. Whether the BCCH/SYS info 13 actually contains GPRS-specific cell information or merely a pointer to a packet broadcast control channel (PBCCH; to be discussed in further detail later) depends on the presence of a PBCCH.

1.5.2 The 26 Multiframe

The 51 multiframe was presented in the previous section as a time basis for circuit switched control channels. We shall now turn to the 26 multiframe as a time basis for all circuit switched traffic channels. The 26 multiframe consists of only 26 times the repetition of the same time slot and accordingly has a duration of exactly 120 ms. Its structure, however, is significantly more simple than that of the 51 multiframe, as Figure 1.19 shows.

There are three peculiarities to note. While a time slot with 51 multiframes is divided up for many different control channels and users, a time slot with 26 multiframes is used exclusively by one subscriber (except in half-rate configuration when there are two subscribers). In position 12 of the 26 multiframe we find the permanently configured slow associated control channel (SACCH), which amongst other things transmits information to timing advance and power control. Since a SACCH message is also segmented, four 26 multiframes, or 480 ms, are required for the transmission of a SACCH message. In position 25 we find an idle frame, which is used by the mobile station for the measurement of the SCHs of the various neighboring cells.

This context makes it clear for the first time why there are two different types of multiframe. If the traffic channels also used the 51 multiframe, the corresponding (traffic) idle frame for measuring the SCH of the neighboring stations would always fall on the same frame number in the (control) 51 multiframe of the neighboring stations. All, or most, of the SCHs could never be read, and the handover function would have to be reconsidered because BTSs would then have to be finely synchronized to each other.

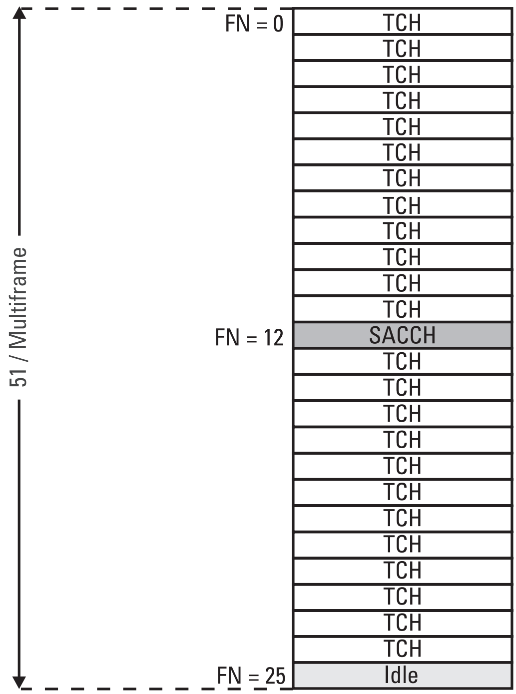


Figure 1.19 Example of a 26 multiframe of 120-ms duration in the uplink or downlink direction.

1.5.3 The Frame Hierarchy

GSM introduces the frame hierarchy as a BTS internal clock in order to be able to regulate occupation with logical channels per time slot (i.e., to be able to know and inform the mobile stations when which time slot is being used by which logical channel).

The frame hierarchy (Figure 1.20) starts with the TDMA frame as the smallest unit, which one should imagine as the repetition rate of the same time slot. We then find, on the next levels of the hierarchy, first the 51 multiframe for signaling channels and the 26 multiframe for traffic channels. The framework hierarchy then continues with the superframe and ends with the hyperframe, which has a cycle of about 3.5 hours. It is interesting to note that the hyperframe was only introduced because the FN (i.e., the number of the respective TDMA frame) is an entry parameter for the ciphering algorithm A5 in GSM. If the frame hierarchy had ended with the superframe, undesirable listening in on GSM would be much easier since the encrypting sequences would be repeated every 6.12 sec.

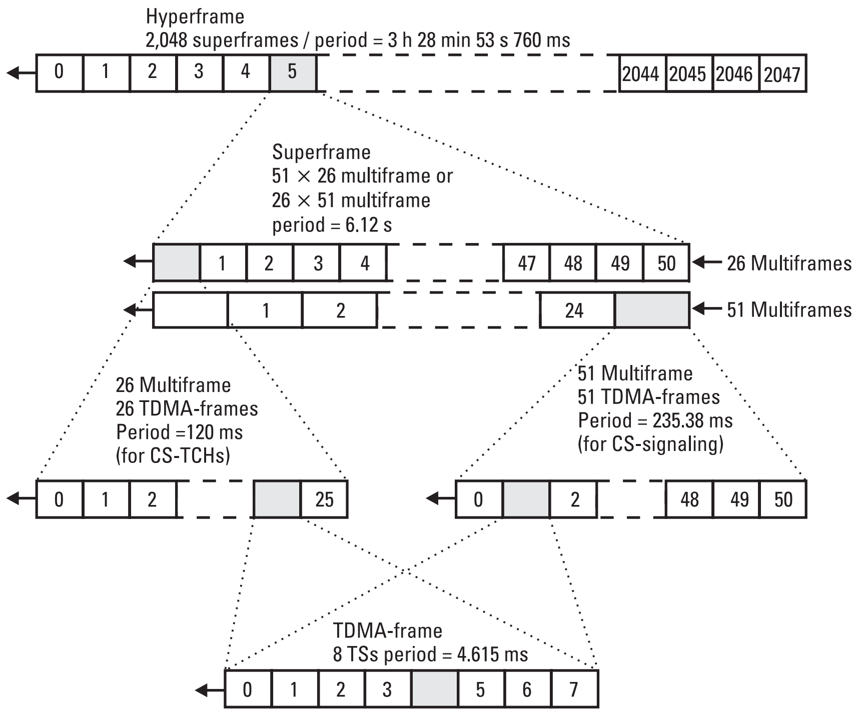


Figure 1.20 The frame hierarchy in GSM.

1.6 The GSM Signal Processing Chain

1.6.1 Introduction and Overview

In order to understand GSM, it is essential to understand the individual steps of signal processing. Frequently asked questions include the following:

- Which network element takes on which function?
- Where does each particular type of signal processing take place?
- What are the differences and/or similarities between data and speech processing?

Figure 1.21 presents a simplified version of the course of signal processing in GSM in the downlink direction, with a reference to the network element responsible in each case. The most important steps are TRAU framing or speech compression, channel coding for forward error correction

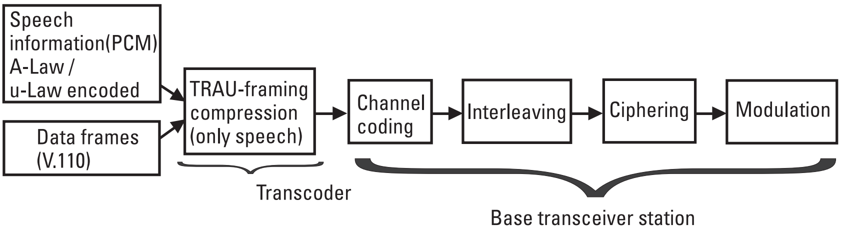


Figure 1.21 The signal processing chain in GSM.

(FEC), the process of interleaving, the encryption procedure on the burst level, and the actual modulation. These steps will be outlined briefly in the following sections. For a detailed description of all processes, see [1–3].

1.6.2 Data Formats at the Entrance to the PLMN

As illustrated in Figure 1.21, there are two data formats in particular that must be differentiated:

1. PCM coded speech information, which can be coded internationally as μ *law* (United States and Japan) or *a law* (rest of the world);
2. Data information that is converted on entry into the GSM network within the interworking function (IWF) into V.110 frames.

There is also signaling data, such as that for the establishment and release of a connection or short messages (SMS), but these formats should be omitted at this stage to ensure clarity.

An interesting point here is the fact that GSM allowed connection to packet switched networks [e.g., the Packet Switched Public Data Network (PSPDN)] even before the introduction of GPRS. Here too, however, there is conversion into V.110 frames, which are then processed and channel switched further within the GSM network.

1.6.3 Channel Coding

GSM uses different methods for so-called channel coding, depending on the type of information that is to be processed (e.g., various data channels, signaling, speech). In this respect, the introduction of new coding methods with GPRS does not entail anything fundamentally new. Even the process of puncturing was introduced before GPRS in GSM. The new element with respect to channel coding with GPRS is the process of adaptive adjustment of the coding method according to reception quality in order to optimize throughput rates. For this reason, this introductory part will present the process of channel coding in a conceptional way.

1.6.3.1 Introduction to Backward Error Correction

The transmission of information via the air interface is by its very nature susceptible to any kind of disturbance, especially in mobile use. As a consequence of fading or interferences, an unspecified number of bit errors will occur with indeterminate regularity and predictability. It is the greatly varying frequency and strength of bit errors in particular that make handling this problem on the wireless channel complicated, and this is precisely where the air interface or wireless channel as a transmission channel differs from a terrestrial connection such as a glass fiber or coaxial cable.

In order to correct such transmission errors, the method most frequently used is that of backward error correction (BEC). However, this requires the retransmission of a defective data frame by the original transmitter after an error has been recognized by the receiver.

In brief, the BEC method is based on the following procedures (see also Figure 1.22):

- Calculation of a checksum in the transmitter and attachment of this checksum to the actual data block (2);
- Transmission of the data block and checksum (3);
- Testing the consistency of the data block and checksum at the receiving end (4);
- Rerequesting a defective data block using an appropriate message (e.g., *Not Acknowledgement* = NACK) (5);
- Retransmitting the defective data block (6), which must be temporarily stored in the transmitter for this purpose, until it receives positive confirmation.

The course of this process is illustrated in Figure 1.22. BEC is successfully used in all common protocols such as Signaling System No. 7 (SS7), Link Access Procedure for the D-Channel/Modified (LAPD/$LAPD_m$), or X.25 as a method for error recognition and correction. The following problems do occur, however, when handling BEC:

- BEC assumes that at least the NACK information can pass through the transmission channel without any problems. If this is not the

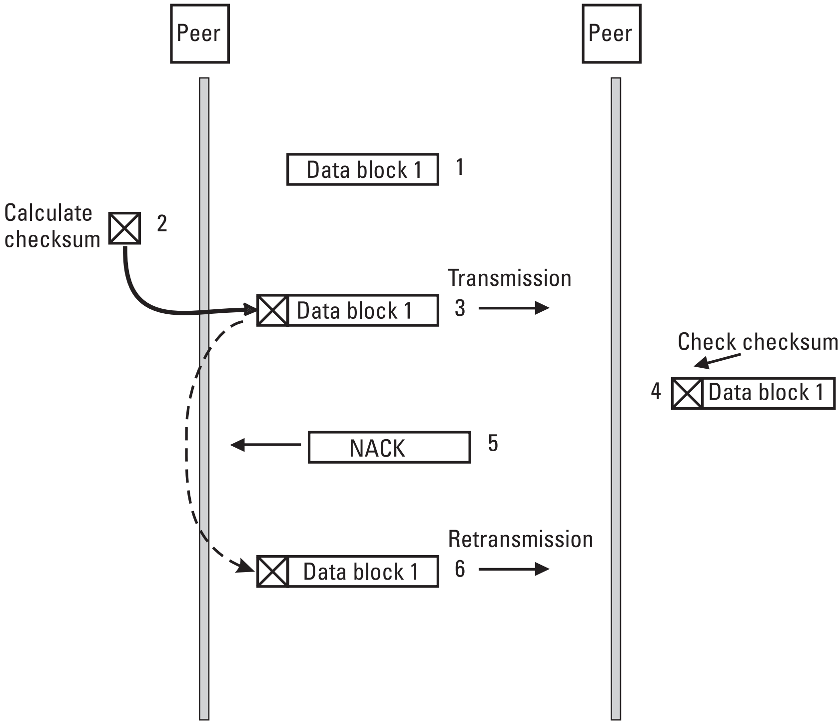


Figure 1.22 The processes of BEC.

case, then an endless loop will occur or the NACK information can even be misinterpreted due to bit errors.

- A transmission error in the checksum field will also be interpreted as an error in the data block, which will then lead to the repeated request for a data block that is in itself error-free.
- Even one single bit error requires the retransmission of the entire data block. This means that resources will be wasted when an error occurs.

It can be said that BEC functions very well as long as the transmission channel is only rarely, and then only slightly, disturbed. These conditions most certainly apply in terrestrial transmission or a fixed wireless link, but not in the field of mobile communication. In any case, the exclusive use of BEC in mobile communication is not advisable.

This means that as GSM was being conceived and developed, additional error correction methods were required, which could be realized using FEC.

1.6.3.2 Introduction to FEC

As the name suggests, FEC functions in exactly the opposite way to BEC. In general it can be said that redundant information is added to a data block via FEC, with whose help the receiver is enabled not only to recognize but also to correct bit errors after transmission—and it can do this without having the data block transmitted again. This capability also depends on the bit error rate not being too high or the coding rate being appropriately chosen. Here, the coding rate expresses the ratio of the number of input bits to the number of output bits in the channel coder. We shall first introduce the process of the channel coding of speech information in GSM, since data will become the dominant subject later.

1.6.3.3 The Processes of Convolutional Coding in Full-Rate Speech

The process of convolutional coding, full-rate (FR) speech, is illustrated in detail in Figure 1.23. The content of a TRAU frame forms the input of the convolutional coder; in FR speech this is exactly 260 bits. It is to be noted that these 260 bits are not homogenous, but rather divided into class 1 and class 2 bits, depending on their importance. The first 50 class 1 bits are the most important of their class, since they contain the decisive presettings for the receiver's speech interpreter. Unrecognized errors cannot be tolerated in this area. For this reason, a 3-bit-long checksum is calculated for the first 50 class 1 bits and attached to them. Note that the receiver will reject the entire TRAU frame if the test of this checksum should result in a bit error.

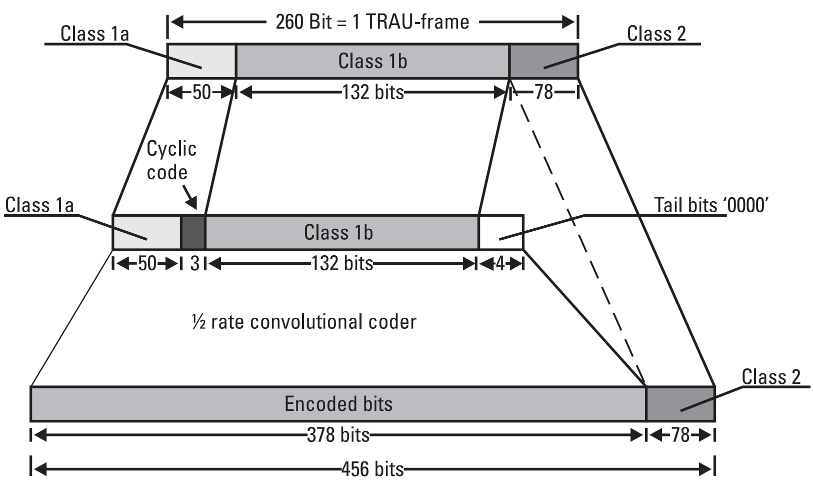


Figure 1.23 The process of convolutional coding for FR speech.

The other 132 class 1 bits are attached to the protected class 1a bits. Before the entire block can be added to the actual convolutional coder, four so-called *tail bits* are added to the end. These tail bits are intended to reset the convolutional coder to its initial status after each frame, which is why it is possible to talk of a reset signal for the convolutional coder (four flip-flops = four tail bits). Note that the four tail bits are actually an unpleasant overhead, which bears down more, the fewer payload bits there are to be transmitted. In other words, the shorter the actual data block is, the more unpleasant the tail bits are. This disadvantage was not eliminated with the standardization of GPRS. It was, however, eliminated with the introduction of EDGE.

In GSM, the convolutional coder is always similar for all applications up to the introduction of EDGE (i.e., also for GPRS). To put it simply, it is a 4-bit shift register whose outputs are connected appropriately. A simple example is illustrated in Figure 1.24. Consequently, the convolutional coder has two outputs, which is why it is called a 1:2 (half) rate convolutional coder. Note that Figure 1.24 does not illustrate the actual convolutional coder, but is merely intended to show its approximate construction. It should also be noted that a half-rate convolutional coder has nothing to do with the GSM half-rate transmission technology.

In our example of full-rate speech coding, there are, at the entry to the convolutional coder, 50 class 1a bits with 3 bits checksum, plus 132 class 1b bits, and finally 4 tail bits, making 189 bits in all. According to the rules presented above, the convolutional coder doubles the number of these bits, and the above-mentioned redundancy is added to the data block. At the end of the process chain, the class 2 bits are simply attached to the convolutionally coded class 1 bits. In other words, there is no convolutional coding for the class 2 bits as they remain unprotected.

A convolutionally coded block thus consists of exactly 456 bits. Although we will be anticipating the later section on bursts, it should be

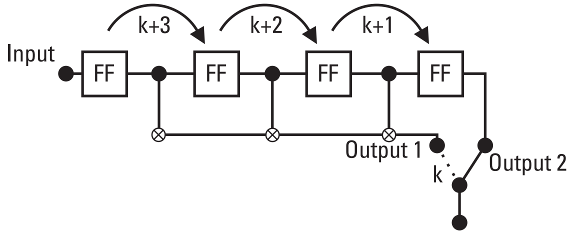


Figure 1.24 Example of a 1:2-rate convolutional coder.

mentioned at this point that a normal burst can transfer 114 bits (i.e., exactly four bursts are required to transmit a convolutionally coded speech block). Note here that the data flow in GSM and GPRS always aims for these 456 bits, and four bursts are always needed for the 456 bits, independently of the type of data and data rate.

1.6.3.4 Effects of Convolutional Coding

It has already been mentioned that the aim of convolutional coding is to recognize and correct transmission errors at the receiver's end. The decoder does not only have a special sort of intelligence but also determines probabilities for the next bit value from the course of the previous bit values. This process can be explained as follows.

A 4-bit shift register, in which the bits are also linked up, increases the presence of an individual bit eightfold. Why? Because each individual bit must pass through the entire shift register and is linked up to the previous and following bits during this time.

Here, a single bit is spread over a wider area using convolutional coding. For example, the bit sequence 11 00 at the input could be imagined as 11 01 10 00 at the output.

Defective bit combinations can thus be recognized as such by the receiver. The convolutional decoder (the Viterby decoder in GSM) recognizes and corrects these bit errors. But even the capacity of the Viterby decoder has its limits. The Viterby decoder is especially powerless when there are errors in many consecutive bits (bursty bit errors) because there is no possibility of determining probabilities for consecutive defective bits.

Let us look at Figure 1.25. The single bit error in A (0) is not a problem, and the Viterby decoder can easily interpolate a 1. This is no longer possible for B, since there is a series of consecutive bit errors.

The problem here is that mobile communication is particularly subject to sudden and short disturbances in reception, which will lead to many consecutive bit errors. Consequently, convolutional coding cannot make full use of its strengths in mobile communication without further measures. But what are these further measures?

The answer is actually quite easy: trick the receiver into thinking that these errors are also single bit errors. In addition, the already-coded information is mixed up to varying degrees. The technical term for this process is *interleaving*. We will examine this subject in more detail in Section 1.6.4.

1.6.3.5 Puncturing and Other Convolutional Coding Methods in GSM

Convolutional coding for FR speech has already been presented in detail. Although we were able to treat the principle of convolutional coding in

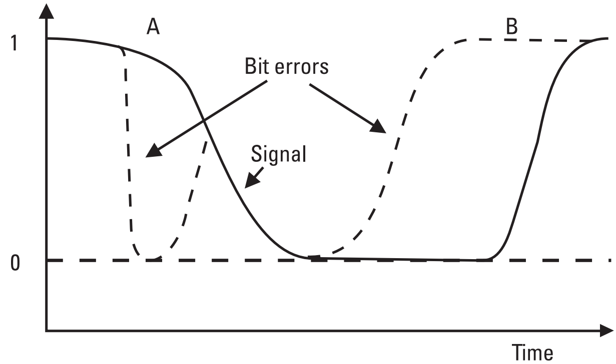


Figure 1.25 Single and multiple bit errors at the receiver.

detail at that point, we have not yet addressed a phase in convolutional coding that is absolutely decisive for GPRS.

This is the topic of puncturing, which we will discuss generally before going into specific detail in later chapters.

Let us note in advance that in many of the GPRS courses available to us, especially in "cultural circles" outside code division multiple access (CDMA), the subject of puncturing causes a great deal of astonishment, or is even frowned upon. Even acknowledged GSM experts are amazed that puncturing has already been used in normal GSM. This experience further underlines the complexity of GSM. Before we go into detail, however, we should examine what puncturing actually is. Imagine a convolutionally coded data block in GSM that is bigger than the aforementioned 456 bits, let us say 488 bits. What happens to these 32 bits too many?

Incredible as it may sound, these 32 bits are simply not transmitted.

Puncturing is thus nothing more than the deliberate nontransmission of the punctured bits. These are not the first or last 32 bits in a block but the *same* bit numbers in each block (e.g., bit numbers (11), (26), (41), . . .).

To continue with our example from above: If puncturing is carried out, transmitter and receiver know exactly which bit positions have not been transmitted. Although puncturing basically requires an interpolation of the bits that have not been transmitted, knowing where these bit errors occur in the data block makes correction considerably easier.

Puncturing is also a simple way of enabling a more flexible coding rate than is possible with a convolutional coder. The coding rate is obtained by dividing the number of bits at the input of the convolutional coder by the number of bits actually transmitted. Without puncturing, the result is a half-rate convolutional coder or a coding rate of one-half for most channels in GSM because exactly double the number of bits is produced.

It is even possible to have convolutional coders that generate 3, 4, or 5 bits at the output, but it is not possible to have one that generates 1 2/3 bits at the exit. This is where puncturing enables more flexible coding rates.

As already mentioned, normal GSM also uses puncturing. Indeed, the transmission of the GSM data channels (TCH/F 9.6 Kbps and TCH/F 14.4 Kbps) actually require puncturing because behind the convolutional coder there are 488 bits and 588 bits to be transmitted, respectively. GSM 05.03 defines precisely which bits per channel type or coding method have to be punctured.

1.6.4 Interleaving and Burst Generating

It has already been mentioned in Section 1.6.2.4 that consecutive bit errors can make active correction at the receiver difficult, or even prevent it entirely. To alleviate this situation, a method is usually used that is called interleaving. Depending on the data type [i.e., on whether speech, data, or signaling (SACCH, FACCH, . . .) is contained in a data block], the bits of a convolutionally coded data block are mixed, either with each other or with bits from previous or following blocks. The effect of interleaving can be understood when taking the above-mentioned facts into account. If consecutive bit errors occur during transmission, these are treated by the receiver as single bit errors after deinterleaving.

Figure 1.26 shows an example of interleaving for speech, in which the bits of a convolutionally coded block are mixed with those of the previous and following blocks. For speech, eight (not four) bursts are required for

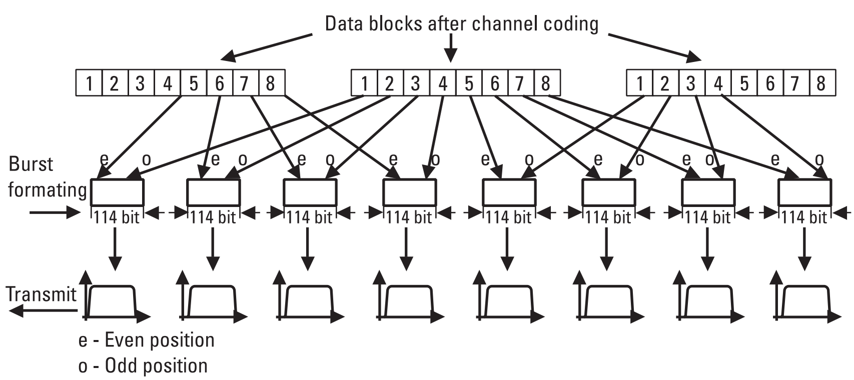


Figure 1.26 Interleaving for speech information.

transmitting the 456 convolutionally coded bits. However, it can also be said that instead of four TDMA frames (about 18.46 ms), exactly twice as much time (36.92 ms) is required for transmission. In other words, if consecutive blocks are mixed with each other in interleaving, the application must be able to handle extra delay times.

In interleaving for speech, this delay is relatively moderate because speech, by nature, requires real-time transmission; otherwise the receiver will gain the impression of a satellite connection.

With data transmission, however, a convolutionally coded block is not divided into eight bursts, as is the case with speech, but into 22 consecutive bursts. Consequently, the time delay caused by interleaving is very high with data transmission (total transmission time per block = 101.53 ms). However, data is much less sensitive to delay times than speech but much more sensitive to bit errors. It is clear that the more data blocks are intermixed with each other, the greater the positive (and negative) effects of interleaving are.

1.6.4.1 Interleaving in Signaling Channels and GPRS

There are also cases in which, unlike speech and data, there is not an unlimited number of bursts available for transmission in a traffic channel. Especially when transmitting signaling information on the SACCH or SDCCH, only four bursts are available per message transmitted. For this reason, the 456 convolutionally coded bits are only mixed with each other for interleaving in SACCH or SDCCH. This also applies to GPRS, as will be shown in the following sections. For this reason, GPRS uses the same interleaving method as described for the SAACH in GSM 05.03. In other words, GPRS does not introduce a new form of interleaving.

1.6.5 The Encrypting Function in GSM (Ciphering/Encryption in GSM)

After interleaving, the convolutionally coded and mixed information in the base band is ready for actual modulation and transmission. This data, however, is still not encrypted after interleaving, which is why the process of encrypting should now be examined.

Data encryption in GSM is based on the presence of the same key (Kc) in the VLR and in the mobile station. This key is determined by the mobile station during authentication. In other words, encryption in GSM is impossible without prior authentication. The period of time that has elapsed since authentication is initially a secondary matter. As already described in Section 1.1.2.1, authentication and calculation of the key ensues by means of the A8 algorithm, either at the beginning of every transaction (location update, call setup) or only once when a subscriber initially registers on a VLR area. All this is dependent on configuration and can be set by the network operator on the VLR. If authentication only takes place at initial registration in a VLR area, the use of the same key in the mobile station and the VLR must be secured by comparison of the ciphering key sequence number (CKSN) before encryption is activated. CKSN can be transferred unencrypted and is only a reference to the Kc that is to be activated.

In any case, the use of encryption in GSM is controlled functionally by the VLR but executed by the BTS. When the encryption is activated, the Kc and the A5/X algorithm to be used are communicated to the BTS by the VLR, whereas the mobile station only finds out which A5/X algorithm is to be used via the air interface. In other words, the Kc will not be transmitted via the air interface under any circumstances.

The encrypting function itself is based on the use of the A5/X algorithm in the mobile station and the base station. This is illustrated in Figure 1.27. The X stands for seven different Xs, in the mobile device, the A5/1 and A5/2 must be implemented, while A5/3 to A5/7 are optional. It is important that A5/1 and A5/2 are actually stored in the device and *not* in the SIM. In addition to the Kc, A5/X also uses the FN as a dynamic entry parameter, which changes once per TDMA frame or every 4.615 ms. Note that this means that the ciphering sequence changes with every burst to be transmitted because the FN also changes with each burst.

This also makes it clear why the introduction of the hyperframe into the frame hierarchy is advantageous to encryption, offering the longer frame

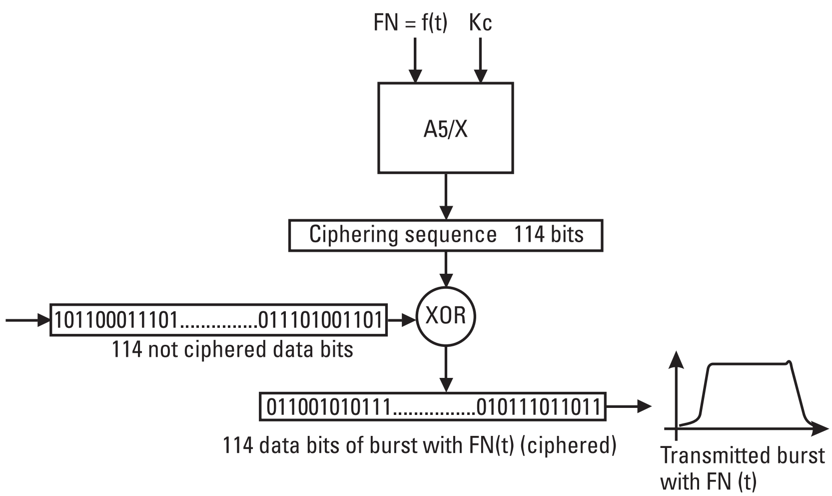


Figure 1.27 The process of data encryption in GSM.

number and greater security due to a considerably longer cycle. Otherwise, the FN would be repeated every 6.12 sec and not every 3 hours 28 minutes.

Before encryption and after interleaving, the 456 convolutionally coded bits of a block are in the correct order and ready for actual transmission. These 456 bits are added to the A5/X algorithm in portions of 114 bits. It is exactly 114 bits because every normal burst can transport 114 bits of payload.

The actual encryption takes place by means of a simple exclusive or (XOR) link of the 114 unencrypted bits with the 114 bits of the ciphering sequence. The encrypting function in GSM is thus relatively easy to follow and, at the same time, highly effective. The deciphering process takes place in precisely the reverse order and is illustrated in Figure 1.28. Here it is clear that a correct XOR link or deciphering is only possible when the correct ciphering sequence is available. Otherwise, the subsequent data processing will not produce any comprehensible information.

1.6.6 Burst Forming and Modulation

At the beginning of Section 1.6, the signal processing chain of GSM was described. Following encryption, which is optional, burst forming takes place, then modulation, and finally actual transmission via the air interface.

Burst forming itself is a normal mapping of the encrypted bits onto the actual bursts. The burst types in GSM, however, require detailed consideration.

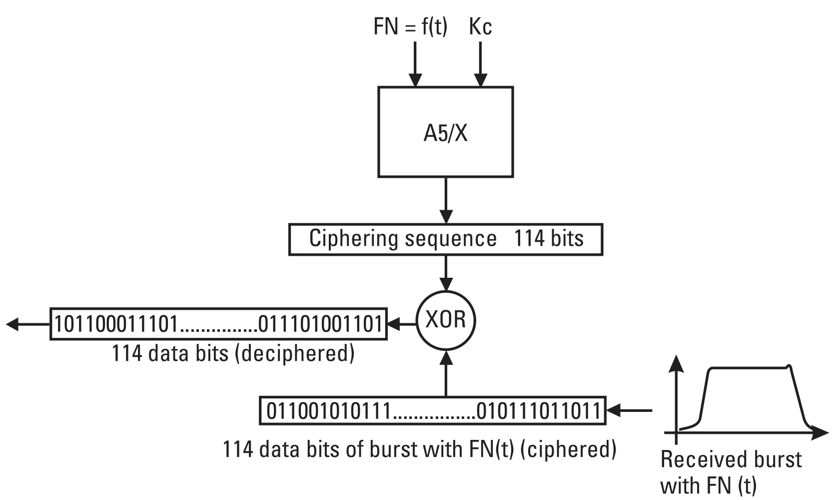


Figure 1.28 The process of deciphering in GSM.

1.6.6.1 The Burst Types in GSM and GPRS

For GSM a total of five different burst types have been defined, of which three can only be used in the downlink direction and one only in the uplink direction. Only the so-called normal burst can be used for both the uplink and the downlink directions. These definitions also apply to GPRS. The following list introduces the individual bursts and explains the most important details.

The Frequency Correction Burst

The frequency correction burst (FB) consists of 142 consecutive 0 bits and the respective 3 tail bits at the beginning and the end, which are all coded with 0 (Figure 1.29). The FB can only be used in the FCCH (i.e., in the downlink direction). In this context it is also worth mentioning that because of the GMSK modulation used in GSM and the 142 0-coded bits, a pure sine wave is produced that can be found exactly 67.7 kHz above the carrier frequency.

When considering the 51 multiframe described in Section 1.5.1 and the use of the FCCH, it becomes clear that a mobile station, when selecting its cell, will first seek out a strong signal in the respective frequency band without any qualitative evaluation. If this search is successful, the mobile station will then try to find a 577-μs impulse exactly 67.7 kHz above the carrier frequency every 46.15 ms (10 TDMA frames). If this signal cannot be found, it is then clearly not a GSM carrier. If the FB is found, then it will not only positively identify a GSM carrier for the mobile station but will also help it to adjust its own frequency generator.

The Synchronization Burst

The synchronization burst (SB) is, like the FB, a type of burst that is only used in the downlink direction (Figure 1.30). In contrast to the FB, however, the SB is used on the synchronization channel, which is always transmitted on time slot 0 of the BCCH carrier exactly one TDMA frame later than the FCCH, as shown in Section 1.5.1. The SB has a number of specific characteristics. The so-called extended training sequence is found at its center (midamble). This extended training sequence is a series of bits, always 64

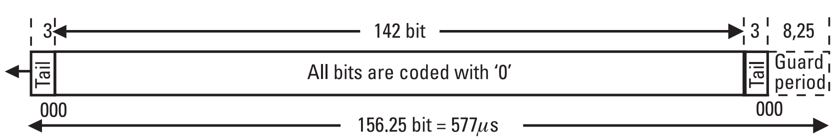


Figure 1.29 The frequency correction burst.

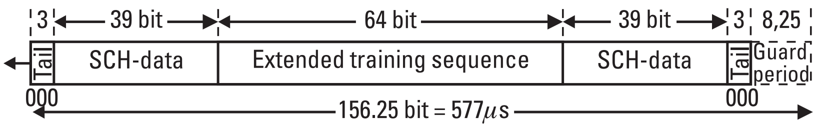


Figure 1.30 The synchronization burst.

bits in length, which is used by all BTSs in the various GSM families. In particular, the extended training sequence serves the mobile station in chronological fine synchronization with the base station. This must take place at a precision of at least one-quarter of a bit period—that is, at 1 μs (1 bit period = 577 μs/156.25 bits = 3.692 μs).

In addition to the extended training sequence, the synchronization burst also contains real data, especially the FN (i.e., the current time of a base station within the frame hierarchy) as well as the base station identity code (BSIC/6 bits) of a BTS. The latter is composed from the network color code (NCC/3 bits) and the base station color code (BCC/3 bits). Further details on BSIC, NCC, BCC, and FN can be found in [1, 2].

The Dummy Burst

The dummy burst (DB) is not widely known because it cannot be used for any kind of information transmission (Figure 1.31). The dummy burst owes its existence to the requirement that the BCCH carrier of a base station must be operated at a constant output level independently of whether all time slots are occupied or not. If individual time slots of the BCCH carrier are temporarily unoccupied, the BTS transmits dummy bursts from these time slots. This rule also applies to the idle frames in the 51 and 26 multiframes. The dummy burst is thus a pure downlink burst, which is not used in the uplink. The dummy burst uses a fixed bit sequence, called mixed bits.

The Access Burst

The access burst (AB) is the only burst used exclusively in the uplink direction. Before the introduction of GPRS, the only channels to use the access burst

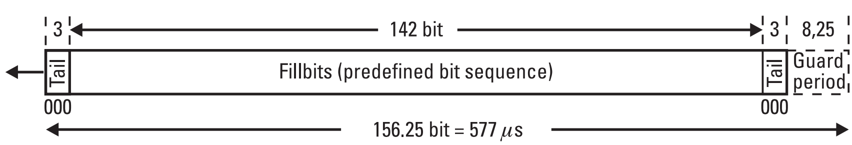


Figure 1.31 The dummy burst.

were the FACCH (only at handover) and the random access channel (RACH). The mobile station must use the access burst when there is no information available at the current distance to the base station. This is clearly the case at initial access to a BTS and during a handover. This principle has already been introduced in Section 1.4.1. What is noticeable about the access burst is the shortness of its actual length. The access burst is, without its excessively long guard period, only 88 bits long as compared to the 148 bits + guard period of the other burst types. However, it is precisely this short length that enables an access burst to be sent from a mobile station to a base station at a distance of up to 35 km, using the slotted Aloha method (assumed distance of 0 km) without colliding with the next receiving window. The 41 bits of the synchronization sequence, always coded the same, enable the base station to identify an access burst arriving at a given time and estimate the distance from the mobile station. In order to distinguish an access burst from possible background noise, there are also, in addition to the synchronization sequence, color bits (see Figure 1.32), which contain, amongst other things, the BSIC of the target BTS.

Due to its short length, an access burst can only contain very little information. This includes the 41 bits of the synchronization sequence, always coded the same, and the 36 bits of payload information. Figure 1.33 states 36 payload bits, but it must be remembered that this is already convolutionally coded information. The number of actual data bits is thus accordingly low.

An access burst really only contains 8 bits of payload or, new in GPRS, an option of 8 or 11 bits. The precise contexts will be explained below. What is surprising, however, is that with GPRS an access burst can contain 8 or 11 bits of payload. The network must, of course, constantly monitor

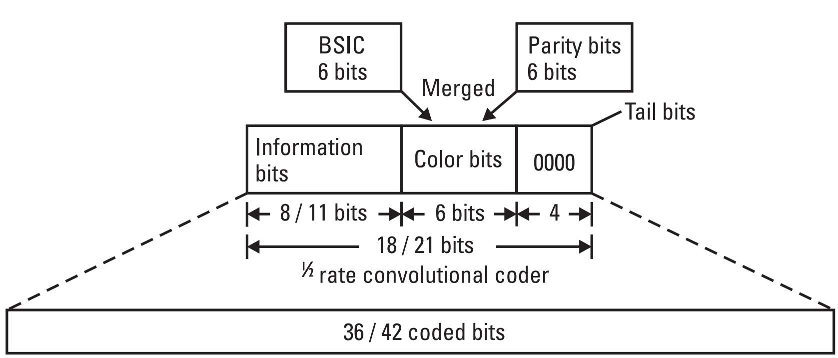


Figure 1.32 The convolutional coding of access bursts.

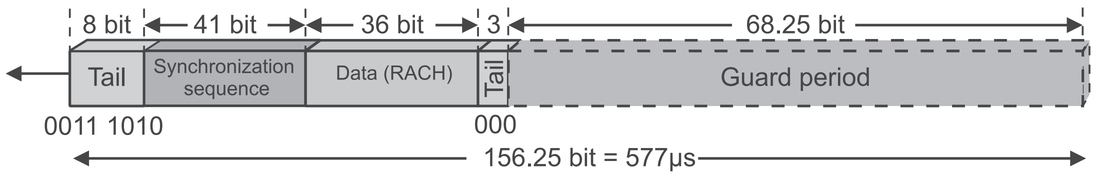


Figure 1.33 The access burst.

this. The *access_burst_type* parameter, transmitted in the system information, informs the mobile stations parked on a BTS about the access burst type to be used.

Convolutional Coding of Access Bursts in GSM and GPRS. The process of convolutional coding of access bursts for GSM and GPRS is illustrated in Figure 1.32. The 8 or 11 information bits are first expanded by the 6 color bits. The color bits consist of the result of a bitwise XOR link of the BSIC (6 bits) with the parity bits (6 bits), which are in turn generated from the information bits. Before convolutional coding actually takes place, 4 tail bits are added at the back, which set the convolutional coder to a defined final status, as shown in Section 1.6.3.3.

Convolutional coding doubles the number of input bits, which means that 18 bits become 36 bits, and 21 bits become 42 bits. The difference of 6 bits is determined by the 11 or 8 bits of information at the input. In any case, 42 bits do not fit into an access burst, which means that six of these bits are not transmitted in accordance with the puncturing principle as described in Section 1.6.3.5.

The Normal Burst

In general terms, the normal burst is used by the BTS and the mobile station for transmission on all types of channel that have not yet been mentioned in Section 1.6.6.1 (Figure 1.34). These include channel types such as the BCCH, SACCH, CBCH, or TCH. The mobile station may only use the normal burst after valid timing advance information has become available. Every normal burst transmits exactly 114 bits of convolutionally coded payload to the receiver, which are ordered in two packets of 57 bits each around the only 26-bit long training sequence in the normal burst. Consequently, four bursts of 114 bits of transfer volume are required to transport a convolutionally coded block of 456 bits in each direction.

Eight different training sequence codes have been defined for the normal burst. The BCC transmitted in the synchronization burst of a BTS states which of these TSCs is used on the BCCH and the CCCHs. The TSC

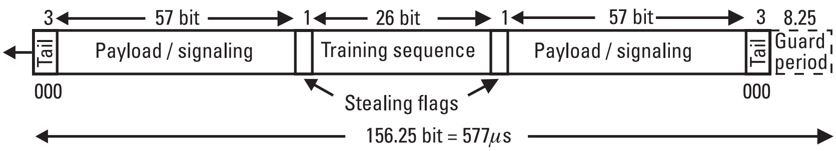


Figure 1.34 The normal burst.

basically has two functions. First, it allows the base station to make continuous measurements and possibly make changes in transmission time in the uplink direction that arise due to increasing or decreasing distance from the mobile station and require adjustment to the timing advance. Second, the TSC serves as a reference for bit error estimation in the uplink and downlink directions.

In GSM, the two stealing flags are only of significance on the TCH. They state whether any parts of a burst have been stolen for inband signaling (FACCH) and, if so, which ones. As we will describe, stealing flags are used for another purpose in GPRS.

1.6.6.2 GMSK Modulation

Until EDGE is introduced, GSM only uses GMSK as its modulation method. Note that this also applies to GPRS. In terms of effort and complexity, GMSK is a very simple modulation method that is also tolerant of errors. Only 1 bit is transferred per symbol. GMSK can also be applied without amplitude modulation (AM), which keeps the cost of HF amplifiers low. All these characteristics led to GMSK being chosen as the modulation method for GSM, although as already stated, it is relatively slow (1 bit per symbol).

In general, GMSK is a special form of frequency modulation (FM), or rather its binary variant, frequency shift keying (FSK). In FSK, the output signal is switched to and fro between the two frequencies, $F_T + f_t$ and $F_T - f_t$, depending on the value of the bit to be transmitted.

The disadvantage of pure FSK is the frequent, drastic frequency change that can theoretically occur with every new bit, thus leading to increased bandwidth requirement. One way of avoiding this constant change of frequency (i.e., minimizing the number of frequency changes) is the logical XOR combination of two consecutive data bits before modulation. As shown in Table 1.1, bit (N) and bit ($N + 1$) are combined.

As can clearly be seen in Figure 1.35(b), this method indeed results in a reduction in the number of status changes for most bit combinations.

Table 1.1
Truth Table for MSK Modulation

bit N	bit ($N + 1$)	XOR	Frequency
0	0	0	$F_T + f_t$
0	1	1	$F_T - f_t$
1	0	1	$F_T - f_t$
1	1	0	$F_T + f_t$

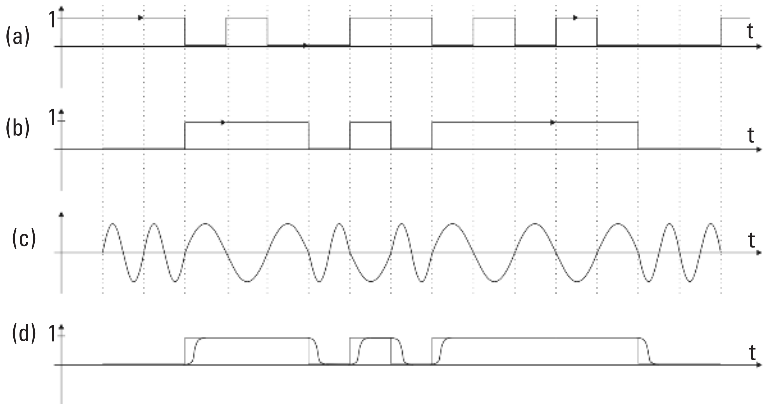


Figure 1.35 The path from FSK to GMSK: (a) Bit stream, (b) bit stream after XOR operation, (c) MSK signal, and (d) behind Gaussian filter.

Only the combination 11 00 11 00 . . . leads to an increased number. By using the XOR operation, FSK becomes the so-called minimum shift keying (MSK).

Although in MSK the number of frequency changes is reduced as compared to FSK, every frequency is still "hard" and thus requires a great deal of bandwidth. Further optimization is possible by leading the digital bit series, after the XOR operation, to a bandpass with a Gaussian characteristic, which leads to looping of the status junctions as shown in Figure 1.35(d). In the subsequent modulation the switching between the two frequencies, $F_T + f_t$ and $F_T - f_t$, takes place slowly or via all intermediate frequencies. This measure led to MSK becoming GMSK as used in GSM.

The Gaussian filter used in GSM displays the following characteristics:

$$B \times T = 0.3$$

where B = 3 db bandwidth and T = bit duration = 577 μs/156.25 bits = 3.693 μs. At the I/Q level, a delta phase modulation is produced for GMSK, as shown in Figure 1.36. Depending on the input bit, the phase of the modulated signal is rotated by ±90°. An animated version of this illustration can be downloaded from our Web site (www.inacon.com).

The frequency fluctuation is derived from the following formula:

$$f_t = (\text{data rate} \times h)/2$$

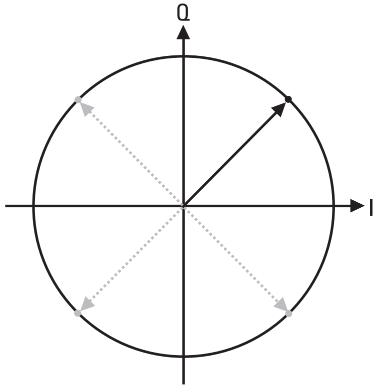


Figure 1.36 GMSK on the I/Q level.

where data rate = 1/bit duration = 1/3.963 μs = 270.833 KHz, and h = modulation index = 0.5.

Consequently, the result is $f_t = 270.833 \text{ KHz} \times 0.5 \times 0.5 = 67.7$ KHz. It also becomes clear after this calculation why the FCCH can be found 67.7 KHz above the actual carrier frequency. In accordance with the truth table (Table 1.1), a series of 0 bits lead to a constant sine wave at $F_T + f_t$ (i.e., exactly 67.7 KHz above the carrier frequency).

1.7 Data Services in GSM

Even during standardization, a whole variety of data services belonged to the envisaged range of GSM services. In the 1980s and early 1990s, however, mobile data services were something rather exotic. At the beginning of the 1990s, however, mobile faxing was of some interest to a larger number of customers. This situation has not really changed since GSM was put into operation. The illustration in Figure 1.37 makes this situation clear. In today's Internet era, a great majority of mobile communication customers are still using GSM exclusively for speech services.

The most important examples of GSM data services before the introduction of GPRS are the following:

- Circuit switched data (CSD), with various channel types and throughput rates ranging from 300 bps to 14.4 Kbps. CSD is available in transparent (without inband error recognition and correction) and nontransparent (with error recognition and correction)

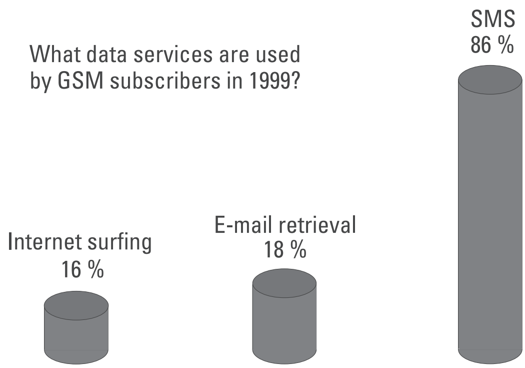


Figure 1.37 Use of data services in GSM in 1999.

form. As a rule, the nontransparent version is used in order to keep the error rate low. In the following, the TCH/F 9.6 Kbps is introduced in detail as an example of CSD;

- Fax/Group 3 with 9.6 Kbps;
- Unstructured supplementary services data (USSD) for the transmission of text messages between the mobile station and a proprietary network operator application. The WAP can also use USSD as a carrier service;
- Short Message Services (SMS) for the transmission of text messages in both directions between an SMS service center (SMS-SC) and the mobile station (Figure 1.38);
- Short message service cell broadcast (SMS-CB) for the transmission of regionally relevant short messages in broadcast operating mode to all mobile stations in a given region;
- WAP, which can present Internet-like contents on the display of a mobile station. WAP is not a GSM data service as such, because it

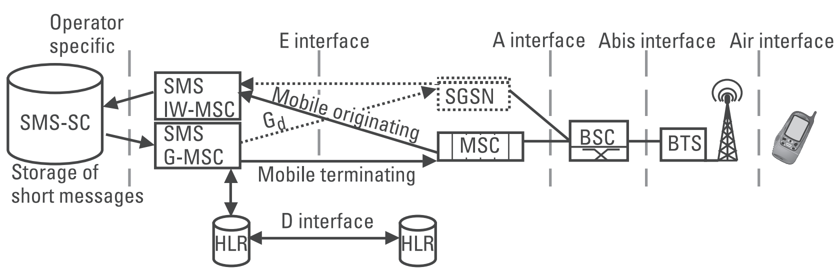


Figure 1.38 The transmission of short messages in a combined GSM/GPRS network.

uses other GSM data services such as CSD as carriers for the transmission of binary WAP contents;

- HSCSD, which concentrates several traffic channels in order to enable greater bandwidths than CSD alone.

1.7.1 CSD (TCH/F 9.6 Kbps)

As can be seen in Figure 1.38, channel switched GSM data services are, with the exception of SMS, used mainly for the transmission of e-mail and for surfing the Web. These are also the applications that are primarily targeted by GPRS.

For this reason, this introductory chapter shall end with a detailed examination of the processing and transmission of the channel switched GSM data channel, TCH/F 9.6.

First of all, it must be made clear that the TCH/F 9.6 is always bidirectional (i.e., it offers a symmetrical service and is principally allocated to control channels), together with the SACCH and FACCH, which are also bidirectional. We are stressing these finer points in order to highlight the difference to the unidirectional TCH/FD 9.6. This is, however, only defined for HSCSD, and then only in the downlink direction. The TCH/FD 9.6 contains a unidirectional SACCH, but no FACCH.

1.7.1.1 The Signal Course of the TCH/F 9.6 Kbps

Figure 1.39 illustrates the signal course of the TCH/F 9.6 in the GSM network and via all interfaces. The frequent data rate adaptation (RAA, RA 0, . . .) is described especially clearly. A whole range of these rate adaptations has been defined for the various channel types and transmission speeds. The individual processing steps will be explained in more detail below. We shall start with G-MSC/IWF.

- *Step 1: The RAA/conversion into 80-bit V.110 frames.* V.110 is an ITU-T standard that amongst other things defines a frame format for data and regulates synchronization during data transmissions. Note that there are different types of V.110 frame. At this juncture we wish to present one type only, the 80-bit format that is most widely used. In Figure 1.40 it becomes especially clear that of these 80 bits, only 48 contain payload. Taking into account Figures 1.39 and 1.40, we arrive at the following result: 48 bits ÷ 5 ms = 9.6 Kbps.

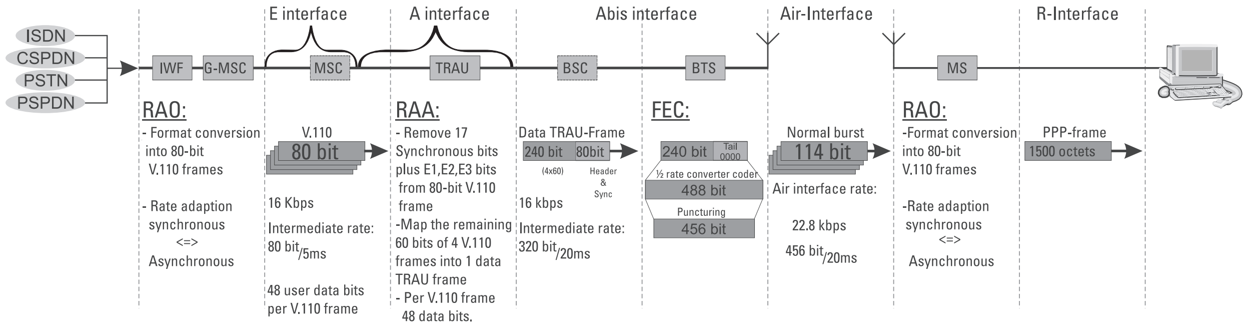
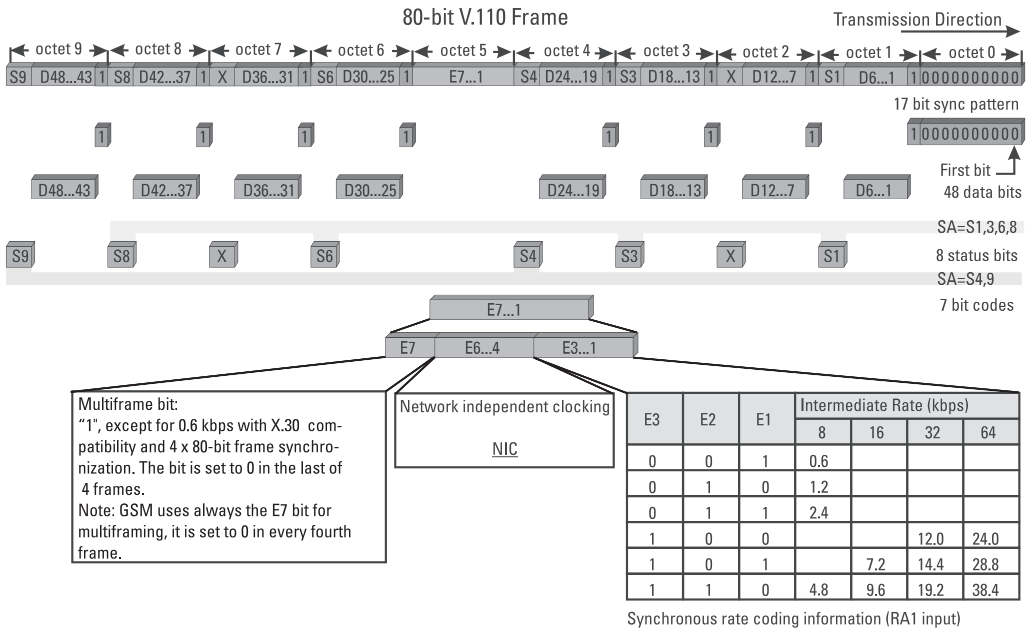


Figure 1.39 The processing steps for the TCH/F 9.6 in the GSM network.



E3	E2	E1	Intermediate Rate (kbps)			
			8	16	32	64
0	0	1	0.6			
0	1	0	1.2			
0	1	1	2.4			
1	0	0			12.0	24.0
1	0	1		7.2	14.4	28.8
1	1	0	4.8	9.6	19.2	38.4

Figure 1.40 The format of the V.110 frame/80 bits.

- *Step 2: The RA 0 (Rate Adaptation 0)/TRAU functions during data transmission.* In the TRAU, the 17 synchronization bits and bits E1, E2, and E3 are removed irreplaceably from the V.110 frame in order to reduce the gross data rate.

  There are then 60 bits remaining per V.110 frame. Four consecutive V.110 rump frames are then gathered in the TRAU into a data TRAU frame with a length of 320 bits (i.e., content = 4 × 60 bits = 240 bits) and then transferred further to the BTS. Here, a data TRAU frame is transmitted every 20 ms. The data rate here is thus 320 bits ÷ 20 ms = 16 Kbps.
- *Step 3: FEC.* Before actual transmission via the air interface, redundancy is added to the content of the data TRAU frame in terms of convolutional coding (1:2 rate convolutional coder; see Section 1.6.3). The 240 bits per TRAU frame plus 4 tail bits thus become 488 bits. By means of puncturing (see Section 1.6.3.5), 32 bits are deleted from this data block, which means that ultimately, 456 bits are transmitted via the air interface in 20 ms (22.8 Kbps). I would like to ask those readers who are now wondering why four TDMA frames at 4.615 ms = 20 ms to send me an e-mail (gheine@inacon.de).

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2

Introduction to GPRS

2.1 The History—From GSM to UMTS

Up until the mid-1990s, the Internet was predominantly used by international universities, research institutions, and U.S. government offices. It was at this time, however, that the Internet began to penetrate the consciousness of a broader public. Opening up to the mass market, the Internet became an increasingly important facility for databases, information hosting of all kinds, and especially e-mail transfer. It may be recalled that this was also about the time when GSM subscriber numbers were also beginning to increase dramatically.

The mobile Internet user found the very slow speed of GSM data channels (a maximum of 14.4 Kbps) a painful experience. Furthermore, mobile Internet users had the disadvantage of having to dial themselves in for every Internet link. Although this is true for home use as well, many users are accustomed to the famous always-on Internet link in their offices. You can, of course, object that it is still possible to surf for hours on the Internet, whether at home, in the office, or on the road. However, one will soon be convinced of the opposite, if by nothing else, by the time the next GSM network operator bill arrives. Only business customers who must be able to access the company's intranet or their e-mail from outside their office are prepared to accept the high GSM connection costs.

There was then little of more importance than to address the two disadvantages: the low throughput rates of the data and the unavailability of an always-on connection in GSM by expanding the GSM standard.

Another thought was to merge the two growth areas of Internet and mobile communication. Various technical ideas resulted from these considerations, which, in the end, formed the next step towards third generation mobile communications. The various stages of development of mobile communications and the technologies behind them are illustrated in Figure 2.1. The following sections are an introduction to these technologies.

2.1.1 HSCSD

HSCSD is intended to offer the user greater bandwidths by means of a concentration of several GSM data channels. Up to 64 Kbps (i.e., ISDN speed) is theoretically possible. In the end, 38.4 Kbps was used. As a comparatively small extension of GSM with a short development time, HSCSD was only supposed to be an additional intermediary step to other options. Note in particular that HSCSD does not address the problem of always on. The user simply receives more resources.

In its role as a short-term solution to the above-mentioned problems, however, HSCSD had three problems:

1. The requirement for new mobile stations is opposed to the relatively low effort needed on the network side.
2. Network operators were concerned that heavy use of HSCSD would use up the limited air interface resources and that this would have a negative effect on their main business (i.e., telephone calls).
3. The development of usable HSCSD network expansions and mobile stations was taking too long. Standardization began in 1994 but HSCSD was not available for use until 1999. By this time, the other alternatives for high-speed mobile data had already become close to realization.

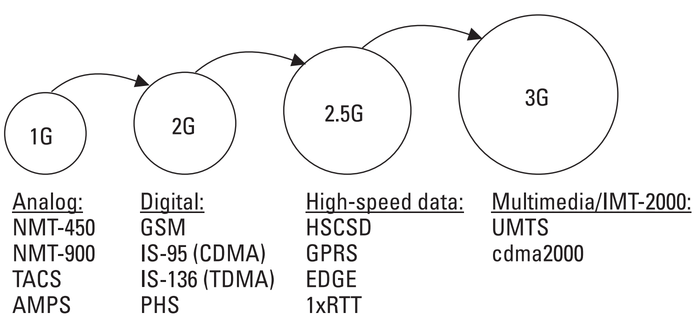


Figure 2.1 Mobile communications generations 1 to 3.

2.1.2 GPRS

GPRS, like HSCSD, is intended to achieve greater bandwidths by concentrating time slots together. As a real innovation as opposed to HSCSD, GPRS also allows for variable code rate settings, depending on receiving conditions. Accordingly, greater or smaller numbers of redundant bits are added to the net data. Consequently, GPRS enables transmission speeds of up to 160 Kbps.

The most important innovation of GPRS, however, is not the higher transmission speeds, but its packet switched characteristic. By means of new protocols, GPRS enables almost simultaneous use of the same time slot by several users. In particular, GPRS allows resource allocation as required, or resource on demand. One or more channels are only then allocated to a user as and for as long as they are actually required. This means that better use is made of available network resources.

2.1.3 Enhanced Data Rates for GSM or Global Evolution

EDGE is not a third alternative for the introduction of high-speed mobile data. There are actually several variations of EDGE, as shown in Figure 2.2.

2.1.3.1 Classic EDGE

Classic EDGE is the logical further development of GPRS and HSCSD for even higher throughput rates. With the introduction of Classic EDGE, GPRS and HSCSD mutate to enhanced GPRS (EGPRS) and enhanced

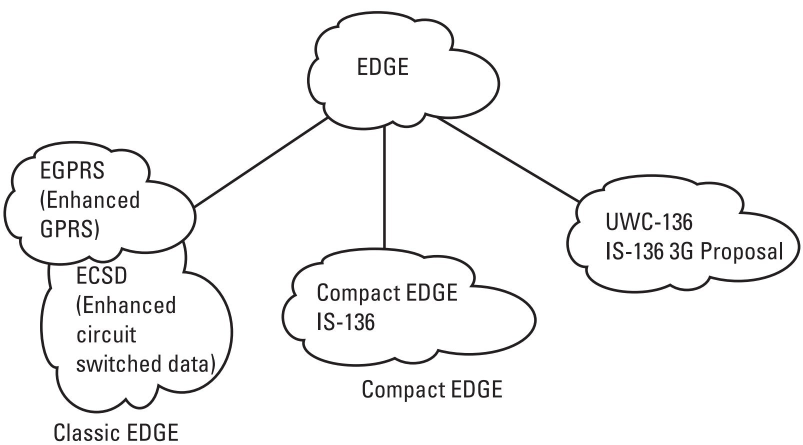


Figure 2.2 The variations of EDGE.

circuit switched data (ECSD). The increased throughput rates are made possible by the use of a new modulation method, 3/8 π offset 8-PSK. In contrast to GMSK, which only has a speed of 1 bit per symbol, 8-PSK allows the transmission of 3 bits/symbol. Assuming the same symbol rate, this is the reason for EDGE's trebled transmission speed. Theoretically, using EDGE can achieve throughput rates of up to 480 Kbps per absolute radio frequency channel number (ARFCN) (3×160 Kbps = GPRS). The negative aspects of using 8-PSK shall be omitted at this stage.

In addition to these increased throughput rates, Classic EDGE introduces more intelligent retransmission methods that do not yet exist in GPRS and HSCSD.

The chances for the development of ECSD-enabled networks are not good. This is on account of the domination of GPRS and the lack of market penetration of HSCSD. On the contrary, manufacturers and network operators are moving towards EGPRS. By using real-time capable voice-over-IP (VoIP), the EGPRS networks of the future should not only be able to transport data but should also transport speech. This will enable better use of resources in the existing GSM networks.

2.1.3.2 Compact EDGE and UWC-136

Compact EDGE is not a GSM standard. This explains the alternative expression for EDGE: Enhanced Data Rates for Global Evolution as opposed to Enhanced Data Rates for GSM. Compact EDGE is more of a possibility for the further development of the U.S. TDMA standard, IS-136, to high-speed mobile data. With a worldwide market share of only 7%, it is difficult for IS-136 to make its own way on the market. Although these attempts were made with the upgrades IS-136+ (43.2 Kbps) and IS-136 HS (high speed; 384 Kbps), in the end the resources were too scarce for independent development. The answer to this problem is Compact EDGE, which was standardized mostly by Ericsson. To put it simply, Compact EDGE is an overlay network to IS-136 that is based on the EGPRS standard (i.e., on GSM). It is the North and South American IS-136 network operators in particular who are interested in Compact EDGE. Compact EDGE, however, also suffers from resource problems in the frequency range and especially the unclear path of development towards 3G. There is an appropriate 3G proposal from the IS-136 community, or the Universal Wireless Convergence Consortium (UWCC), in the form of UWC-136. This can make available speeds of up to 2 Mbps via 8-PSK modulation and using a 1.6-MHz carrier. UWC-136 also requires several thousand man-days for standardizing and development work, which no one is prepared to sacrifice at present.

2.1.3.3 The GSM/EDGE Radio Access Network

The GSM/EDGE Radio Access Network (GERAN) is the synthesis of the existing and future GSM/GPRS/EGPRS access networks set in the wider context of 3G mobile communications. By means of various modifications, the former network switching subsystem becomes the intelligent core network, which provides most of the services. Various radio access networks (RANs), including GERAN and UMTS Terrestrial Radio Access Network (UTRAN), can be connected to this core network. The term GERAN thus came to be used as a differentiation to 3G RANs. This context is illustrated in Figure 2.3.

2.1.4 Universal Mobile Telecommunication System and UTRAN

The Universal Mobile Telecommunication System (UMTS) is the GSM community's answer to the International Telecommunication Union–Telecommunication Sector's (ITU-T) demand for a third generation of mobile communications. At the beginning of the 1990s, the prototypes of such a third generation in ITU-T were originally called Future Public Land Mobile Telecommunication Systems (FPLMTS), and were then synthesized under the name of IMT-2000 (International Mobile Telecommunications beyond the year 2000). Colloquially, 3G is often used for third generation.

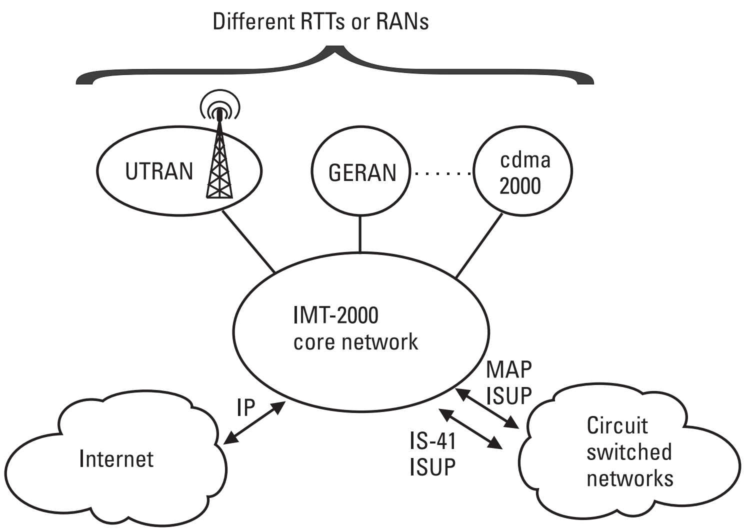


Figure 2.3 A fully developed 3G network with various radio access networks/radio transmission technologies (RTTs).

UMTS's radio access network is, as already mentioned, the UTRAN, which is defined in two versions:

1. The UTRA-FDD (frequency division duplex) mode uses wideband code division multiple access (W-CDMA with 5-MHz carrier) as multiple access method. Transmission rates of up to 384 Kbps are possible with UTRA-FDD.
2. The UTRA-TDD (time division duplex) mode uses a combination of W-CDMA and TDMA and achieves transmission speeds of up to 2 Mbps in semistationary mode.

UTRAN is the preliminary development goal in the world of GSM. Many network operators worldwide already have an appropriate license and have declared their support for UMTS, and many more will follow in the coming years.

UMTS does not only consist of higher speeds on the air interface. The conception of UMTS will, in particular, enable real multimedia applications and will support packet switched and circuit switched services both simultaneously and flexibly. This leads us directly to the next section, which aims to illustrate the differences between packet switched and circuit switched.

2.1.5 GPRS as a Forerunner of 3G and UMTS

As is generally known, GPRS does not target the transmission of speech services or aim to optimize their transmission. Although EDGE does go in this direction, GPRS mainly targets the following:

- *Applications that profit from the always-on state of a subscriber.* Always-on is the constant connection of the user to the Internet and allows, for example, instant transmission of an e-mail to the user terminal without additionally having to dial in. This is, assuming a flat rate, the principle difference from any form of circuit switched data.
- *Applications that can run in the background.* Examples of this are the downloading of e-mail and files. This is also called background traffic.
- *Applications that are based on push services in connection with location-based services.* For example, when passing a restaurant, you are informed that there is a voucher with today's menu. The big questions here, however, are who is to pay for the transmission of such

a push service and to what extent are customers prepared to accept such services?

- *Any form of interactive Internet application that is also suitable for mobile use.* Note that the above-mentioned points also belong to the wide range of interactive Internet applications.

Note that we are not stressing the higher transmission speeds of GPRS. In the heavily used GSM networks and with the limited speeds of the GSM mobile devices, the user will notice very little of this at first. We will discuss the actual data throughput in detail at a later stage.

Many of the services that are interesting for GPRS can hardly be used with traditional mobile telephones. The main problems stem from the limited capabilities of a display that is too small and monochrome, and a number keypad without a mouse function. The mobile phone must therefore become a mobile device where speech services play a major part but are by no means the only service provided. A further developed PDA with a GSM/GPRS module and speech recognition would seem the logical consequence. Another alternative would be devices constructed exclusively for mobile (i.e., wireless) Internet access and thus function without speech facilities such as loudspeakers and a microphone. Plug-in laptop modules (PCMCIA cards) are just one example of this category.

As compared to EDGE or its variation, EGPRS, GPRS is still essentially based on GSM characteristics and does not involve any groundbreaking technological innovations. The packet switched means of transmission is the decisive change. Apart from the lower data rate, in comparison with UMTS, GSM and GPRS together can offer the user almost all the services that are envisaged for UMTS.

It should be asked, however, with a view to the order of market launch, to what extent these possibilities for GSM/GPRS/EGPRS, as compared to UMTS, can actually be exploited.

2.2 Definitions: Circuit Switched and Packet Switched

All telecommunication networks are either circuit switched or packet switched. For the transmission of speech (i.e., "normal" telephone calls), circuit switched networks are mainly used, whereas packet switched networks are mainly used for data transmission. What is interesting here is that the proportion of data connections has increased dramatically in recent years. Indeed, the volume of data connections has actually overtaken the somewhat stagnant volume of speech connections. The lion's share of this development is accounted for by the increasing use of the Internet and the increasing networking of international companies.

The important thing here was to link these two networks up and carry out speech and data transmission via a single, mutual network. Since the proportion of data transmission has already overtaken that of speech transmission, speech should, in the future, also be carried over the worldwide data network. Historically speaking, this development is a 180-degree turnaround since data connections were also transmitted via speech networks until special data networks were introduced. The private Internet user can still follow this principle every evening when he logs on to the Internet.

The problem with network integration is that channel and packet switching have completely different characteristics. It can also be said that some services, such as language, are better circuit switched, while packet switching is better for other services, such as the Internet. What is interesting here is that it is not only speech that demands circuit switching. The same also applies to the transmission of video information. The context is thus obviously more complicated that it initially appears. This section will therefore first present the essential differences between circuit switched and packet switched and then attempt to classify the services.

2.2.1 Circuit Switching

In order to understand circuit switching, look at Figure 2.4. To establish a circuit switched connection, user A states that he would like to be connected to user B. In telephoning, this happens by dialing user B's number. It is then the task of signaling to connect user A to user B. Most of today's signaling methods for circuit switching are, interestingly and by nature, packet switched. It is a fact that confuses many of those involved. This applies, for example, to SS7, as well as to the link access protocol for the D-channel (LAPD). In the signaling process, as shown in Figure 2.4, fixed connections are established between the individual network nodes, which are intertwined with each other to varying degrees. This link-up always involves a certain delay time that can also vary in length. These so-called fixed connections consist in the reservation and allocation of a certain proportion of the transmission channels available between the network elements. These are usually 64 or 56 Kbps for a telephone call. Users A and B can then use this direct connection for the transmission of information. The task of the very important network nodes is then to transfer the necessary resources at the beginning of a connection. During the connection, the network nodes

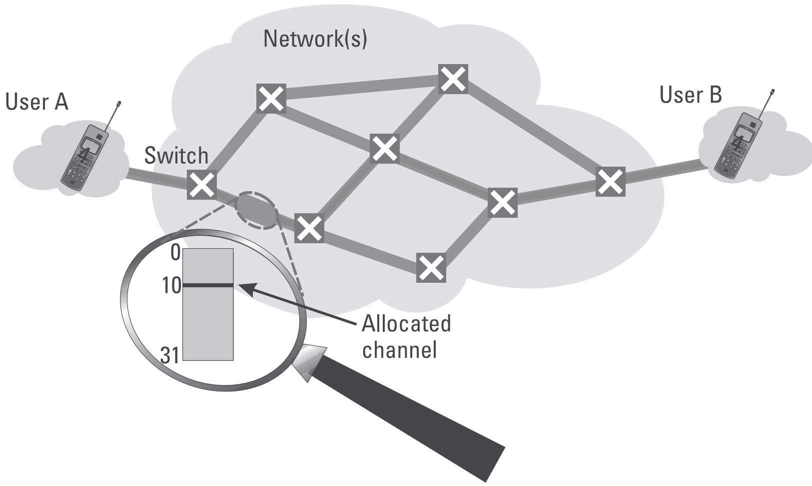


Figure 2.4 The principle of circuit switching.

do not have to execute any switching functions. They merely have to release the occupied resources at the end of the connection. During the connection, there is a tunnel between users A and B, which can only be used for the transmission of their information. In any case, this simple form of transmission was the only way of setting up a telecommunication network 100 years ago. After all, connections have not always been established using digital networks [digital switching network (DSN)] but with mechanical magnetic switches. Packet transmitting technology, which expects considerably more switching functions from the network nodes, has only recently become even possible.

The problems and characteristics of circuit switching can be presented as follows:

- If user A wants to be connected to user B and there are no resources available either in one of the network nodes or the lines between these network nodes, the connection cannot be made and the caller receives an engaged signal.
- In circuit switching, only small demands are made on the switching capacity of the network nodes. These can be fulfilled satisfactorily with mechanical elements.
- Many applications in telecommunications have greatly varying demands on the speed of the transmission channel during a transaction. In speech transmission, for example, there is usually only one

side talking while the other listens. In other words, at least 50% of the resources made available for a telephone call are not even needed. One could say that these resources are wasted. This also applies to GSM, in spite of DTX.

- In circuit switching, a channel is set up between user A and user B. The format of the information to be transmitted is now largely the affair of users A and B because the network nodes situated in between do not have to interpret the signals and information transmitted.
- The problem of the varying demands made on the speed of the transmission channel can mean the end for some applications. You may consider video conferencing techniques in which only the delta information on the respective previous frame must be transmitted. For a certain time, the switched channel of 64 Kbps, as an example, is sufficient because no one is moving, but several Mbps suddenly become necessary because everyone in the room stands up. This capacity is then not available, which then results in ghosting and distorted pictures. This can happen even though all the other channels may be free on neighboring channels on the various transmission channels between the two partners.
- In circuit switching, at least the switched resources are continuously available. Even after a longer rest phase, the reserved resources can be used in full immediately without having to be requested again with an inevitable time delay as a consequence. In other words, once a connection has been made in a circuit switched network, even real-time applications are possible.
- The problem of constantly changing demands on the speed of the transmission channel occurs during Internet access and annoys the subscriber because he has to pay according to time. While the information on the first Web site is being analyzed, the resources are at rest. Nonetheless, the charges are still being levied. You then proceed to the next page and so, for a short time, considerably more resources are required than the fixed modem link, with a maximum of 56 Kbps, can handle.
- In speech transmission in particular, considerable progress has been made in recent years in the compression of speech information. GSM, for example, can function with only 6.5 Kbps at half-rate transmission with acceptable quality and with adaptive multirate codings—even 4.75 Kbps is possible. Compare these values with

the fixed network rate of 64 Kbps. Here, circuit switching does not have the flexibility, for example, to offer the user a greater or smaller bandwidth dependent on price.

2.2.2 Packet Switching

Packet switching basically has the same task as circuit switching. Two subscribers wish to exchange information, and the network elements situated in between must carry out the appropriate switching functions. The great difference to circuit switching is that there is never a fixed connection set up between computers A and B in packet switching. Instead, the information to be transmitted must be segmented and embedded in several packet frames. Each individual packet contains all the address information (header), so that the network nodes between computers A and B can forward this packet correctly and thus deliver it to the target address.

As shown in Figure 2.5, this method of forwarding information leads to each packet being able to take a different route within the network. The network nodes decide, for example, on each respective step depending on the burden on the transmission channels. The order in which the packets reach the receiver is thus uncertain.

The demands made on the network nodes are then extraordinarily high as compared to circuit switching. A switching procedure must be carried out for each individual packet. It could be said that each individual packet is a short circuit switched activity in which an address must be evaluated, a resource allocated and then released again. The packet must be forwarded during this activity.

Furthermore, computer A or B must have free network access whenever a packet is to be sent; otherwise, potentially unpleasant delays can occur,

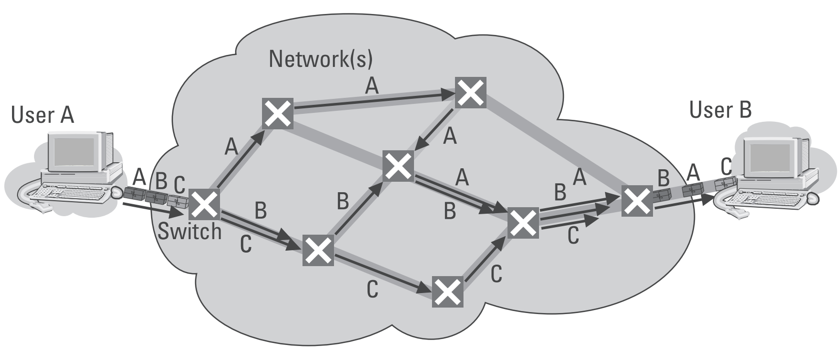


Figure 2.5 The principle of packet switching.

depending on what type of information is being forwarded. Take speech as an example. When subscriber A says his first sentence, this information should not arrive at subscriber B after the second sentence. In other cases, such as e-mail transfer, delays are secondary provided that there is a suitable protocol at the receiver's end that is able to reproduce the correct order of the individual segments.

The problems and characteristics of packet switching can be presented as follows:

- There is a very high demand on the switching performance of the individual network nodes. It is therefore difficult, or even impossible, to use channel switching network nodes for packet switching.
- In packet switching, resources are not permanently allocated, but only on demand. Theoretically, depending on the requirements of an application, all resources or, during pauses, no resources can be occupied. Packet switching is much more flexible than circuit switching as regards resource allocation.
- In packet switching, it is very difficult to guarantee constant delay times or data rates. Conditions in the network are constantly changing, which means that large variations can occur for both parameters. For applications with a conversational character (speech), packet switching thus leads to serious disadvantages, which can only be avoided with the help of special resource reservation protocols.
- In most telecommunication applications, there are inactive periods during a transaction. Consider our previous example of a normal telephone call where at least 50% of the resources are inactive at any given time. The transmission channels in a packet switching network can be thus more modestly sized.
- In packet switching, an overhead will inevitably arise because each packet has to carry all the routing information with it.

2.2.3 Summary

In the previous sections, we outlined some differences between packet switching and circuit switching. Packet switching has many advantages that make it the preferred transmission method especially in many kinds of data transmission. However, circuit switching also has considerable advantages. This especially applies to interactive services such as speech transmission and all applications in which delays cannot be tolerated.

If the immediacy of a transmission is also to be guaranteed for packet switching, appropriate protocols must constantly have the necessary resources reserved. This is the case even during possible phases of inactivity, which would then more or less eliminate the difference from circuit switching. You must then consider what kind of services you wish to transport via the network before calling.

The vast majority of existing mobile communication standards, including GSM, have primarily targeted speech transmission and have thus concentrated on circuit switching.

This situation changes with GPRS because GPRS is *really* packet switched. We stress the "really" here because many GSM specialists are surprised by the actual packet switching characteristics of GPRS, especially on the air interface.

What is also important here is that GSM and GPRS can coexist in a single network, even on a single ARFCN. Accordingly, after the introduction of GPRS, both circuit switched and packet switched services can be offered in GSM. The aim of this is to deal with the differing switching requirements of the various applications.

2.3 Problems of Packet Switching in Mobile Communication

Before we take a closer look at GPRS, some general issues should be broached concerning packet switching in mobile communication. We shall, of course, always place these individual points in the context of GPRS.

2.3.1 A High Number of Accesses of Short Duration

Packet switching differs from circuit switching in that resources are only occupied when they are actually needed. This definition also includes the inverse conclusion that resources must also be released as soon as they are no longer required.

By way of example, Figure 2.6 shows a Telnet session in which a user types in the string "Miriam." While in circuit switching, a suitable resource is reserved for the whole period; in packet switching a resource must be allocated from inactivity status every time a key is pressed.

Of course, this resource must then also be released as soon as a single character has been transmitted. If the user types quickly enough (i.e., as long as further characters are occupying the mobile station's output queue), it

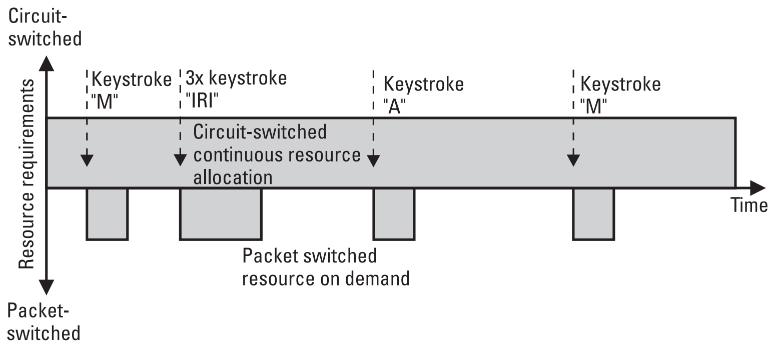


Figure 2.6 Example of a Telnet session, once circuit switched and once packet switched.

can certainly happen that several characters are transmitted one after the other during an occupation. In the example illustrated in Figure 2.6, this applies to the "iri" chain of characters. The actual duration can be less than 1 second per occupation.

The following questions or problems arise directly from this:

- Can the mobile user be allocated resources in the uplink and/or downlink directions from inactivity and almost without delay?
- If so, can these resources be released quickly enough and under control?
- Can collisions on the channel arising from different mobile stations requesting resource allocation be avoided?

All the points mentioned are important and must now be addressed. We first want to examine the third question because this problem has considerable influence on the capabilities of GPRS.

2.3.1.1 The Slotted Aloha Access Method

In 1969, a team at the University of Hawaii, led by Professor Hitachi Nosi, installed a wireless data transmission network for connecting the base station in Honolulu with the outposts on the various islands. As Figure 2.7 shows, the system had a range of approximately 200 miles. This network was given the lovely name of Aloha.

The Aloha system was intended to support bidirectional transmissions. There were two channels available for this, one for the uplink and one for the downlink direction. For this purpose, each of the outposts received its own identity number. When transmitting to the outposts (downlink), the

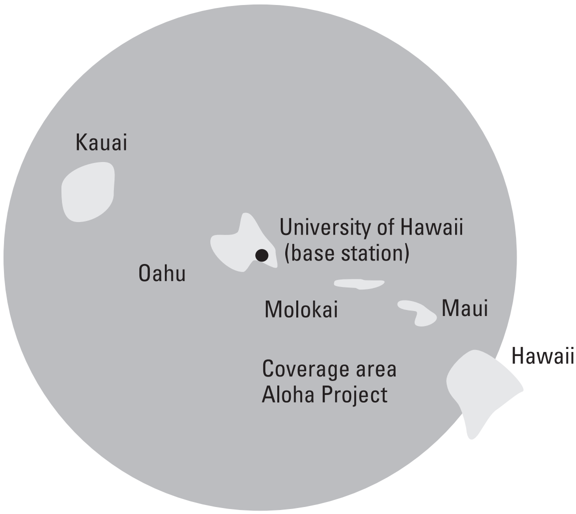


Figure 2.7 The range of the Aloha network.

identification was placed at the front of the data packet, and the outposts, which were constantly on air, would only forward or process those packets that were addressed to themselves.

In the opposite direction (uplink), the outposts would give every packet to be sent to the base station their own identification in order to enable the base station to differentiate. After transmission, the sender would wait for confirmation. If this did not arrive within a certain time, the data packet would be transmitted again.

In this way, all the outposts used the same channel and the same resource in the uplink direction.

The Aloha network performs well as long as the uplink burden does not become too heavy. More precisely, the Aloha system works at maximum effectiveness at an uplink channel load of 18%. If the uplink channel is burdened more heavily, there occurs an increasing number of partial or total collisions of data packets traveling uplink, as shown in Figure 2.8. Colliding data packets produce nothing more than an increased noise level at the receiving base station, and can thus neither be processed nor confirmed. Accordingly, each outpost will resend an unconfirmed data packet, which will then increase the probability of collisions even further.

Professor Nosi's team was not satisfied with this result and considered what measures would increase the effectiveness of Aloha.

Collisions of this kind can be prevented if the outposts are synchronized. If an outpost knew that another was transmitting, it would not be allowed

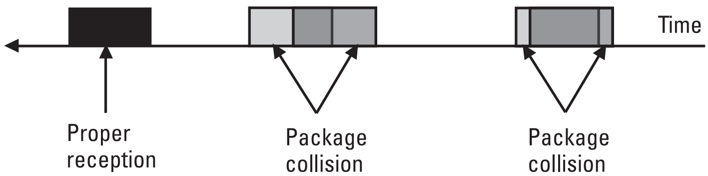


Figure 2.8 Collisions of uplink data packets in the Aloha system.

to transmit itself. This idea was rejected because synchronizing the outposts was financially impracticable.

The main problem was identified as being the high probability of partial uplink data packet collisions. Even a partial collision of two data packets renders perfect reception at the base station impossible. There is, in fact, a cheaper way of avoiding partial collisions than synchronizing the outposts. This way is called slotted Aloha and is used today by all mobile communication standards, including GSM, for network access.

For the slotted Aloha method, the base station must carry out the transmission in the downlink direction in time slots of a strict, predetermined period. The outposts must be configured so that they synchronize themselves to this period and may only transmit uplink when they perceive the start of a downlink time slot. One may object here that even in slotted Aloha, the outposts do not know whether another outpost is transmitting. What can be avoided with slotted Aloha, however, is all partial collisions of uplink data packets. Here there is either a total collision or a data packet is received perfectly. This improvement is shown in Figure 2.9. The time slots missing in Figure 2.8 can be seen clearly.

As already stated, even slotted Aloha does not inform the outposts about any other outpost's intentions to transmit. Collisions can therefore not be avoided here either. However, the maximum effectiveness of the uplink channel with slotted Aloha increases to 36%, double that of normal Aloha.

2.3.1.2 Consequences of Slotted Aloha for GSM and GPRS—The Uplink

The slotted Aloha method has been described in detail, not only because it is used by all major mobile communication systems as a method of access,

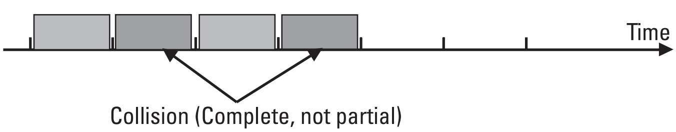


Figure 2.9 Avoidance of partial collisions by using slotted Aloha.

but also because the maximum efficiency of 36% in the uplink channel is of some importance for GSM and even more so for GPRS.

It must be stressed, however, that the 36% efficiency limit in GSM and GPRS is only valid for the RACH on which the mobile station sends its access burst to the base station in order to receive resource allocation.

For circuit switched processes such as in GSM, this limitation is not usually a major problem. As shown in Figure 2.6, every resource is allocated for a relatively long duration in circuit switched GSM. This duration can certainly run into minutes.

For GPRS, however, with occupation phases that can be as short as a fraction of a second, a new access burst must also be sent for each new occupation. The burden on the RACH increases very quickly and the 36% mark is soon reached. The probability of collisions, and thus delayed access, also increases accordingly. This problem can only be solved by increasing the number of RACHs. We will see in the following that for this reason a special packet random access channel (PRACH) has been defined for GPRS that can be configured in a cell as an option and according to burden.

2.3.1.3 Burden Problems in the Downlink Common Control Channel

So far, we have only considered the burden on the uplink common control channel, or, to be more precise, the RACH. Although the burden on the corresponding downlink common control channel, the paging channel (PCH), has nothing to do with slotted Aloha, for the sake of completeness we should point out that the PCH has to cope with burden problems even in GSM. These problems could be alleviated for GSM if several mobile stations (up to four stations to be more precise) could be addressed at the same time per paging message.

The situation with resource allocation in GPRS is similar in downlink to that in uplink, at least as far as the duration of occupation periods and the desired immediacy of mobile station access are concerned. Here too the burden increases considerably and this problem can only be solved, as with the uplink direction, by making more resources available on demand. For these reasons, GPRS introduces the optional packet paging channel (PPCH), which we shall examine in more detail later.

2.3.1.4 Immediate Occupation and Release of Resources

We stressed in previous sections that both sides, the mobile station and the network, must be able to request and allocate resources immediately or within moments. Otherwise, applications that are very sensitive to delays can be excluded from the start. Speech transmission is an example of such an application.

This requirement also applies equally to the release of resources; otherwise, the next user would not be able to gain access quickly enough.

The conclusion to be drawn here is that in a mobile packet switched network, complicated signaling procedures for the authentication of a subscriber or for the allocation of Internet addresses or for the activation of encryption must either be eliminated completely or must not have to be carried out for each new occupation. The latter way has been chosen for GPRS, as will be discussed later in the book. A subscriber carries out all necessary registration procedures at the beginning, so that he can then transmit or receive data according to the situation at any time (Figure 2.10). What is surprising to many is that these registration procedures from the fields of GPRS mobility management (GMM) and session management (SM) constitute a payload transfer, from the point of view of the GPRS air interface. When the mobile station is switched off, it logs off the network.

At this point we would like to summarize as follows:

- At the beginning of this section, the demand for and allocation of resources were discussed separately. Note that in GPRS, the mobile station can only request allocation of uplink resources, while the network monitors this allocation. On the other hand, the mobile station cannot request the allocation of downlink resources since the network always has to request this. Furthermore, the network also always monitors the allocation of downlink resources. In any case, complicated signaling processes for such a form of resource allocation must be excluded for GSM.
- A further problem arises from the principle of discontinuous reception (DRX) in the allocation of downlink resources. This is always the case when the network wants to transfer packet data to the mobile station.

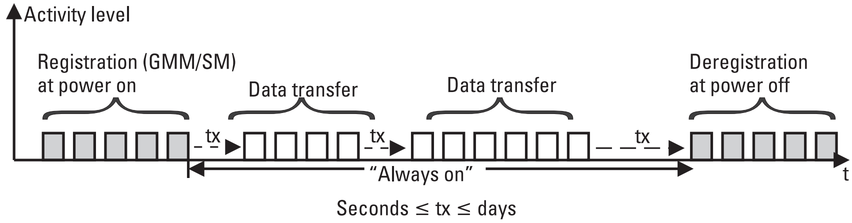


Figure 2.10 After registration, GPRS enables the subscriber to have always-on status.

GPRS also optionally uses the possibility of reducing the power consumption of the mobile station by means of DRX. Accordingly, the waiting time for the paging group of a single subscriber alone can amount to a second or even longer. It is not possible to be more precise at this point because these waiting times are dependent on configuration. The GPRS standard itself states a target delay time of 0.5 to 1.5 seconds between an application and the output of the GPRS network. These values include, in particular, the unavoidable network access delays. Figure 2.11 illustrates these delay times, which lie between the requirements of an application and the actual transfer phase. Here, the long delay in a downlink transmission, caused by DRX, becomes especially clear.

2.3.2 Packet Data Overhead

One additional problem that especially applies to mobile communication arises from the fact that packet switching demands that each individual data packet carries its own header with all the necessary routing information along with it. One example of this is the typical Transaction Control Protocol/ Internet Protocol (TCP/IP) packet, in which the header is usually 40 bytes long, with TCP and IP being of 20 bytes each.

The actual net data share can vary greatly. The Point-to-Point Protocol (PPP), as used in dial-up connections, has a maximum frame length of 1,500 bytes. Depending on the applications, shorter frames are by no means unusual, such as the above-mentioned Telnet protocol, supported by TCP/ IP. Here, the net data share per IP frame is only a few bytes. While the unavoidable overhead is possibly annoying for the home Internet surfer, it can present a real problem for the mobile Internet user, because in GPRS every byte transmitted has to be paid for, as does every second in GMS.

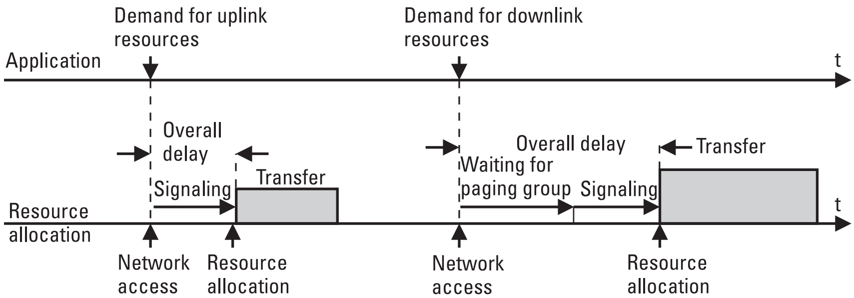


Figure 2.11 Delay times in resource allocation in uplink and downlink directions.

The compression of data to be transmitted via GPRS is thus a useful performance characteristic, and indeed, GPRS offers two different compression methods: V.42bis for the compression of all kinds of data and RFC 1144 exclusively for the compression of TCP/IP headers. We will discuss the compression methods in more detail in Chapter 5.

2.3.3 Permanent Control of the Distance to the Mobile Station (TA Control)

An additional problem for packet switched transmission in mobile TDMA systems is the timing advance control (i.e., the permanent checking and adjustment of the TA value in the mobile station). We have already briefly referred to this problem in Section 1.4.2 but would now like to examine it in more detail.

First of all, it must be made clear that packet switching is based on resources only being allocated when they are actually required. This has already been discussed in detail.

In a packet switching system, forward and backward directions are fully independent of one another. For a mobile packet switched system such as GPRS, this means independence from uplink and downlink directions. So if there is only a downlink transmission, no resources are allocated to the uplink direction (i.e., nothing is transmitted in the uplink direction). Accordingly, GPRS cannot simply take over the TA control function of GSM during a live connection.

One may object that the TA control is simply unnecessary if nothing has to be transmitted uplink. Unfortunately, this is not quite correct because even with purely downlink transmission, some kind of confirmation is always required.

So there are two possibilities: either to have these confirmations sent as access bursts with TA = 0, or to find additional alternatives to TA control for GPRS. In Chapter 3 we shall learn that GPRS takes both these paths.

2.3.4 The Handover Problem

In circuit switched GSM, there is a clear differentiation between a cell change in what is known as *idle* mode, when no RR resources are allocated on the air interface, and a *handover* during an active connection with allocated RR resources.

- In the idle mode, the mobile station will carry out a cell reselection procedure, which, in the event of a change in location area, can also lead to a location update scenario [1, 2]. Cell reselection is always monitored by the mobile station in GSM, apart from operation and maintenance (O&M) interventions. The mobile station decides independently for a possible cell reselection, based on the results of the level measurements for its own cell (serving cell) and the various neighboring cells.

- In the RR active mode (i.e., during an active connection), the mobile station also carries out level measurements for its own cell and the neighboring cells. The difference is, however, that the mobile station sends these results to the BTS, which then forwards the mobile station's measurements (downlink measurements) and its own measurement results (uplink measurements) on to the BSC. Now it is the BSC that has to decide on a possible cell change in the mobile station on the basis of these complete measuring results. In other words, during an active connection the mobile station is in slave mode to the BSC.

These two very different methods work perfectly in GSM. With GPRS, however, the following difficulties arise:

1. In Section 1.1.1.2, we referred to the fact that the GSM-BSC cannot be used for the RR functions of GPRS. Accordingly, the BSC cannot be entrusted with the handover control for GPRS.
2. As Figure 2.12 shows, it is difficult to differentiate between active and inactive phases in GPRS. If active phases only last for fractions of seconds with fractions of seconds between these active phases (as in our Telnet example in Section 2.3.1), when should you talk of handover and when should you talk of cell reselection? In GPRS, such a distinction clearly makes no sense.

In view of the problems addressed, handover procedures, or a differentiation between active and inactive cell changes, are dispensed with completely

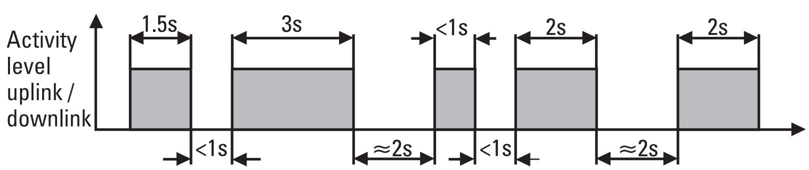


Figure 2.12 Differentiating between handover and cell reselection.

in GPRS. Instead of this, a cell update scenario is defined, which can be monitored either by the mobile station or by the network. In Chapter 4, we will examine the cell update scenario in detail.

2.3.5 Summary

The following sections aim to explain the particular problems of mobile packet switching. These problems are especially serious when it is not a completely new network that is to be developed, but when an existing system such as GSM is due to have a packet switched expansion.

The necessary GPRS expansions should not have a negative influence on the existing GSM services but should enable new functions. In Chapter 1, it became clear that many processes in GPRS, especially the choice of and synchronization to a cell, should proceed exactly as in GSM. In the following, we will present the network architecture of a combined GSM/GPRS network. Note that pure GPRS networks, although perfectly possible, are not very sensible in commercial terms.

2.4 The Combined GSM/GPRS Network Architecture

When a GSM network is set up, between 70% and 80% of the costs are for the BSS. These costs include the hardware and software for the equipment itself and also the expenditure for lines, network planning, locations and location acquisition, often a great problem, and antennas. A relatively modest 20% to 30% remains for the NSS with its digital exchanges (the MSC) and the various databases (VLRs and HLRs).

For this reason, every effort was made to limit the additions to the BSS necessary for GMS as much as possible. One problem here is the BSC, which is basically a small circuit switched exchange.

An additional problem for the conception of a GSM/GPRS network architecture is that GPRS can only be seen as the first step towards UMTS or EDGE. In other words, network operators will have little interest in investing millions of dollars today in a network architecture that cannot be used after any developments in UMTS or EDGE.

Furthermore, efforts were made in the conception of the network architecture and the GPRS protocol stack to ensure independence of the higher layers or the core network from the air interface, the protocols used there, and the BSS in general. This wish is especially based on wanting to connect different RANs to the same core network and also wanting to be able to use, in the process of the UWCC, the GPRS protocol stack and its network architecture for the North American TDMA standard, IS-136.

Based on these demands, the BSS in GPRS is used almost without any additional hardware, while a completely new, packet switched core network is required. This is illustrated in Figure 2.13, which also shows that with the introduction of GPRS, the divided core network uses the same BSS and the same air interface.

In the following sections we approach the necessary additions gradually, starting with the BSS.

2.4.1 The Packet Control Unit

For financial reasons, the BSS should be affected as little as possible by the GPRS expansion. However, on account of the limited capabilities of the BSC, hardware expansion is unavoidable. This hardware expansion is the packet control unit (PCU), which can be installed as part of the BSS at different locations in the network (Figure 2.14).

As a rule, in the basic version the PCU is simply one or a few expansion card(s), including the additional software. Most manufacturers have opted for the medium location version. The PCU thus becomes more or less an integral part of the BSC.

One must consider, however, that the choice of location for the PCU is not completely arbitrary, but is also indirectly influenced by the position of the TRAU. In order to gain a better understanding of this, the tasks of the PCU must be introduced first.

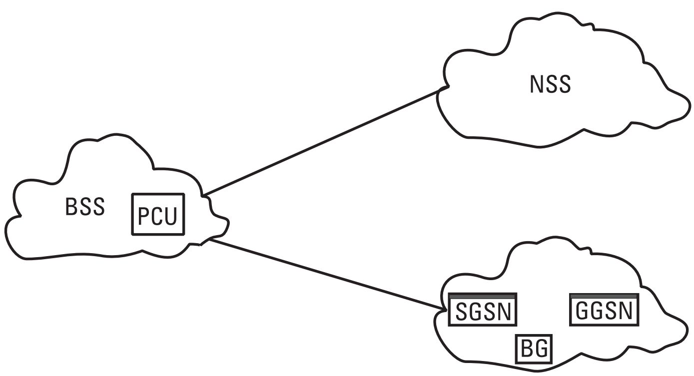


Figure 2.13 The BSS is used equally by the NSS and the GPRS core network.

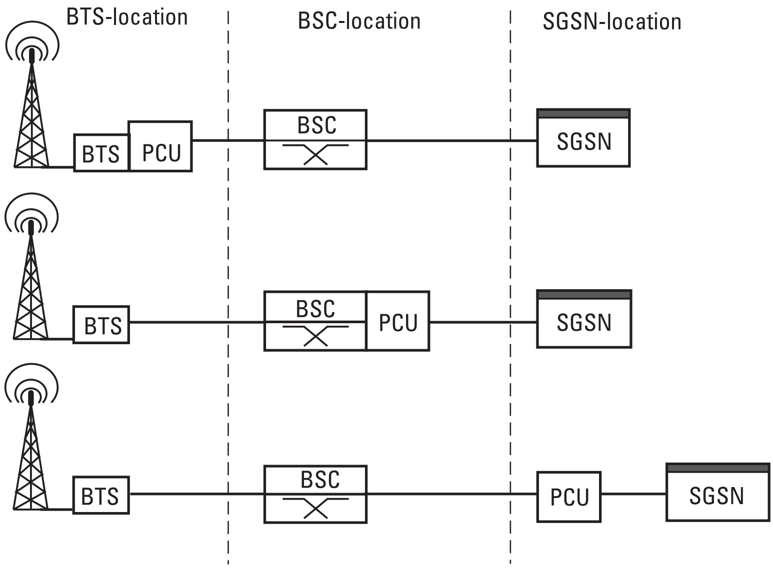


Figure 2.14 Possible locations for the PCU.

2.4.1.1 Functions of the PCU

The PCU primarily takes on the following tasks in a GSM/GPRS network:

- It has the responsibility for the radio resource management functions in GPRS. This is an entirely new protocol as compared to RR in GSM, and it is called the radio link control/medium access control (RLC/MAC) protocol in GPRS. While the BSC thus takes care of administrating the circuit switched radio resources in GSM, the PCU takes over this task for GPRS. It can also be said that the PCU is responsible for the lower layers of the GPRS protocol stack on the air interface.
- The second function of the PCU is at least as important as the first. It is the conversion of packet data into what are called PCU frames. As Figure 2.15 shows, the packet data coming from the SGSN is converted to PCU frames within the PCU. These PCU frames are forwarded transparently through the BSC to the BTS, which then takes on the additional processing in the form of channel coding, interleaving, etc. Transparently for the BSC, RLC/MAC signaling information also reaches the BTS, and finally the mobile station, in this way. The special thing about the PCU frames is the fact

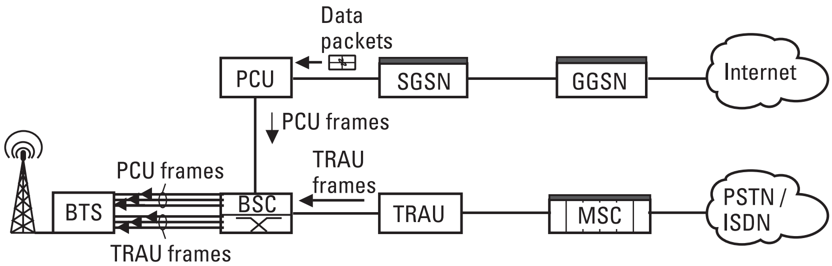


Figure 2.15 Conversion of packet data into PCU frames by the PCU.

that they are the same format as the TRAU frames that are sent transparently through the BSC to the BTS by the TRAU. In this way, the packet switched GPRS network can be connected to the existing GSM network almost unnoticed. An animated version of Figure 2.15 can be downloaded from www.inacom.com.

In this context, the second function of the PCU, in particular, has several consequences:

1. *Data rates above 16 Kbps:* TRAU frames have (as shown in Chapter 1) a length of 320 bits and are sent from the transcoder to the BSC every 20 ms. In other words, 16-Kbps channels are used between the TRAU and the BTS. Here, we are using the example of the standard remote TRAU configuration, in which the TRAU is installed at the MSC location. In this configuration it is then impossible to transmit more than 16 Kbps per time slot.

   However, this is exactly what should happen in GPRS and the higher channel coding methods. This is a major problem that not only affects the channel configuration but also the switching performance of the BSC because the GSM-BSC is equipped for the switching of 8-Kbps and 16-Kbps channels. Achieving greater bandwidths with GPRS or EGPRS thus also requires massive changes in the BSC.
2. *Coordination between BSC and PCU:* The PCU converts packet data into PCU frames that are sent transparently through the BSC and to the BTS for further processing. BSC and PCU must thus share the time slots and resources on the Abis and air interfaces. BSC and PCU must also be coordinated in order to prevent BSC and PCU trying to allocate a time slot that is already occupied by

the other. The network operator has to determine from the operation and maintenance center (OMC) how many resources GPRS should be allowed to use as a maximum.

3. *Manufacturer dependence:* The format of TRAU frames is specific to the manufacturer. The conversion of packet data into something that is the same as the TRAU frames in format terms is therefore also a manufacturer-specific function. Accordingly, the PCU *must* come from the same manufacturer as the BSS.

2.4.1.2 Possible Positions of the PCU

In the previous section it was mentioned that the PCU can theoretically be integrated either into the BTS or the BSC or installed together with the SGSN (see Figure 2.14). All versions have their specific advantages and disadvantages.

By installing the PCU at the location of the BTS, the manufacturer avoids the problem of limited bandwidth, providing that the BSC has already previously switched 64-Kbps channels. Expensive BSC expansion can thus be dispensed with in this case. This advantage, which is not to be underestimated, is, however, counterbalanced by some disadvantages. First, many PCUs are required (i.e., exactly as many as there are BTSs). Second, all cell changes (handovers) must be controlled by the SGSN in this case, which means considerably heavier burdens for this network element.

The standard installation chosen by most manufacturers is the integration of the PCU into the BSC. Here, the PCU can be found exactly where it is to be expected in view of the RLC/MAC control function. In this case, the PCU also becomes an actual physical part of the BSC, or the BSS. For GPRS and lower transmission rates, this is the best solution. With higher transmission rates, however, one is once again confronted with the problem that the packet data must, as shown in Figure 2.15, pass through the BSC, which is designed for the switching of slower 16-Kbps or 8-Kbps channels. As already mentioned, the respective manufacturer has to solve this problem individually.

The third possibility is to install the PCU at the location of the SGSN, which is usually the same as the location of the MSC. Here, many PCUs are concentrated at the location of the SGSN as is common with the remote TRAU and the MSC. This elegant alternative has many advantages, such as the concentration of additional equipment for the PCU (e.g., electricity supply) and a common link between MSC and BSC and between SGSN and PCU. It also has serious disadvantages. For instance, in order to enable dimensioning and transmission via semi-public networks, the manufacturer must describe and reveal in detail the protocols between PCU and SGSN, which are normally secret. Furthermore, the communication between BSC and PCU must also be conducted via this long line, which can lead to an increased risk of synchronization problems during operation. These are both disadvantages that can affect the manufacturer as well as the network operator.

2.4.2 The Serving GPRS Support Node

The SGSN is a component of the GPRS core network. Via an SGSN, many different PCUs or BSSs are supplied with the GPRS. Each SGSN can therefore be allocated one service area of BSSs, as shown in Figure 2.16.

To put it simply, the SGSN for GPRS takes over the functions that are performed by the MSC and the VLR in circuit switched GSM. In GPRS, the physical border between MSC and VLR, which is already blurred in GSM, does not exist at all any more.

In addition to these functions, the SGSN has additional tasks that are specific to GPRS, which are either performed by other network elements or do not exist in GSM.

2.4.2.1 Packet Switching

For SGSN, the switching and forwarding of data packets is more important than any other function. Consequently, manufacturers expand the usually already existing packet switches with the GPRS-specific additional characteristics and then sell these network elements as SGSN. At this point, the comparison of SGSN and MSC presents itself again, because the MSC is an extension of the ISDN exchanges.

The difference between MSC and SGSN, or rather the different times of development, becomes clear when one has seen both network elements.

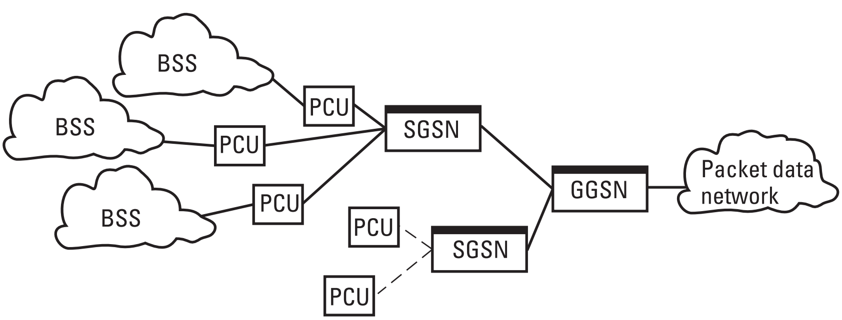


Figure 2.16 The positioning of the SGSN in the network as a whole.

While even a smaller MSC typically fills up a medium-sized room and leaves an accordingly large footprint, a small SGSN can be realized with a few 19-inch plug-in elements.

2.4.2.2 Transfer of Short Messages

In circuit switched GSM, short messages (SMS) are transmitted either via the SDCCH, without a live connection, or via the SACCH, during a live connection.

GPRS should also be able to transmit short messages, as already shown in Chapter 1. In this case, short messages are sent to and from between the SMS-IW-MSC/SMS-G-MSC and the SGSN. Note, however, that short messages in GPRS must always be seen as a payload and thus require traffic channels (PDTCH).

2.4.2.3 Ciphering in GPRS (Data Encryption)

In GSM, the BTS and the mobile station perform the function of encrypting information. As shown in Chapter 1, information in circuit switching GSM is encrypted and deciphered on the burst level, Layer 1. The encryption sequence changes with each TDMA frame due to the FN.

For GPRS, encryption is performed at the packet data level (see Figure 2.17) between the SGSN and the mobile station in Layer 2. We will go into further detail on data encryption in GPRS in Chapter 5, taking into account the protocol stack.

2.4.2.4 Data Compression in GPRS

On the network side, the SGSN is also responsible for the compression of data. Here, GPRS supports two independent compression methods.

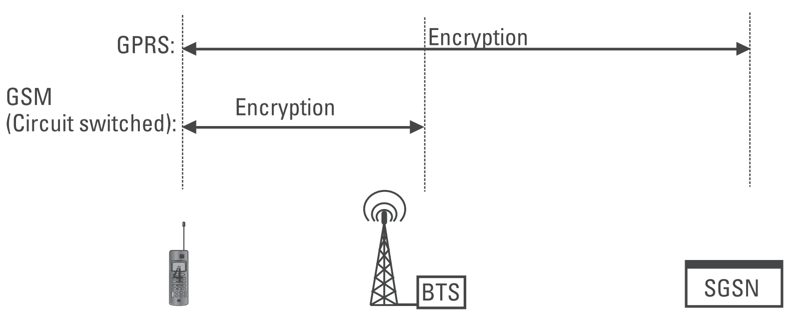


Figure 2.17 The encryption function in GSM and GPRS.

1. *RFC 1144:* RFC 1144 is used exclusively for compressing TCP/IP frame headers. In RFC 1144, use is made of the fact that many parameters in the TCP/IP header are redundant after a virtual TCP connection between the mobile station and the application has been established. These can thus be suppressed, meaning that, as a rule, 40 bytes of TCP/IP header can be compressed to an average of 2 to 3 bytes.
2. *V.42bis:* The ITU-T standard V.42bis, known from the field of modems, can also be used in GPRS. Strings of almost any length can be compressed with the help of V.42bis. Here, so-called decision trees are used, which are constructed from code-words that are agreed upon and saved at both end points. Figure 2.18 shows an example of the function of V.42bis. For optimum V.42bis function,

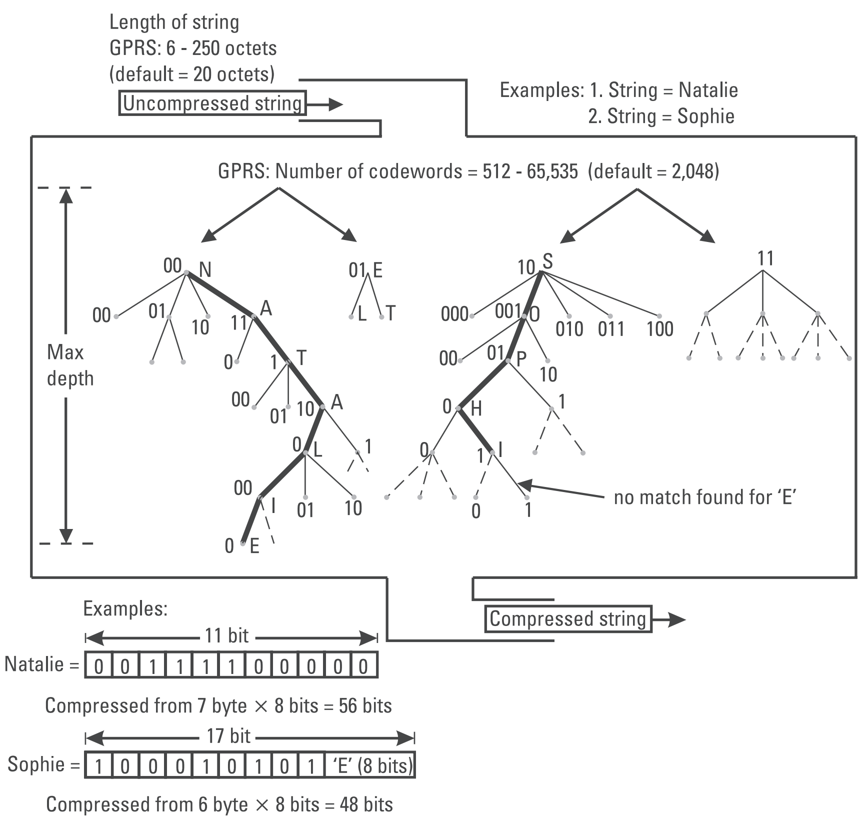


Figure 2.18 Principles of V.42bis compression.

> however, it is essential that both the compressor and the decompressor use the same sets of code-words, or can agree upon them before data is exchanged. The use of V.42bis therefore requires considerable effort.

The advantage of V.42bis is its independence from the raw data to be compressed as compared to RFC 1144. With V.42bis, all nonrandom data strings can basically be compressed (e.g., both TCP/IP headers that have already been compressed with RFC 1144 and Internet files or other data sequences).

Note that in any case, the use and implementation of the RFC 1144 and V.42bis methods are optional for the SGSN and the mobile station. This fact is especially important for mobile station manufacturers because costs can be saved. For instance, the use of compression methods already available in the terminal that is connected to the mobile station (e.g., a laptop) is more sensible than integrating a compression method into every mobile station.

2.4.2.5 The SGSN and Charging

In GPRS, the SGSN is responsible for the collation and, in particular, the forwarding of certain charging information to the charging gateway (CG). However, the SGSN can only collate the charging information regarding the use of its own network or the air interface. This point will be discussed again in the following examination of GGSN. Note in any case that in GPRS, volume and time-based charging must be supported. In view of the packet switched transmission and the many advantages of the always-on status, volume-based charging is to be recommended.

2.4.2.6 SGSN with Cell Change (Handover)

When there is a cell change in GPRS, which also affects the change of SGSN, the SGSN is responsible for ensuring that all data packets not yet confirmed by the mobile station are sent to the new SGSN for retransmission. SGSNs must be connected to each other accordingly in order to enable network-wide roaming. Note, however, that roaming in external networks will also lead to a termination of the connection in GPRS.

2.4.3 The Gateway GPRS Support Node

As the name already suggests, the gateway GPRS support node (GGSN) is essentially the interface between the GPRS network and external packet data networks, especially the Internet. The GSM standard is relatively vague in its definition of functions for the GGSN, and even allows the integration of the SGSN and the GGSN into the same network element. In order to understand GPRS, it is necessary to recognize that the GGSN externally (i.e., from the Internet) seems like a normal router. It is also important that GGSN is essentially a packet switch that has to be modified for additional functions in the GSM/GPRS network. To operate GPRS, a network operator needs at least one GGSN (i.e., a link to external packet networks). It is only the increasing burden, but not geographical necessity, that leads to the connection of further GGSNs in later operation. This independence from geographical limitations distinguishes the GGSN from the SGSN, as illustrated in Figure 2.19. Figure 2.19 also shows how GPRS support nodes (GSNs) communicate with each other. Each GSN is identified via IP addresses. Different IP addresses are used for data and signaling.

In its interface function, a GGSN is responsible for the following functions in particular.

2.4.3.1 Setting up of PDP Contexts

External packet protocols are described as PDPs in GPRS. The most important of these PDPs is undoubtedly the Internet Protocol. Further PDPs supported by GPRS are, with GPRS release 97 and 98, the X.25 protocol and, with release 99, the PPP and the Internet Hosted Octet Stream Service (IHOSS), which sets up a transparent bit pump between the mobile station and the GGSN. The interesting point here is that the support of the mostly European X.25 protocol was terminated with release 99.

Before a mobile station in GPRS can send or receive packet data, activation of a PDP context between the mobile station and the network is required. When the PDP context is activated, network and mobile station agree, in particular, on the PDP address (IP address) and the transmission

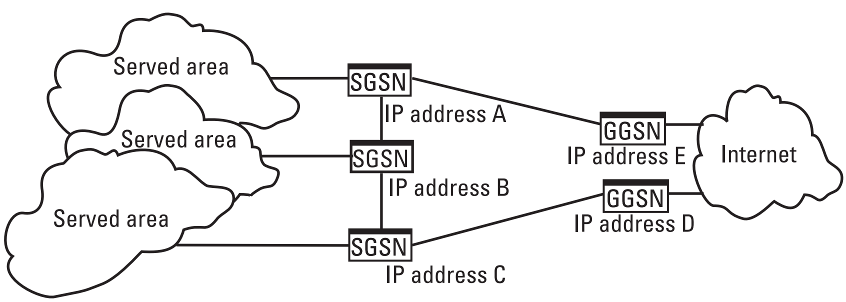


Figure 2.19 Regional dependence of the SGSN and independence of the GGSN.

parameters to be used. These are, for example, the maximum transmission rate, delay times in the transmission of data packets, and resource allocation priority. All these parameters belong to the quality of service (QoS) profile, which will be discussed in further detail in Chapter 4.

Although the SGSN plays an important part in activating PDP contexts, the main burden and control is ultimately a matter for the GGSN.

2.4.3.2 Anchor Function

As Figure 2.20 illustrates, the GGSN is the unchangeable element in GPRS during an active PDP context. This fact is clearly necessary since the data packets coming from the external packet data networks require a fixed point of entry into the GSM/GPRS network. Only the SGSN is changed during the roaming in the network when the respective served area changes. At this point, many of the participants in our seminars ask whether it is possible to change to another PLMN during an active PDP context (e.g., when traveling from Germany to the Netherlands). This question alludes to the frequent breakdown of connections in such situations.

The answer is that it is at least theoretically possible to change PLMN during an active PDP context. In practice, however, this will be the exception even in the longer term because an immensely high degree of coordination would be necessary between the operators in these countries.

2.4.3.3 The GGSN and Charging

In Section 2.4.2.5, the principles of charging via the SGSN were presented. Note, however, that here the GGSN also plays an active part in charging. While the SGSN concentrates on collating all the charges regarding the use

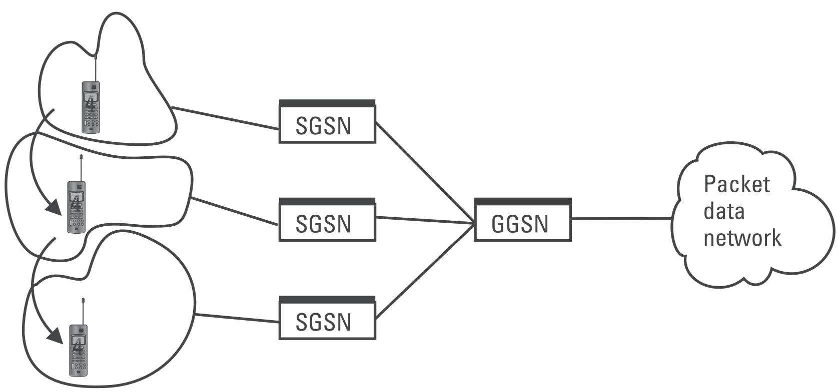


Figure 2.20 The anchor function of the GGSN.

of internal network resources, including the air interface in particular, the GGSN is concerned with collating all charges regarding external network resources. This differentiation is important in the case of roaming or the choice of a particular GGSN by the mobile (see also Section 2.4.4).

2.4.3.4 Types of GGSN

Our example in Figure 2.21 should make clear that GPRS supports basic different types of GGSNs. Although the functions of these GGSNs are basically identical, the differences lie in the various capabilities of the GGSN types A, B, and C.

- *GGSN type B:* GGSN type B is a standard GGSN that is always selected by an SGSN in the same PLMN during a PDP context activation when a mobile station only requests IP service and dynamic IP address allocation. The SGSN accordingly selects the standard GGSN.
- *GGSN type A:* For a type A GGSN, a network operator must become an Internet service provider (ISP). A Dynamic Host Configuration Protocol (DHCP) server, a dynamic name system (DNS) server, and an assortment of dynamic IP addresses are thus required. A customer who receives his mobile service and his Internet service from the same operator will always access this GGSN. Of course, this does not exclude the possibility of a type B GGSN having access

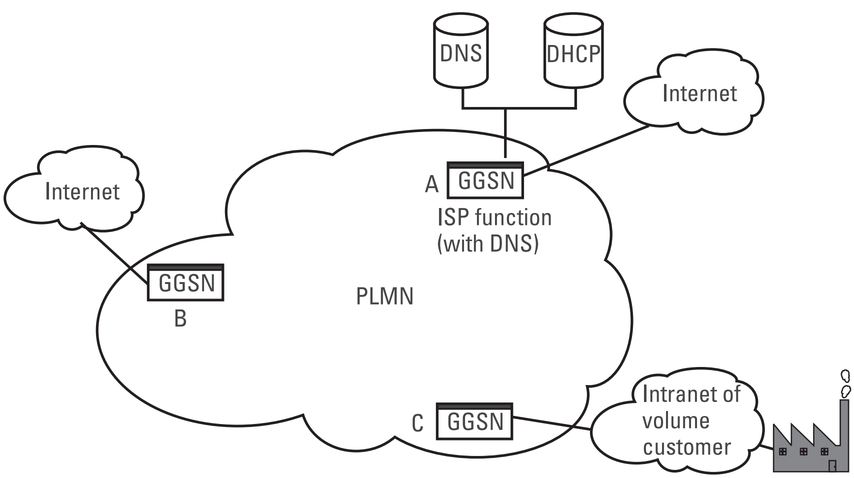


Figure 2.21 Versions of GGSN.

to the respective Internet server. What is decisive is that a mobile station, or the terminal software connected to it, must be configured so that this particular GGSN can always be identified, even during a PDP context activation. The access point name (APN, 3GTS 23.003) parameter, which serves an SGSN as a reference to a GGSN, exists for this purpose. Note that this process also works during roaming in another network.

- *GGSN type C:* A type C GGSN offers one of the most interesting possibilities. Imagine a globally operating company in which many employees frequently require mobile access to the intranet. A permanent e-mail link to these employees is also imaginable. One of the management's most important requirements in such cases is the safe and encrypted transmission of information that may well be confidential. Accordingly, such a company can make an agreement with a GSM/GPRS network operator and reserve a GGSN for itself. Here, too, the selection of the GGSN during a PDP context activation ensues via the APN. Again, a secure transmission channel must exist between the GGSN and the company's intranet.

2.4.4 The Border Gateway

As illustrated in Figure 2.22, the border gateway (BG) is the connection between two PLMNs and is therefore of decisive importance for inter-PLMN roaming.

Physically, a border gateway can also be part of an SGSN or GGSN. What is interesting is that the IP network between two PLMNs can also be the nonsecure Internet. Note that the border gateways must always be used when a particular GGSN is selected by a mobile station during a PDP context activation. If the subscriber is in another PLMN at this time, the SGSN/GGSN connection must be set up via a border gateway.

2.4.5 Overview of the GSM/GPRS Network as a Whole

As has become clear, new network elements are added to those already used in GSM for GPRS. The BSS is used equally by GPRS (packet switched)

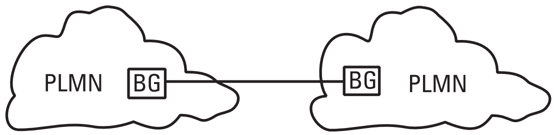


Figure 2.22 The border gateway as a link between two PLMNs.

and GSM (circuit switched). As Figure 2.23 illustrates, interfaces are required between some purely GSM network elements and the new GPRS network elements. These are the Gc, Gd, Gf, Gr, and Gs interfaces. Note that the Gs interface is optional. Where it is available, the Gs interface contributes decisively to increasing the performance of GPRS and decreasing the burden on the mobile station. This will become clear in Section 3.2.6.

Incidentally, all the interfaces named above are based on SS7 and use the GSM–mobile application part (MAP).

The other interfaces, apart from the air and Abis interfaces, are exclusively designed for the communication of GPRS-specific network elements and are packet switched in their construction. IP is used exclusively for the connection between GSNs as well as between GSNs and BGs and CGs in particular. Since this is an IP network individual to one operator, either IPv4 or IPv6 can be used.

Note that similarly to the BSS, the HLR in GPRS is used equally by both services. In fact, the HLR, in the framework of GPRS, is expanded considerably because in addition to previous data, all the GPRS-specific data of a subscriber must be stored there.

2.5 The Mobile Station in GPRS

Although today (in 2003) most mobile stations with GPRS capability still look the same as they did before the introduction of GPRS, we assume that in the not-too-distant future the introduction of GPRS will have considerable influence on the outward appearance of mobile stations.

High-end mobiles will bear a great deal of similarity to PDAs and the telephoning function will be categorized with the other functions of such

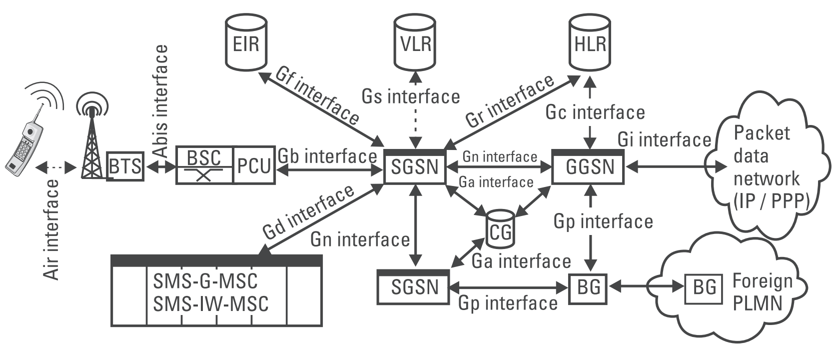


Figure 2.23 Interfaces between the GSM and the GPRS network elements.

devices, such as e-mail software, Internet browser, address book, etc. These devices enable either the simultaneous operation of both types of function (i.e., telephoning and sending/receiving e-mail) or enable either telephoning or working on the Internet. The synchronization between desktop or laptop and these mobile stations will ensue either via a normal docking station or via Bluetooth.

In addition to these high-end mobile stations, totally new opportunities will come to light. An example of one such opportunity is pure GPRS mobile stations in the form of PC cards for laptops, which can provide the user with a permanent link to the intranet or Internet even when mobile.

Such pure GPRS mobile stations are also of interest for a whole series of mobile Internet applications. Although we would not dare to look for the much sought-after killer application, we should present some thoughts at this point. Two of the most interesting mobile Internet applications are object monitoring for alarm systems or navigation systems for cars. In both cases, and for many of the other applications being developed, the GPRS mobile station does not require a telephone function. Finally, the normal GSM telephone, which also, in its upper price range, makes GPRS available for occasional use together with the laptop, will also continue to exist.

Only the more distant future will tell which of these three possibilities will form the majority of the new mobile stations.

Technically, GPRS also introduces class A, B, and C mobile stations, independently from the multislot classes already known from HSCSD. The essential features of the multislot classes are now presented.

2.5.1 The Multislot Classes

Table 2.1 lists all the multislot classes, as defined in 2GTS 05.02. Note that for HSCSD, only the multislot classes 1 to 18 can be used. GPRS can also use the multislot classes 19 to 29. The multislot classes are divided into two types: type 1 and type 2.

2.5.1.1 Type 1 Multislot Classes

Type 1 mobile stations do not need to be able to transmit and receive at the same time. In other words, only semi or half-duplex operation is possible. For this reason, type 1 mobile stations need neither a duplexer to enable simultaneous transmission and reception via the same antenna nor a second antenna as an alternative to a duplexer. An exception here is the type 19 to 29 mobile stations, in which several time slots can be combined. Data transmission can only be operated in either uplink or downlink at any one time with these multislot classes 19 to 29.

Table 2.1
The Different Multislot Classes

Type	Multislot Class	Max No. of Receive Time Slots	Max No. of Transmit Time Slots	Sum Rx + Tx
1	1	1	1	2
1	2	2	1	3
1	3	2	2	3
1	4	3	1	4
1	5	2	2	4
1	6	3	2	4
1	7	3	3	4
1	8	4	1	5
1	9	3	2	5
1	10	4	2	5
1	11	4	3	5
1	12	4	4	5
2	13	3	3	N/A
2	14	4	4	N/A
2	15	5	5	N/A
2	16	6	6	N/A
2	17	7	7	N/A
2	18	8	8	N/A
1	19	6	2	N/A
1	20	6	3	N/A
1	21	6	4	N/A
1	22	6	4	N/A
1	23	6	6	N/A
1	24	8	2	N/A
1	25	8	3	N/A
1	26	8	4	N/A
1	27	8	4	N/A
1	28	8	6	N/A
1	29	8	8	N/A

Multislot class 12 is one item that requires further explanation. This seems to allow the simultaneous combination of four time slots in uplink and downlink directions. The limitation here is on the maximum number of time slots supported at the same time (see the right-hand column of Table 2.1), which is only five for multislot class 12. In other words, up to four time slots in either the uplink or downlink direction and the corresponding delta in the opposite direction can be combined. This is illustrated in Figures

2.24 and 2.25. Here, a further restriction for the multislot classes becomes visible. All channels allocated to one direction in a mobile station cannot be interrupted by allocations in the other direction. Also note that type 1 mobile stations must also perform other activities such as neighboring cell level measurements and the measurement of the SCH of these neighboring cells. These activities must be performed in the unoccupied time slots.

2.5.1.2 Type 2 Multislot Classes

Type 2 mobile stations, in contrast to type 1, allow simultaneous receiving and transmitting in the same mobile station. As explained above, a duplexer is required that can separate uplink and downlink signals, or two antennas must be installed. It is clear that type 2 mobile stations are more expensive to develop and produce, but are indispensable for high throughput rates.

2.5.2 GPRS: The Class A, B, and C Mobile Stations

It seems confusing in the general context that with the introduction of GPRS, a further classification of mobile stations becomes necessary in addition to

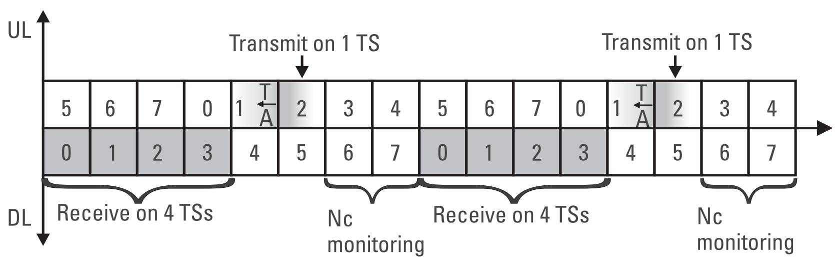


Figure 2.24 Example of the operation of a multislot class 12 with neighboring cell level measurements, taking timing advance into account. The mobile station receives on four time slots and transmits on one.

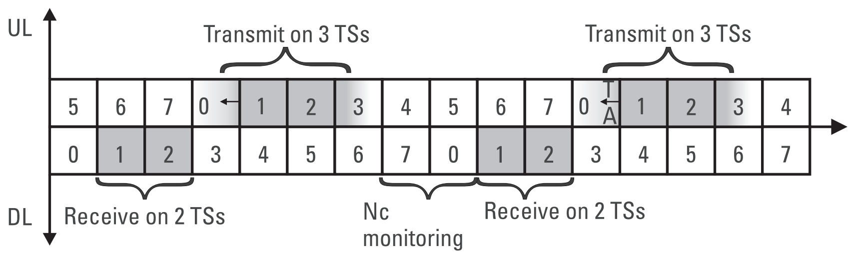


Figure 2.25 Example of the operation of a multislot class 12 with neighboring cell level measurements, taking timing advance into account. The mobile station receives on two time slots and transmits on three.

the multislot classes. This additional classification is provided by the division of GPRS mobile stations into classes A, B, and C. We have already mentioned that there is a high degree of independence between GPRS classes A, B, and C and the multislot classes.

2.5.2.1 The Class A GPRS Mobile Station

In general, it can be said that a class A GPRS mobile station offers the user maximum performance because circuit switched and packet switched services can be operated at the same time. This statement is best explained by an example.

Class A mobile stations support simultaneous telephoning and data transmission. While one is telephoning with the mobile station, the mobile station can be used as a GPRS modem for the laptop. Since GPRS supports always-on, data transmission can ensue both from the mobile station and the network. Anyone who transmits a lot of data with GSM-CSD can understand the advantages of this operating mode immediately.

This operating mode implicitly requires a mobile station to be able to follow the other paging channel while one operating mode is already active. During a telephone call, the GPRS class A mobile station must not only perform all the tasks connected to this but also be able to follow the paging channel, which would indicate an imminent downlink data transmission. This also applies vice-versa—that is, when a GPRS data connection is active and someone tries to call the subscriber. Here we arrive back at the presence of the Gs interface between the VLR and the SGSN, which provides considerable alleviation in such a case. The details, however, cannot be handled until speech has been transmitted via the packet data channels (PDCHs) (see Section 3.2.6). Whatever the case may be, the GPRS class A mobile station must be able to follow the other paging channel when active, even without a Gs interface.

2.5.2.2 The Class B GPRS Mobile Station

GPRS class B mobile stations do not have as high a performance as GPRS class A mobile stations. Their limitation is that they do not support simultaneous operation of packet switched and circuit switched services. During a telephone call, the mobile station cannot be used simultaneously as a GPRS modem and vice-versa.

On the other hand, a GPRS class B mobile station must be able to follow both paging channels—the one for circuit switched services and the one for packet switched services—without an active telephone call or GPRS data transmission. If, however, one of these services has been decided upon, then a GPRS class B mobile station no longer needs to follow the other paging channel. This means that the user is no longer available for the other service. It is clear that the presence of a Gs interface is also of considerable advantage in this case. As already mentioned, this will be discussed in detail in Section 3.2.6.

2.5.2.3 The Class C GPRS Mobile Station

A GPRS class C mobile station is restricted even further in its capabilities regarding simultaneous operation of circuit switched and packet switched services.

The user has to decide between circuit switched and packet switched support when switching the device on. After this decision has been made, the GPRS class C mobile station will follow either the circuit switched or the packet switched paging channel.

Although this class C is by no means unimaginable in a normal mobile station, it is also excellently suited for devices that only support GPRS features. Examples of this would be laptop PC cards or those types of mobile station that are only intended to be able to send data to an Internet server while mobile. Here, the GPRS class C mobile station simply disappears into the housing of the application and therefore becomes reduced to a kind of wireless Internet socket.

References

[1] Heine, G., *GPRS, EDGE, HSCSD and the Path to 3G,* Norwood, MA: Artech House.

[2] Heine, G., *GSM-Signalisierung verstehen und praktisch anwenden,* FRANZIS Verlag, 1997.

3

The Air Interface in GPRS

3.1 The 52 Multiframe

In the first chapter, we paid special attention to describing the frame hierarchy with 26 multiframe and 51 multiframe. One of the reasons for this is the definition of an additional multiframe type especially for GPRS. This new 52 multiframe with a period of exactly 240 ms is essential for understanding the resource administration and operation of GPRS.

Note that the 52 multiframe corresponds in terms of period to exactly two consecutive 26 multiframes as illustrated in Figure 3.1. For this reason, the function of synchronizing the mobile station to the BTS does not have to be altered (see also 1.6.6.1). In short, the 52 multiframe number is obtained by halving the 26 multiframe number. The current GPRS burden determines how many time slots on which ARFCNs traverse the 52 multiframe instead of the 26 multiframe and 51 multiframe. At a given point, there may only be one time slot configured for GPRS, which means using the 52 multiframe, whereas later, all eight time slots may be reserved on this frequency for GPRS and therefore configured with the 52 multiframe. Whatever the case may be, GPRS and GSM cannot coexist on one time slot. Depending on the burden, the services are switched from one to the other. We are pointing this out so clearly here because this is one of the most frequently asked questions in our GPRS courses.

3.1.1 Structure and Multiplexing on the 52 Multiframe

The decisive factor for GPRS is the division of a 52 multiframe into 12 radio blocks, where each radio block is four repetitions of the same time

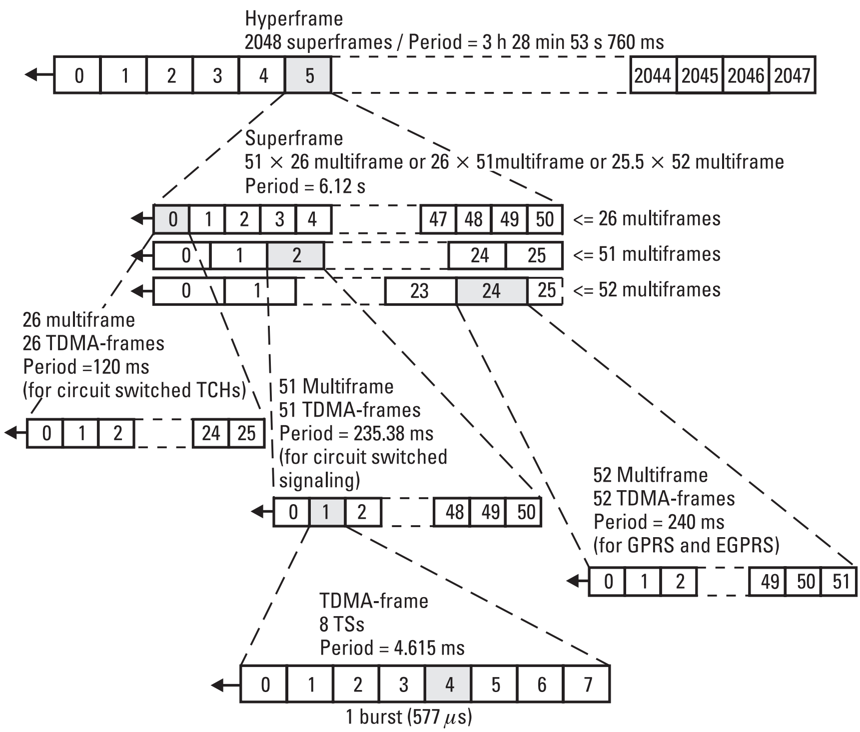


Figure 3.1 The GSM frame hierarchy with the 52 multiframe for GPRS.

slot. Figure 3.2 shows the fine structure of the 52 multiframe with the 12 radio blocks and the TDMA frames for timing advance control and interference measurements.

It is the radio blocks that constitute the atomic resource unit of GPRS. While a user in GSM always receives a complete time slot as a resource for data transmission, only radio blocks, as parts of time slots, are allocated in GPRS. In this way, several subscribers can share a single time slot. A further peculiarity of GPRS is that the resources in the uplink and downlink directions are almost completely independent from one another. In particular, the allocation of radio blocks for data transmission in GPRS is either uplink or downlink. Of course, bidirectional data transmission is also possible and is used. These, however, require two resource allocations, one per direction. There is no certainty that a subscriber will be allocated the same time slot(s) for transmission in both directions. This depends in particular on the multislot class of his device.

In other words, a subscriber only receives the resources that are actually needed at any given time in accordance with the resource on demand concept

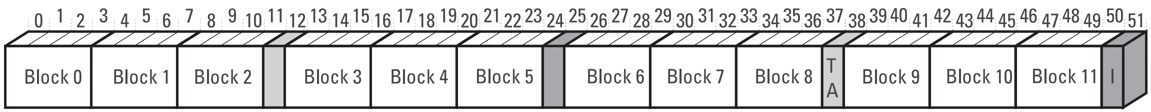


Figure 3.2 The fine structure of the 52 multiframe.

of packet switched data transmission. Here, a single time slot in each direction can be used almost simultaneously by several subscribers.

At this point, it is often asked how many subscribers can be allocated resources simultaneously on one time slot and per direction. It is still too early to answer this question. However, we would like to point out that in accordance with the 52 multiframe period, not more than 12 subscribers can use the same time slot within 240 ms and per direction. The transmission rate is also accordingly low in such a case.

Mixed configurations on an ARFCN are of course also possible—in fact, they are predominant. As Figure 3.3 illustrates, several time slots for circuit switched GSM can be configured on one frequency, while GPRS subscribers share the other time slots.

Resource allocation in GPRS will be discussed in a later section.

3.2 The Packet Data Channels

A whole series of new logical channels was defined for the operation of GPRS, as illustrated in Figure 3.4. The first factor to note is that most of these channels are defined only in the downlink direction or only in the uplink direction.

In a BTS with GPRS support at least a few of these channels must be available [packet data traffic channel (PDTCH) and packet timing advance control channel (PTCCH)], while the presence of other PDCHs [packet broadcast control channel (PBCCH) and packet common control channel (PCCCH)] is optional. Note that the PDCHs have no counterpart to the SDCCH. A dedicated control channel is therefore totally absent.

The following sections will go into more detail on the various PDCHs.

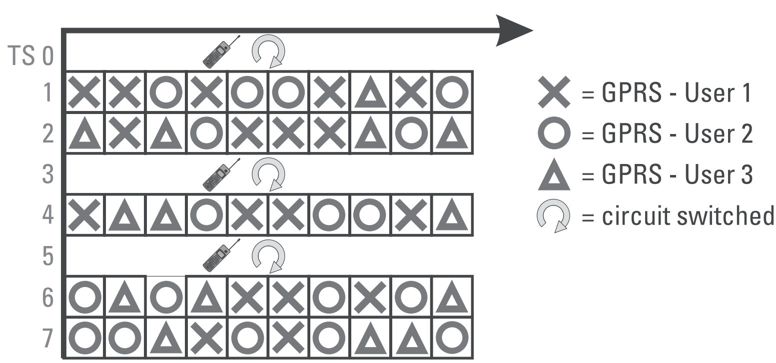


Figure 3.3 Example of resource sharing of GSM and GPRS on an ARFCN.

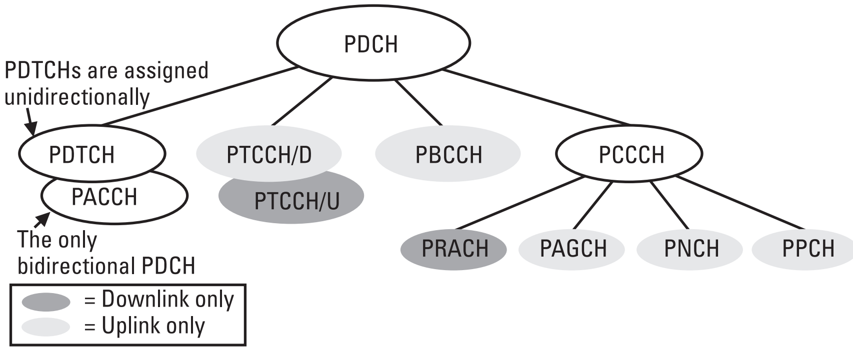


Figure 3.4 The various packet data channels and their directional dependence.

3.2.1 The PBCCH

The PBCCH does not necessarily have to be configured in a base station that supports GPRS. If no PBCCH is configured, the GPRS mobile stations will find all the GPRS-relevant information in the SYS_INFO13 message, newly defined for GPRS, which must be transmitted from every BTS with GPRS via the BCCH. The presence of the SYS_INFO13, and therefore the support of GPRS in a cell, will be announced in at least one of BCCH/SYS_INFO3, 4, 7, or 8. Figure 3.5 shows an example of a SYS_INFO13 message, recorded with a K1205 from TEKTRONIX.

If the PBCCH is configured in a cell, a SYS_INFO13 will also be sent via the BCCH, but this SYS_INFO13 will then only contain an indicator as to the position of the PBCCH. Note that only one single PBCCH may be configured per BTS. It is decisive here that, contrary to the restrictions for the BCCH, the PBCCH can be configured on every time slot and every carrier (ARFCN) of a BTS. However, the PBCCH may only use blocks 0, 3, 6, and 9 of a 52 multiframe where the actual occupation and repeat rate of the PBCCH can be configured.

If the PBCCH is available, class A and B GSM/GPRS mobile stations may only camp on the PBCCH (i.e., they do not have to monitor the BCCH and the PBCCH).

Accordingly, there are several options for a mobile station when it is selecting a cell:

- A BTS either supports GPRS or not—SYS_INFO3, 4, 7, 8 announces the presence of the SYS_INFO13.

```
+---------+-----------------------------------------------+-----------------------------------+
|12:44:34,604,383  1 -2 down SI13 (System Information Type 13)                                 |
|                                                                                               |
|System Information Type 13                                                                     |
|                                                                                               |
|0110---- |Protocol Discriminator                         |Radio Resource Management (RR)     |
|----0000 |Skip Indicator                                 |Skip Indicator                     |
|00000000 |Message Type                                   |System Information 13              |
|101----- |BCCH Change Mark                               |5                                  |
|---0000- |SI Change Field                                |Update of unspec.SI or SI message  |
|-------0 |Bit                                            |0                                  |
|0------- |Bit                                            |0                                  |
|***b8*** |Routing Area Code                              |11                                 |
|-0------ |SPGC Support                                   |not supported on CCCH in this cell |
|--110--- |Priority Access Thr                            |Pkt access allowed prio level 1 to 4|
|-----00- |Network Control Order                          |NC0: MS controlled, no meas report |
|1 GPRS Cell Options                                                                            |
|***b2*** |Network Mode of Operation                      |Network Mode of Operation III      |
|-111---- |Timers 3166 and 3168                           |4000 ms                            |
|----010- |Timer 3192                                     |1500 ms                            |
|***b3*** |DRX Timer Max                                  |Not Supported                      |
|--0----- |Access Burst Type                              |8 bit access burst shall be used   |
|---1---- |Packet Control Ack default format              |Default format is RLC/MAC control bl|
|----1111 |BS CV MAX                                      |15                                 |
|1------- |Bit                                            |1                                  |
|-010---- |PAN_DEC                                        |2                                  |
|----010- |PAN_INC                                        |2                                  |
|***b3*** |PAN_MAX                                        |Max Value T3102 = 20               |
|--0----- |Bit                                            |0                                  |
|1 (end of) GPRS Cell Options                                                                   |
|2 GPRS Power Ctrl Param Struct                                                                 |
|---0011- |Alpha                                          |3                                  |
|***b5*** |T_AVG_W                                        |15                                 |
|***b5*** |T_AVG_T                                        |5                                  |
|-0------ |Measurement Channel Type                       |BCCH                               |
|--1010-- |N_AVG_I                                        |10                                 |
|2 (end of) GPRS Power Ctrl Param Struct                                                        |
|------00 |Spare Padding                                  |0                                  |
+---------+-----------------------------------------------+-----------------------------------+
```

Figure 3.5 Example of a SYS_INFO13 message.

- The SYS_INFO13 either contains all GPRS-specific cell information or not—depending on whether the SYS_INFO13 only contains a pointer to the PBCCH.
- If the SYS_INFO13 only contains a pointer, the mobile station that wishes to use GPRS services, among others, should camp on the PBCCH.
- If the SYS_INFO13 contains all the GPRS-specific cell information, the mobile station should continue to camp on the BCCH and, amongst other things, listen in to the SYS_INFO13.

Since the PBCCH does not have to be configured, the question is then raised as to when a PBCCH actually has to be installed. This question can be answered as follows: if one or more time slots with PBCCH (i.e., PRACH, PPCH, and PAGCH) are to be configured in a cell, the SYS_INFO13 is no longer sufficient to identify the respective time slots. This is exactly where the extended possibilities of the PBCCH are required.

3.2.2 The PCCCH

As illustrated in Figure 3.4, we should not really talk of "the" or "only one" PCCCH. The term PCCCH is rather the collective term for the packet access grant channel (PAGCH), PPCH, packet notification channel (PNCH), and PRACH. Similarly to the PBCCH, the PCCCH can be configured in a cell, but there is no absolute necessity for its presence. In contrast to PBCCH, several time slots within one cell can certainly carry the PCCCH at least partially. However, every mobile station must follow only one PCCCH or the respective time slot. In order to identify this time slot, there is a formula specified in GSM 05.02 that must be used by the respective mobile station to determine its PCCCH time slot.

Sections 2.3.1.2 and 2.3.1.3 present the special problems of the burden on the common control channels (CCCH) due to GPRS. These sections are essential for understanding the allocation of PCCCHs and the background to this as regards this flexibility.

If there is no PCCCH configured in a cell, all network access in conjunction with GPRS will take place via the CCCHs of circuit switched GSM, which will definitely be available. This applies equally to both the uplink and the downlink. This version will work without any problems as long as the capacity of the CCCHs stays within the limits set by slotted Aloha (uplink) or the capacity of the PCH (downlink). If the burden approaches these critical values, however, the GPRS-compatible BSS should be capable of allocating at least a PCCCH and therefore also a PBCCH, independently.

3.2.2.1 The PPCH

Similarly to the PCH, which we already know from GSM, the main task of the PPCH is the notification of a mobile station with regard to a downlink transmission request.

The PPCH can, however, in what is known as the ready state (GMM; see Section 4.1.2.4), also be used for direct resource allocation in the downlink direction to the mobile station.

It must be noted, however, as described in the previous section, that both functions, (i.e., the actual paging and direct resource allocation in ready state) can be performed via the PCH without configured PCCCH/PPCH.

3.2.2.2 The PRAC

There are, of course, many similarities between the RACH, which is known from GSM, and the PRACH, which is reserved for GPRS. Both are used for transmitting an access burst via the mobile station to the base station. For this reason they both only exist in the uplink direction.

The essential difference between the RACH and the PRACH is that the number of information bits in an access burst transmitted via an RACH is limited to eight, while in PRACH 8 or 11 bits are possible.

In this context, however, it must be considered that the mobile station cannot choose freely between these two possibilities. On the contrary, the parameter *ACCESS_BURST_TYPE* (GSM 04.60) specifies which of the two formats is actually to be used by the mobile stations.

The next frequently asked question refers to the additional 3 bits. What extra information is contained in these 3 bits? There is no standard answer to this. One possibility is that the mobile station already states its multislot class in the access burst. Further possibilities include the quantification of the resources actually requested by the mobile station or a greater number of random bits for preliminary identification of the mobile station.

3.2.2.3 The PAGCH

Like the access grant channel (AGCH) in GSM, the PAGCH allocates resources to a mobile station after it has requested them from the PRACH. The only difference between the AGCH and the PAGCH is that the AGCH in GSM only allocates an SDCCH while the PAGCH principally allocates resources on PDTCHs. Of course, the AGCH in GPRS is also used for the allocation of PDTCHs if there is no PCCCH. The decisive difference can therefore only be made out between the application of the AGCH in GSM and its application in GPRS. Although the standard expressly permits the allocation of TCHs in GSM by the AGCH too, this is not practiced because at the time the AGCH is used, it is not yet clear which type of TCH is required.

3.2.2.4 The PNCH

The remaining PCCCH is the PNCH, which is envisaged for the future point-to-multipoint (PTM) functions of GPRS. In such a case, the PNCH informs the participating mobile stations of the imminent PTM transaction.

3.2.3 The PDTCH and the PACCH

The essential difference between the TCHs as known from GSM and the PDTCHs that are used in GPRS is that PDTCH resources are principally allocated in one direction only. This does not exclude the possibility of resources being provided in both directions on demand. These, however, will then be independent from each other. Furthermore, different simultaneously active mobile stations must, by definition, actually share the resources of a single time slot. This means that a PDTCH can be regarded as the sum of the radio blocks used on one time slot.

Note that PDTCHs are only defined on time slots with 52 multiframe. It is also important that every mobile station can only occupy resources on one ARFCN during a transaction (Figure 3.6). This explains why, in GPRS, any mobile station can only be allocated resources on a maximum of eight time slots. Despite this restriction, slow frequency hopping (SFH) can also be used in GPRS, because even in such a case a mobile station only transmits or receives on one frequency at any given time.

As Figure 3.7 illustrates, an additional bidirectional PACCH is defined that can only be used for the transmission of RLC/MAC control information for every unidirectional PDTCH. The question of how resources are provided for the bidirectional PACCH will be explained in Section 3.8.

3.2.4 The PTACC

The PTCCH is dependent on direction. For this reason, the uplink PTCCH must be precisely differentiated from its downlink counterpart. The PTCCH is only defined for the TDMA frame numbers 12 and 38 in the 52 multiframe. Both PTCCHs are required for the performance of what is called the continuous timing advance update procedure, which will be introduced in Section 3.3.

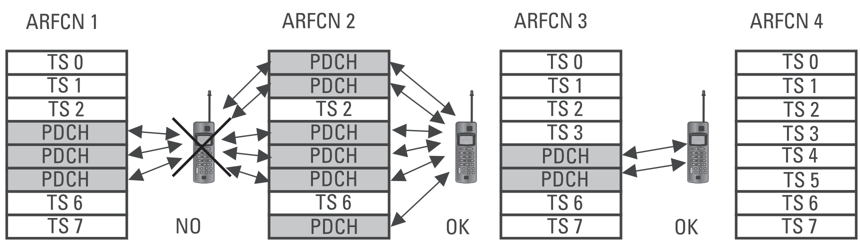


Figure 3.6 GPRS resources can only be occupied on one ARFCN.

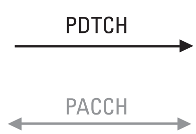


Figure 3.7 Each PDTCH has a bidirectional PACCH.

3.2.5 Multiplexing of Different PDCHs on One Time Slot

In Chapter 1, the 51 and 26 multiframe types were introduced. What is particularly noticeable about the 51 multiframe is its relatively rigid structure. First, only control channels such as SDCCH, BCCH, or FCCH can be installed on it. Second, a set configuration can only be changed with great difficulty. For example, in what is known as the combined configuration, we find on time slot 0 of C0 (Carrier 0), in addition to BCCH (set frame numbers), FCCH/SCH (also set frame numbers) and CCCH (only variable between PCH and AGCH), an additional four SDCCH subchannels with their SACCHs, also in set positions. As previously mentioned, although such a configuration can be carried out by the operator, they are very rarely carried out in practice.

The situation with the 52 multiframe is exactly the opposite. Although there is also the limitation that the PTCCH can only appear at positions 12 and 38 and that the PBCCH can only be transmitted in blocks 0, 3, 6, and 9, some or all of these blocks can be used equally for PDTCH or a PCCCH in the following 52 multiframe.

So how does a mobile station identify what kind of channel type has been used for a given block? The answer to this is simple, since identification is made using the 2-bit *payload type* information element in the MAC header (see Section 5.1.2). On the other hand, this solution means that a channel type cannot be determined until the appropriate content has been received and decoded. Furthermore, traffic channels are multiplexed with control channels on the 52 multiframe, in which the configuration can constantly be adapted according to demand and ideally, automatically (i.e., without the operator having to intervene).

The PRACH, the only uplink PCCCH, is worthy of special mention. According to demand and the current burden, a different number of blocks per 52 multiframe and even a different number of time slots can be allocated to PRACH through the system.

In other words, the multiplexing of logical channels on the 52 multiframe is many times more flexible than one is accustomed to from circuit switched GSM.

3.2.6 The Network Operation Modes (NOM I, NOM II, and NOM III)

Now that the various PDCHs have been introduced, we must once again examine the network architecture of GPRS and its connection with the availability of the PCCCH. The optional Gs interface does actually permit coordination of paging for incoming calls. Table 3.1 presents the context

Table 3.1
The Connections Between NOM and Paging

	Mobile Station State	**CS Paging**	**PS Paging**
NOM I PCH available PPCH optional *Gs interface is available*	RR idle	SGSN (on PPCH, if available; otherwise on PCH)	SGSN (on PPCH, if available; otherwise on PCH)
	RR active	VLR (on FACCH via call waiting indication)	SGSN (on PPCH, if available; otherwise on PCH. Applicable only for mobile station class A)
	Packet idle	SGSN (on PPCH, if available; otherwise on PCH)	SGSN (on PPCH, if available; otherwise on PCH)
	Packet transfer	SGSN (on PACCH)	SGSN (on PACCH)
NOM II No PPCH available *No Gs interface*	RR idle	VLR (on PCH)	SGSN (on PCH)
	RR active	VLR (on FACCH via call waiting indication)	SGSN (on PCH)
	Packet idle	VLR (on PCH)	SGSN (on PCH)
	Packet transfer	VLR (on PCH)	SGSN (on PACCH)
NOM III PCH and PPCH available *No Gs interface*	RR idle	VLR (on PCH)	SGSN (on PPCH)
	RR active	VLR (on FACCH via call waiting indication)	SGSN (on PPCH)
	Packet idle	VLR (on PCH)	SGSN (on PPCH)
	Packet transfer	VLR (on PCH)	SGSN (on PACCH)

in detail. In this way, the SGSN can also trigger pagings for circuit switched transactions when a Gs interface (NOM I) is implemented. These contexts are illustrated in Figure 3.8. The respective network operation mode is communicated to the surrounding mobile stations via the system information in the SYS_INFO13 message (Figure 3.5) or the PACK_SYS_INFO messages on the PBCCH/PACCH.

An interesting fact is that an existing Gs interface not only has positive effects on the execution and availability of pagings, but also facilitates other procedures such as attachment.

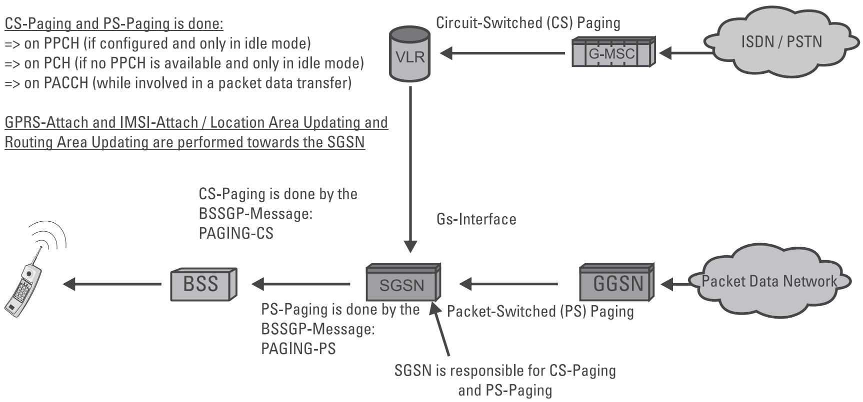


Figure 3.8 The network in NOM I.

- It is especially important that MM transactions such as attachment and location or routing area updating can be performed in combination when a Gs interface is available. Class A and B mobile stations therefore only have to carry out one combined procedure instead of two different ones. The SGSN and the VLR must then be synchronized via the Gs interface.
- The performance capability of the network is also increased by the availability of a Gs interface in that even a class B mobile station remains available for circuit switched paging (calls) while in active packet transfer mode. The person being called can then decide whether to continue the packet data transmission or to answer the call. As we already know, a class B mobile does not have to follow the other paging channel when one service has already been activated.
- Network operation modes II and III do not have a Gs interface. These are different in that they have a packet paging channel, which is only configured when for example, there is a higher GPRS burden. Without a PPCH, the network is in NOM II, while it operates in NOM III with a PPCH.
- In NOM II, packet switched and circuit switched network elements (i.e., the SGSN and the VLR) are independent of each other. However, both SGSN and VLR carry out their paging via the circuit switched PCH.
- In NOM III, as in NOM II, packet switched and circuit switched network elements are independent of each other. However, in NOM III a PPCH is configured, which is a disadvantage for the mobile

station, because class A and B mobile stations have to follow two different time slots, which may also be on two different carriers, in order to listen in to pagings.

A book cannot sufficiently describe the connections between NOM and mobile station class. However, you will find more information on this topic on our Web site, in the form of a demo CD that you can download.

3.3 Timing Advance Control in GPRS

In GPRS, network resources are only allocated to a mobile station for as long as they are actually needed. As explained in the description of the PDTCH, this also means that usually either uplink only or downlink only resources are allocated. Bidirectional transmission, while not being the exception, can certainly not be taken for granted. However, knowing the distance from the mobile station is important even for a downlink-only transmission, because the mobile station must at least be able to transmit confirmations of receipt of a given protocol stack to the network.

The continuous regulation of the timing advance (i.e., the transmission time of the uplink signals) is based on the measurement of the bursts from the mobile station that arrive periodically at the BTS, or, to be more precise, the training sequence code in these bursts. This process has already been described in its entirety in Section 1.4.2.

If these periodical uplink transmissions are absent, the timing advance cannot be regulated in the normal way. This will be described below. One must further consider that access bursts must also be used in GPRS by the mobile station until a valid timing advance value is available. As explained in Chapter 2, this is ensured by an access burst also being transmitted to the BTS in GPRS using the slotted Aloha method. This access burst, amongst other things, serves to estimate the distance between the mobile station and the BTS. The following two methods are thus valid for the regulation of timing advance when the mobile station is in active transmitting or receiving mode.

The traditional method of measuring timing advance is also possible in GPRS. As previously mentioned, the position of the training sequence code can be used in the uplink bursts received for the adjustment of timing advance.

3.3.1 The Continuous Timing Advance Update Procedure

The standard procedure in GPRS for the regulation of timing advance is called the *continuous timing advance update procedure*.

This procedure is based on the logical channels PTCCH/U and PTCCH/D, which are defined in the frame numbers 12 and 38 of the 52 multiframe. As illustrated in Figure 3.9, the uplink direction of the PTCCH/U is segmented into 16 subchannels via eight consecutive 52 multiframes, where a subchannel can be allocated to every active mobile station. Allocation takes place as a time slot and timing advance index (TAI) on the subchannel within a resource allocation message. Figure 3.10 shows an example in which a PACK_DL_ASS message is presented. The period of such a subchannel is thus 416 TDMA frames, or exactly 1.92 seconds.

3.3.1.1 The Course of the Continuous Timing Advance Update Procedure

After a PTCCH/U subchannel has been allocated, a mobile station waits until the allocated PTCCH/U TDMA frame appears on this time slot and

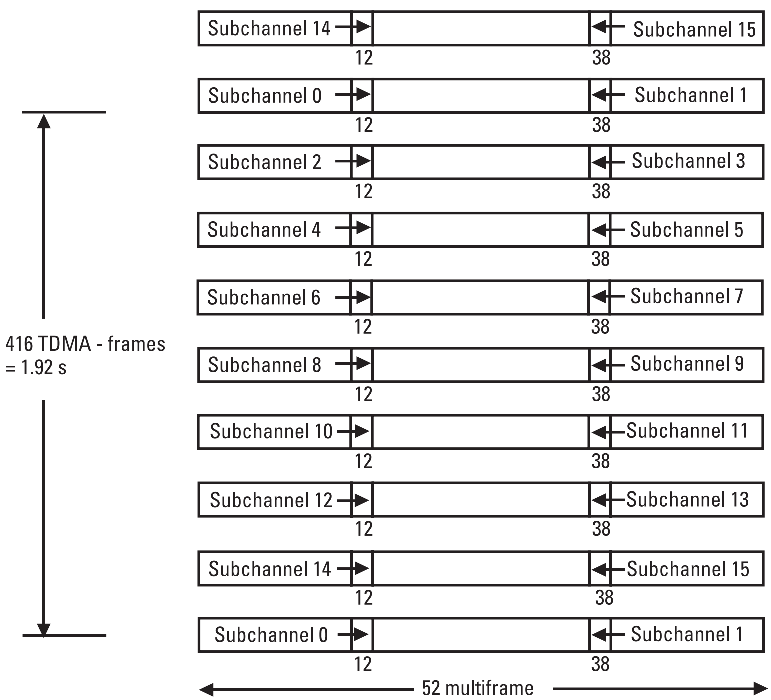


Figure 3.9 The definition of the 16 PTCCH/U subchannels.

```
+---------+------------------------------------------+----------------------------------+
|BITMASK  |ID Name                                   |Comment or Value                  |
+---------+------------------------------------------+----------------------------------+
|20:18:35,720,119  1 -2 down  PCU-DOWN  DATA SYNC  RMAC-DOWN  PDAS                      |
|Downlink (PCU-DOWN)  DATA SYNC (= Data with Syncronisation)          |
|Data with Syncronisation                                                               |
|00000001 |Mapping Flag                              |Data in RLC-MAC format            |
|00000000 |Spare                                     |0                                 |
|----100- |Frame type                                |Data                              |
|---0---- |Code Field                                |Coding Scheme 1                   |
|------0- |next expected uplink burst                |4 normal bursts                   |
|-----0-- |Parity-even Bit of int. Block number      |0                                 |
|***b4*** |Power Control                             |15                                |
|--111--- |Uplink State Flag                         |7                                 |
|GPRS RLC/MAC V2.4.0 Downlink (RMAC-DOWN)  PDAS (= Packet Downlink Assignment)          |
|Packet Downlink Assignment                                                             |
|01------ |Payload Type                              |RLC/MAC Control Block, no opt.octets|
|--00---- |Relative Reserved Block Period            |TDMA frame N+13                   |
|----0--- |Supplementary/Polling                     |RRBP field is not valid           |
|-----000 |Uplink State Flag                         |0                                 |
|000010-- |Message Type                              |2                                 |
|Core Parameter                                                                         |
|------00 |Page Mode                                 |Normal Paging                     |
|0------- |Bit                                       |0                                 |
|-1------ |Bit                                       |1                                 |
|--0----- |Bit                                       |0                                 |
|**b32*** |TLLI                                      |c1c00a45                          |
|---0---- |Bit                                       |0                                 |
|----00-- |MAC Mode                                  |Dynamic Allocation                |
|------0- |RLC Mode                                  |RLC Acknowledged Mode             |
|-------0 |Control Ack                               |New Downlink TBF Established      |
|0------- |Timeslot 0 Allocation                     |Timeslot is not assigned          |
|-1------ |Timeslot 1 Allocation                     |Timeslot is assigned              |
|--1----- |Timeslot 2 Allocation                     |Timeslot is assigned              |
|---0---- |Timeslot 3 Allocation                     |Timeslot is not assigned          |
|----0--- |Timeslot 4 Allocation                     |Timeslot is not assigned          |
|-----0-- |Timeslot 5 Allocation                     |Timeslot is not assigned          |
|------0- |Timeslot 6 Allocation                     |Timeslot is not assigned          |
|-------0 |Timeslot 7 Allocation                     |Timeslot is not assigned          |
|1 Packet Timing Advance                                                                |
|0------- |Bit                                       |0                                 |
|-1------ |Bit                                       |1                                 |
|--0000-- |Timing Advance Index                      |0                                 |
|***b3*** |Timing Advance Timeslot No                |2                                 |
|1 (end of) Packet Timing Advance                                                       |
|-0------ |Bit                                       |0                                 |
|--1----- |Bit                                       |1                                 |
|2 Frequency Parameters                                                                 |
|---000-- |Training Seq Code                         |0                                 |
|------00 |Bit                                       |00                                |
|**b10*** |ARFCN                                     |102                               |
|2 (end of) Frequency Parameters                                                        |
|--1----- |Bit                                       |1                                 |
|---00000 |Downlink TFI Assignment                   |0                                 |
|1------- |Bit                                       |1                                 |
|3 Power Control Parameters                                                             |
|-0011--- |Alpha                                     |alpha = 0.3                       |
|-----0-- |Bit                                       |0                                 |
|------1- |Bit                                       |1                                 |
|***b5*** |Gamma TN1                                 |3                                 |
|----1--- |Bit                                       |1                                 |
|***b5*** |Gamma TN2                                 |3                                 |
|--0----- |Bit                                       |0                                 |
|---0---- |Bit                                       |0                                 |
|----0--- |Bit                                       |0                                 |
|-----0-- |Bit                                       |0                                 |
|------0- |Bit                                       |0                                 |
|3 (end of) Power Control Parameters                                                    |
|-------0 |Bit                                       |0                                 |
|0------- |Bit                                       |0                                 |
|-0101011 |Spare Padding                             |43                                |
+----+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+
|HEX |0 |1 |2 |3 |4 |5 |6 |7 |8 |9 |A |B |C |D |E |F |
+----+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+
|0   |01|00|89|3f|40|08|58|38|01|48|a0|60|41|20|19|a0|
|10  |9a|38|c0|2b|2b|2b|2b|2b|2b|2b|2b|02|e0|ff|ff|ff|
|20  |ff|ff|ff|ff|ff|ff|ff|ff|  |  |  |  |  |  |  |  |
```

Allocation of the timing Advance parameters
1) Timeslot numbers
2) Timing advance index (TAI)

Repetition of the Message Context in hex.

Figure 3.10 Example of a PACK_DL_ASS message with allocation of a PTCCH/U subchannels to time slot 2.

transmits an access burst with an assumed timing advance of 0 to the BTS within this frame.

The coding of this access burst is defined precisely, but the information field can, according to the system parameters (*ACCESS_BURST_TYPE*), contain either 8 or 11 bits of information.

The BTS receives the access burst and can estimate the distance to the mobile station, and thus the timing advance, using the time of entry.

These timing advance values are embedded in the newly defined timing advance message, which is illustrated in Figure 3.11, for up to 16 mobile stations.

The following points are important with regards to the timing advance message and transmission to the mobile stations:

- The timing advance values in the timing advance message are 7 bits wide and thus also allow extended cell operation.
- The continuous timing advance update procedure allows up to 16 mobile stations to synchronize timing advance on the same time slot.
- The timing advance message is transmitted per time slot as a broadcast message and thus does not require a header. Each mobile station obtains its timing advance value from the timing advance message according to the TAI.
- The timing advance message is expanded from 16 to 23 octets using filler octets. This happens so that 184 input bits can be handed over to the channel coder, which then become 456 bits in all during channel coding.
- Since the timing advance message is 456 bits long after channel coding, segmenting into four parts is necessary before transmission so that four normal bursts in the downlink direction can transmit the timing advance message on the PTCCH/D.
- The segmenting of a timing advance message via four occurences of the PTCCH/D is shown in Figure 3.12. Note that the timing

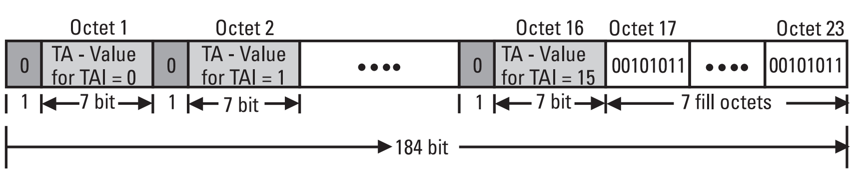


Figure 3.11 The timing advance message.

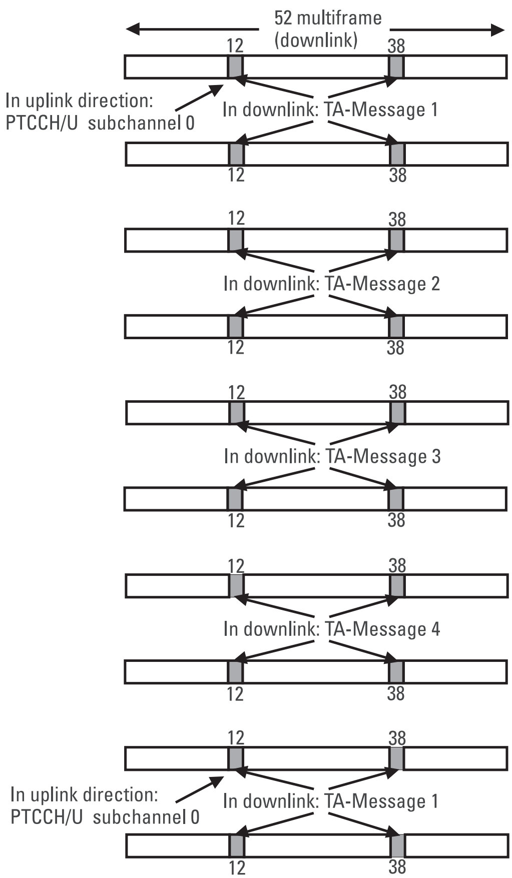


Figure 3.12 Transmission of the new timing advance values in the timing advance messages.

advance message is also subordinated to the block interleaving (i.e the bits within the four normal bursts are packaged) in accordance with the rules for SACCH familiar from GSM.

- Only those four mobile stations that have sent their access burst to the BTS in the PTCCH/U in the two 52 multiframes before this timing advance message actually receive an updated timing advance value per timing advance message. For the other mobile stations, or for the remaining TAIs, the "old" timing advance values are repeated. If for any reason a mobile station has not been able to read its new timing advance value from a timing advance message, it can still do so in the following timing advance messages.

Example in Regard to Figure 3.12. A mobile station transmits its PTCCH/U access burst in subchannel 0. It therefore recevies its updated timing advance value in the next TA message, no. 2. Furthermore, this new value is repeated in the following timing advance messages until there is a new value.

3.3.2 Timing Advance by Means of Polling and Access Bursts

This is probably the most unusual method for determining the distance from the mobile station and consequently the transmission time of the uplink signal.

Here the base station polls the mobile station as required, and the mobile station transmits a PACK_CTRL_ACK message in the displayed uplink radio block. This message actually consists of four identically formatted access bursts with 8 or 11 information bits (Figure 3.13). These access bursts are transmitted with a timing advance of 0 and can thus be used for determining the distance.

The coding of a complete PACK_CTRL_ACK message into only 11 bits is purely convention. Nondifferentiated contents can, of course, also be transmitted here, but this is not the aim of this message.

Since these four access bursts are identically formatted, receiving access bursts with the same coding by coincidence from background noise is more or less impossible.

Although this method may at first seem unconventional, it is certainly justifiable in a transaction with rare uplink transmissions. This is because the polling and transmission of the access bursts is only carried out when a new timing advance value is actually needed, and not periodically as with the continuous timing advance update procedure.

3.4 The Coding Schemes CS-1 to CS-4

In GPRS, higher transmission rates can be achieved than in GSM because it is possible for time slot concentration to take place (i.e., the use of several time slots by the same mobile station).

What is more important for this section is that in GPRS, new channel coding methods are used that provide higher payload rates per time slot. However, the laws of physics must also be considered here. This means that with GMSK as the modulation method and the normal burst as a medium, the highest possible net transmission rate is 22.8 Kbps per time slot. "Net" simply means that the payload rate can only lie below the limiting value. This value can be explained easily.

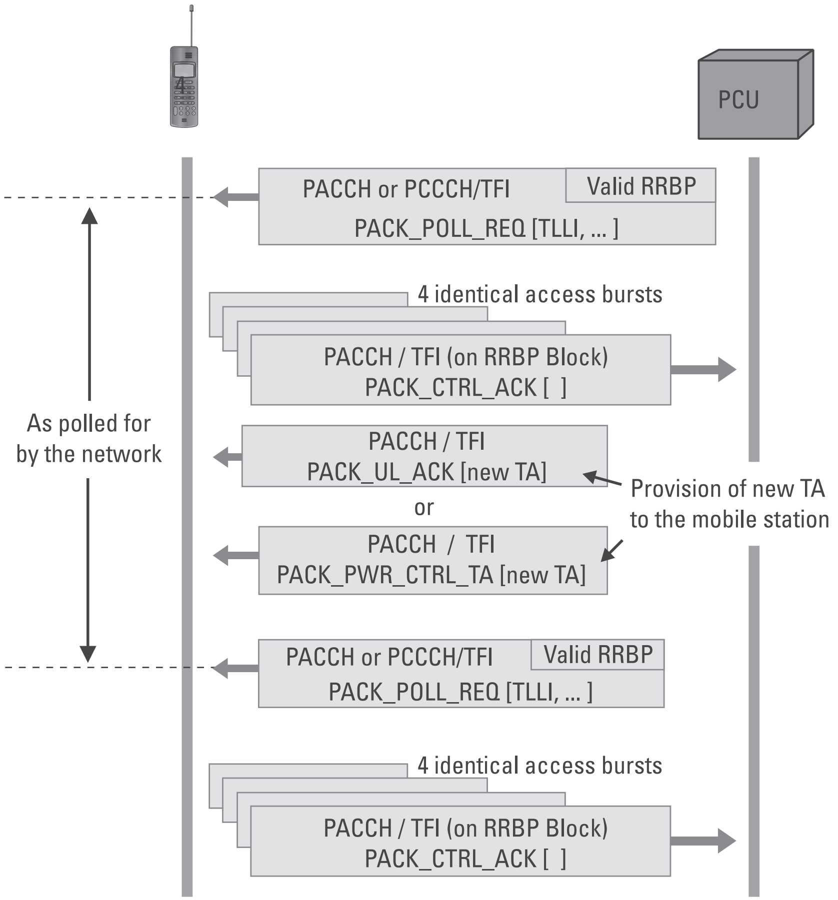


Figure 3.13 The use of polling as an alternative to the determination of the timing advance.

How does the net transmission rate of 22.8 Kbps in GSM arise?

1. In a 26 multiframe with a period of 120 ms, 24 × TCH, 1 × SACCH, and 1 × nothing are transmitted. Normal bursts are used for the transmission. This produces a result of 24 × 114 = 2,736 bits. This value of 2,736 bits is divided by the period of the 26 multiframe of 120 ms: 2,736/120 ms = 22.8 Kbps.
2. In GSM, TRAU frames are sent to the BTS every 20 ms and must then be completely processed and transmitted every 20 ms. After channel coding, every TRAU frame is 456 bits. This produces a result of 456 bits/20 ms = 22.8 Kbps.

Answer 1 is certainly more concrete, while answer 2 is easier to comprehend. The derivation of the limit value of 22.8 Kbps in GSM was important to us to show that even such refined methods as these cannot push the net transmission rate above 22.8 Kbps.

What can be done, however, is to modify the channel coding in GSM such that more payload information can be contained in each TRAU frame. Note that in the case of full-rate speech, there are only 260 payload bits in each TRAU frame. For 9.6 Kbps of data there are only 192 payload bits plus 48 synchronization and signaling bits (see Section 1.7.1.1). The delta to the 456 bits, which are then transmitted via the air interface, is accordingly large.

As explained in Sections 1.6.3.3 and 1.6.3.4, these additional bits are used sensibly for the FEC. If more payload bits and less redundant space are transmitted via the air interface, the probability of transmission errors, and therefore the amount of payload that has to be transmitted more than once, also increases. This then automatically leads to a reduction in the payload rate.

On the other hand, the radio channel is often of sufficient quality to be able to dispense partially or entirely with the FEC. This especially applies if the mobile station is not in motion or is moving very slowly at a short distance from the base station.

An acceptable compromise would therefore be an adaptive method in which the degree of FEC can be chosen flexibly, according to the actual given bit error rate. Such a method must be able to react quickly enough to changing reception conditions and switch stepwise between maximum FEC and no FEC at all.

Precisely this method was chosen for GPRS, and it is reflected in the four coding schemes CS-1 to CS-4.

Figure 3.14 illustrates the principal connection between payload rate and FEC. Clearly there is no form of FEC whatsoever with CS-4, and transmission errors can only be corrected with repeated transmission. The use of CIRCUIT SWITCHED-4 is thus restricted to such situations in which a mobile station has a very good wireless link to the base station (direct line of sight). The use of CS-1 is the other extreme. Here the payload rate is moderate, but CS-1 can still be used even on a very poor radio channel.

It will become clear that CS-2 to CS-4 may only be used on PDTCHs, while CS-1 can be used both for packet switched control channels (PPCH, PAGCH, PNCH, PBCCH, PTCCH/D) and PDTCHs.

When regarding Figure 3.14, also consider the statement that the base station only has to support CS-1, while the mobile station must master all

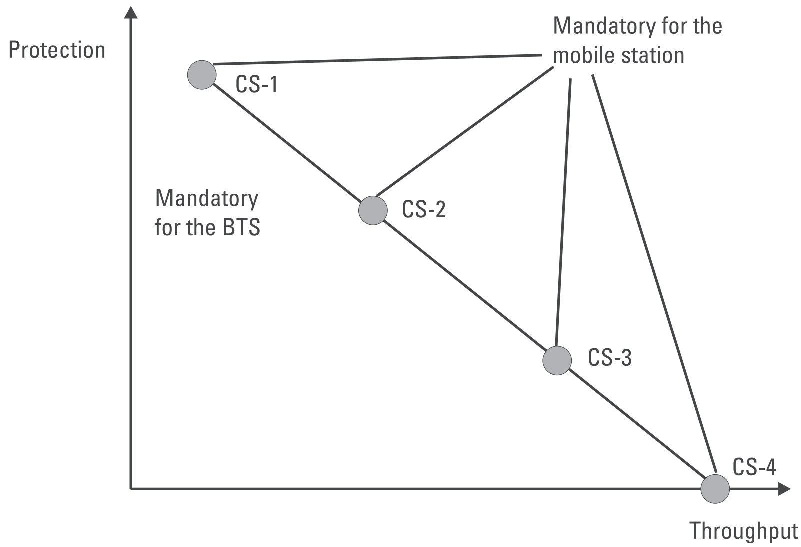


Figure 3.14 The connection between payload rate and FEC.

the coding methods, CS-1 to CS-4. This measure is in place to prevent the situation arising in which the customer can only buy devices supporting CS-1 during the starting phase of GPRS. It is difficult, however, to follow this demand when special mobile GPRS devices are only envisaged for telemetric use. Such devices clearly do not require higher coding schemes.

What is also questionable here is that the PCU frames to be used between PCU and BTS (see Section 2.4.1.1) can no longer be used via the existing 16-Kbps channel with CS-3 and CS-4. As previously mentioned, radical (and thus expensive) measures are required on the network side before CS-3 and CS-4 can actually be supported.

The coding methods CS-1 to CS-4 will be presented in detail below. It will then become clear how much payload can actually be transmitted with each method.

3.4.1 Coding Scheme 1

As illustrated above, CS-1 can be used for packet switched control channels and for PDTCHs. Figure 3.15 accordingly shows the CS-1 for PDTCHs, which always transmit logical link control (LLC) segments, while Figure 3.16 shows the CS-1 for control channels. The difference between the two is the different size of the RLC header. The CS-1 was not specially defined for GPRS but is actually the coding method for circuit switched control

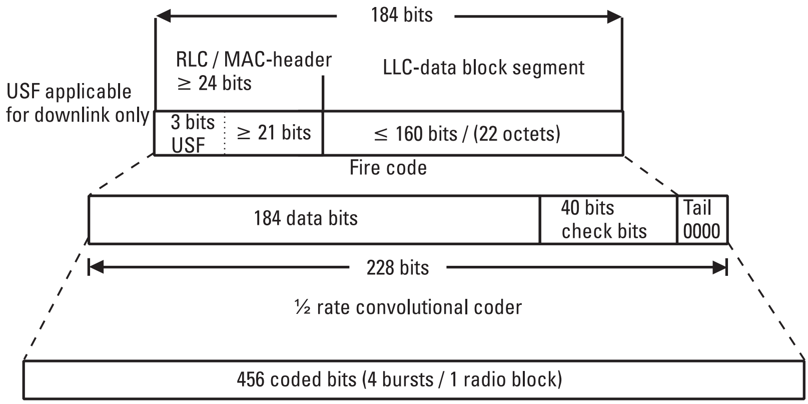


Figure 3.15 CS-1 in the coding of LLC data.

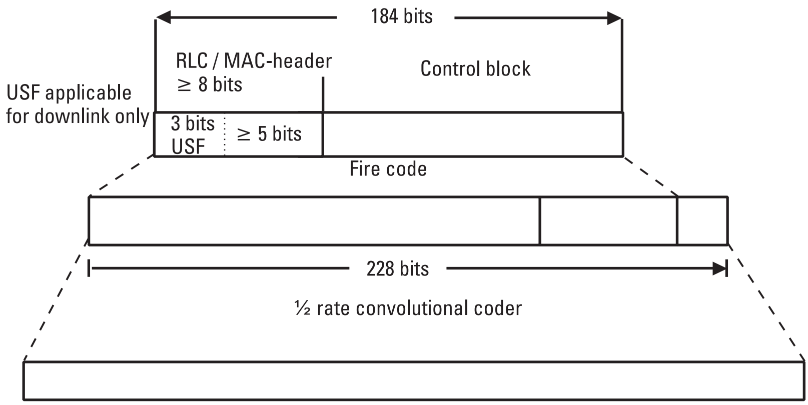


Figure 3.16 CS-1 in the coding of RLC/MAC control messages.

channels such as the SDCCH or the AGCH known from GSM. The significance of the uplink state flag (USF) will be explained in a later section.

In the channel coding of PDTCHs with CS-1, the total length of the RLC/MAC data block is 184 bits. Eight of these bits are the MAC header, while the size of the RLC header is at least 16 bits. The necessity for further RLC header octets depends on whether or not segments of several LLC frames are contained in the RLC/MAC data block. The additional header octets would be used for delimiting such segments.

In any case, a maximum of 160 data bits is contained in such an RLC/MAC data block (CS-1). If one also considers that such an RLC/MAC data block is sent from or to the PCU every 20 ms, the maximum net throughput rate for RLC/MAC at CS-1 is 160 bits/20 ms = 8 Kbps. This value is approximately –3% of the actual net throughput rate of GPRS, which is why we want to use it for CS-1 from now on.

In control channels, the RLC/MAC header is at least 8 bits long and can be as long as 24 bits if the optional downlink RLC header is available. The remaining bits contain the RLC/MAC control messages.

In the first step of CS-1, the fire code for BEC, already known from GSM, is used. Accordingly, 40 check bits are added on to the 184-bit-long block.

Independently of whether LLC segments or control channels are to be channel coded, channel coding in CS-1 proceeds as follows.

The second step is the addition of 4 tail bits, coded with 0000bin, whose function has already been described in Section 1.6.3.3.

The third step is then convolutional coding with a half-rate convolutional coder (see Sections 1.6.3.3 and 1.6.3.4). There are 456 channel-coded bits at its output, which are intermixed according to the interleaving rules for the SACCH and then transmitted in four bursts or a radio block. Incidentally, the half-rate convolutional coder is the same one that is used for the channel coding of the TCH/FS.

3.4.2 Coding Schemes 2 and 3

The main, but by no means only difference between CS-1 and CS-2 and CS-3, which we will examine now, is the number of input bits. In CS-2, 264 bits are fed to the channel coder; in CS-3 it is 312 bits. As Figure 3.17 illustrates, a maximum of 240 and 288 bits, respectively, actually contain payload.

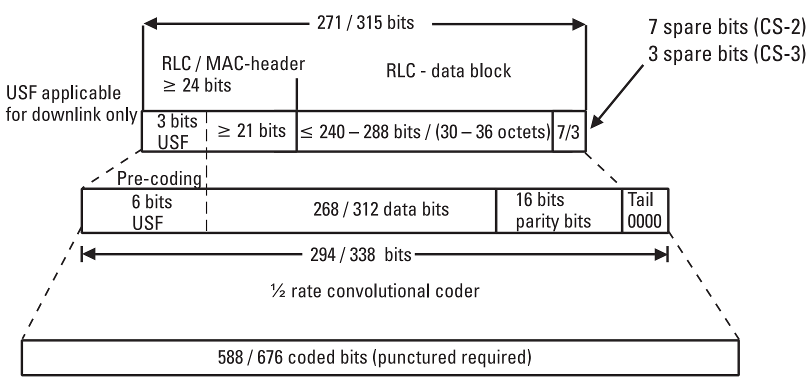


Figure 3.17 Channel coding according to CS-2 and CS-3.

This gives for CS-2 a maximum net throughput rate of 240 bits/20 ms = 12 Kbps. Accordingly, the value for CS-3 is 288 bits/20 ms = 14.4 Kbps. As for CS-1, these values correspond to the actual net throughput rate with an accuracy of approximately −3%.

In CS-2 and CS-3, 7 and 3 spare bits, respectively, coded with 0, are attached to the data block. Now back to the actual process of channel coding. In the first step of CS-2 and CS-3, 16 parity bits are produced, which are necessary for determining the block error rate. It is then possible, by using these parity bits, to perform BEC.

Precoding is then carried out for the first 3 bits of the MAC header. This precoding is intended to afford particularly good protection for the downlink parameter, USF. Although precoding also takes place in the uplink direction, it only happens there for reasons of simplification. Different coding schemes for the uplink and downlink directions are not desirable.

The USF occupies the first 3 bits of the MAC header in all four coding schemes in the downlink direction. The USF is so important because it is used for resource allocation to the mobile stations in the uplink direction. Almost all active mobile stations have to read the USF constantly whether the respective downlink lock is addressed to them or not. Misinterpretations of these 3 bits would therefore be fatal. This information regarding USF should be mentioned at this point, although we will go into considerably more detail on the USF later.

First, let us return to the channel coding of CS-2 and CS-3. As usual, 4 tail bits are attached to the data block before the 294 or 338 bits are led to the half-rate convolutional coder. This doubles the number of bits, so that there are 588 or 676 bits at the output. As with CS-1, the half-rate convolutional coder is the same one that is used for the channel coding of the TCH/FS.

Clearly, the first 12 bits in the channel coded block now form the strongly secured USF.

By now, the attentive reader will have noticed the problem with CS-2 and CS-3. The 588 or 676 channel-coded bits do not fit into four bursts, which have a total transmission capacity of 456 bits. For this reason, we must apply the principle of puncturing, which was introduced in Section 1.6.3.5, for CS-2 and CS-3.

Accordingly, at least 132 bits in CS-2 and 220 bits in CS-3 are deleted from the block of channel-coded bits before interleaving takes place according to the principles for the SACCH. Whoever may think of the special protection for the USF at this point need not be concerned. The first 12 bits are never touched by the puncturing process.

After puncturing there are 456 bits, which can be mapped onto four normal bursts or a radio block after interleaving.

In accordance with the statements already made in Section 1.6.3.5, the coding rate of CS-2 is calculated to be approximately 2/3 and that of CS-3 approximately 3/4.

Calculation of the Coding Rates for CS-2 and CS-3

- Number of input bits on the channel coder/number of output bits = coding rate;
- For CS-2: 294 bits/456 bits = 0.645 ≈ 2/3;
- For CS-3: 338 bits/456 bits = 0.741 ≈ 3/4.

3.4.3 Coding Scheme 4

Compared to CS-1, CS-2, and CS-3, coding scheme 4 is the simplest form of processing. Here there are at least 431 bits at the input, which means that considering the maximum of 456 bits, the addition of redundant bits must be excluded from the start. The only exception here is made for the USF, which is given special treatment as it is in CS-2 and CS-3. The first 3 bits in the MAC header (i.e., the USF in the downlink direction) are precoded to 12 bits in order to enable higher transmission security even when the channel quality is poor.

The precoding table used is defined such that the 12 bits at the output look as if the 3 bits at the input have passed through the half-rate convolutional coder twice in succession.

If one now remembers that at the end of the coding process, the USF is also folded out to 12 bits in CS-2 and CS-3, it becomes clear that for CS-2, CS-3, and CS-4, the first 12 bits of the channel-coded data block no longer depend on the respective coding scheme, but only on the USF. This procedure clearly makes it easier for the mobile stations to identify the USF, since the mobile stations cannot know which coding scheme the respective downlink data block is actually using.

In order to calculate the block error rate, 16 additional parity bits are also calculated and attached to the data block in CS-4. A data block in CS-4 is thus exactly 456 bits long. As in the other coding schemes, these 456 bits are subordinated to the same interleaving scheme as is defined for the SACCH and is then transmitted in four bursts or a radio block.

Finally, the maximum net throughput rate has to be determined for CS-4. As illustrated in Figure 3.18, there are a maximum of 400 data bits

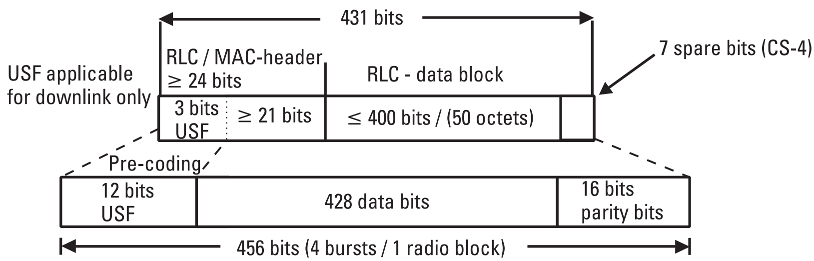


Figure 3.18 Coding according to CS-4.

in a CS-4 data block. This gives, at a transmission duration of 20 ms, a maximum net throughput rate of 400 bits/20 ms = 20 Kbps. As with CS-1, CS-2, and CS-3, this value is accurate to −3%.

3.4.4 Use of the Different Coding Schemes

Table 3.2 summarizes the most important key data of the four coding schemes used in GPRS. The net throughput rates are given for one time slot. However, the same user requires exclusive access to all radio blocks of this time slot.

As already mentioned in Chapter 2, channels with more than 16 Kbps are required for the use of CS-3 and CS-4 on the Abis interface and in the BSC, which is why these coding schemes require a considerable increase in performance, even outside the BTSs and mobile stations. Such an increase in performance also means considerable expenditure.

Independently of this, the different coding schemes can, being controlled by the PCU, be adapted to the changing channel conditions. The

Table 3.2
The Most Important Framework Parameters for CS-1 to CS-4

	Max. No. of Data Bits	Bits for BEC	Punctured Bits	Coding Rate	Max. Net Throughput Rate (Per Time Slot)
CS-1	160	40	0	≈ 1/2	8 Kbps
CS-2	240	16	132	≈ 2/3	12 Kbps
CS-3	288	16	220	≈ 3/4	14.4 Kbps
CS-4	400	16	0	≈ 1	20 Kbps

priority is maximizing the net throughput rate with a minimum of time delay.

Example. During a packet data transmission, there is initially a direct line of sight between the mobile station and the base station with little or no interference. Hence, the number of bit errors occurring is accordingly low. In this case, CS-3 or CS-4 can be used for transmission.

Note that in CS-4, every bit error that occurs requires retransmission of the entire data block. This is why the throughput rate decreases in CS-4 as soon as bit errors occur despite the maximum throughput rate on the application level. This means that the BSS must react immediately to such increased bit error rates by switching to a lower coding scheme. Imagine in our example that such bit errors occur either through interference or by the mobile station moving.

On the other hand, the BSS cannot stay at the slower CS-1 or CS-2 when the bit error rate is low. It must then switch back up to higher coding schemes as soon as the communication channel allows it to do so.

Conclusion. For GPRS, the BSS must be in a position to adapt the coding flexibly and above all dynamically to the actual given channel quality. The reaction speed of the BSS and the effective throughput rate, in conjunction with the appertaining delay times, will play a central part in the quality evaluation of a BSS. This is a new state of affairs that played no part in GSM.

3.4.5 Differentiation of the Coding Schemes

We have already mentioned that the various actively transmitting mobile stations in one cell have to evaluate the radio blocks transmitted downlink to the USF. Furthermore, a mobile station receiving data transmitted downlink must constantly receive all the data blocks transmitted on the time slots allocated and check if a data block is addressed to it itself.

In order to do this, it is essential to know the coding scheme used downlink per radio block. For this very purpose, *stealing flags,* familiar from GSM, are used in the normal bursts in GPRS and EGPRS. As Figure 3.19 illustrates, the eight stealing flags are placed in the four normal bursts of a radio block in accordance with the coding scheme used. Figure 3.19 shows, by way of example, the coding for CS-3, but there are also other sequences for CS-1 (1111 1111), CS-2 (1100 1000), and CS-4 (0001 0110).

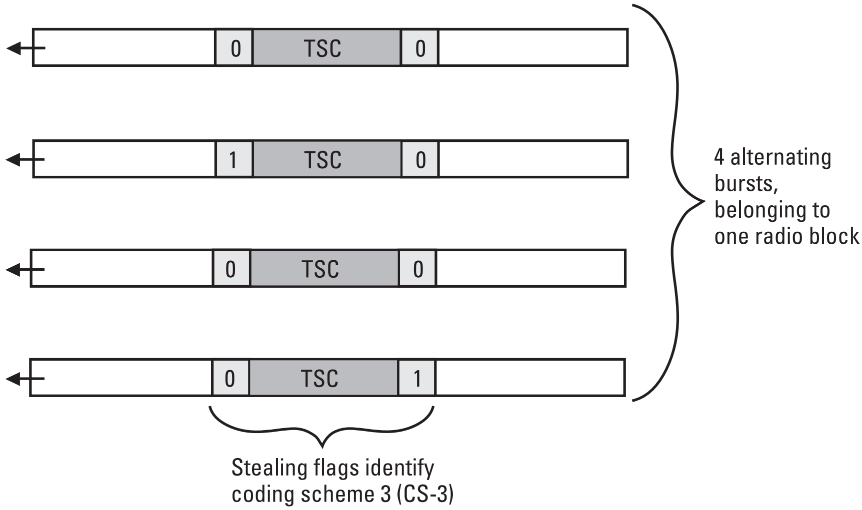


Figure 3.19 The stealing flags in the new function of identifying the coding scheme.

3.5 Identification of Data Packets

We have already referred to the fact that with the introduction of GPRS and the 52 multiframe, several users can use the same time slot uplink or downlink (almost) simultaneously, and every user can transmit and/or receive on several time slots (almost) simultaneously.

The allocation of a time slot or subchannel number to a user, as known from GSM, thus no longer functions in GPRS. For this reason, a new mechanism for identifying all data packets that belong to the same dataflow is required. Furthermore, one must also take into account the fact that in GPRS, uplink and downlink are independent from one another.

3.5.1 The Definition of TBF and TFI

- In order to identify all data packets that belong to the same uplink or downlink dataflow, the temporary block flow (TBF) is introduced in GPRS.
- A TBF encompasses all data packets in one direction that are sent from one side to the other during a dataflow. These two sides are the PCU and the mobile station. A TBF is thus defined between mobile station and PCU.
- A mobile station can only have one TBF per direction activated at any given time (i.e., a maximum total of two), one in each direction.

- In order to be able to allocate every data packet on every time slot to a TBF, the 5-bit-long temporary flow identity (TFI) is introduced (see Figure 3.20).
- Every data packet must be coded via the TFI in order to be allocated to an individual TBF.
- The TFI is a logical parameter without physical significance.

These definitions lead to the following questions:

1. How many TBFs can be defined and operated simultaneously per time slot or per ARFCN?
2. With consecutive TBFs, does a user always use the same TBF?
3. How long can a TBF be activated for?

Answer to Question 1. Up to 32 TBFs can be differentiated via the 5-bit TFI. The question remains as to what the TFI refers to—it could be the ARFCN or the time slot. Both are possible and can be supported by the system manufacturer.

- If the TFI refers to the entire ARFCN, then 32 simultaneously active TBFs per direction can be identified per ARFCN 32, which means a total of 64 TBFs per carrier.
- In an extreme case, the TFI can also refer to the individual time slot or a set of time slots. This case is illustrated in Figure 3.21. Then 2×32 TBFs can be identified per time slot, and, at least theoretically, 2×256 TBFs can be active on one carrier at the same time. This means that in theory, up to 512 mobile stations can transmit or receive data on the same ARFCN.
- As illustrated in Figure 3.21, the PCU can also collate time slots into a group. In resource allocation, however, only radio blocks can be allocated on the time slots of one group.

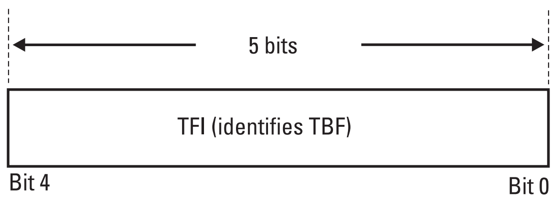


Figure 3.20 The 5-bit TFI.

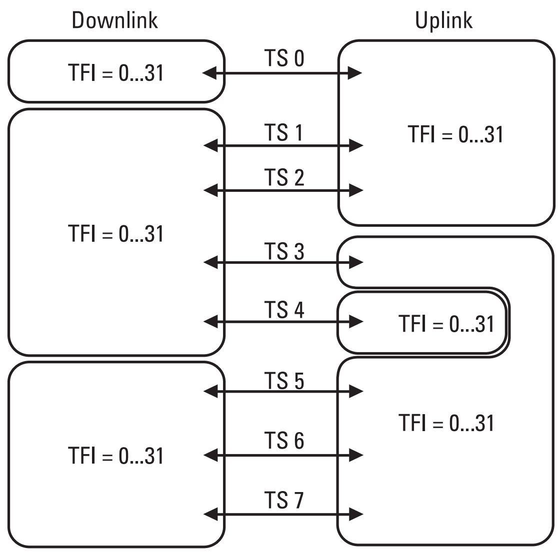


Figure 3.21 Groups of one or more time slots can be combined.

Example for Figure 3.21. Mobile station A uses TFI no. 3 on time slots 1, 2, 3, and 4 in the downlink direction. At the same time, and without any danger of ambiguity, mobile station B uses the same TFI no. 3 downlink on time slots 5, 6, and 7. This method is clearly more flexible than the restriction to "only" 64 TBFs per ARFCN. The PCU, which, for example, is then able to allocate resources to mobile stations according to their multislot class on a given group of time slots, always has control. Again, 32 different TBFs per direction are possible per group.

Answer to Question 2. The simple answer to this question is no. With consecutive TBFs, the PCU of the mobile station can allocate totally different TBFs to a mobile station. Which special TFI value is used is not relevant here. During simultaneous uplink and downlink transmission, a mobile station will also use different TFIs.

Answer to Question 3. There is no clear answer to this question in terms of hours or seconds. A TBF is active for as long as there is data to be transmitted in the output loop of the mobile station or the PCU. A TBF can thus be extremely short (e.g., only 1/10 of a second) or extremely long and can last for several hours (e.g., with file transfer).

3.6 Access to the GPRS Network

As already mentioned in Chapter 2, a packet switched network must enable more or less immediate access to network resources. Any long identification and authentication procedures should take place in advance. On the other hand, network resources should only be allocated to a subscriber for as long as they are actually required (resource on demand).

In the case of GPRS, which is to be built up on GSM and coexist with the circuit switched services in GSM, network access has to be similar to that of circuit switched GSM but also lead to immediate resource allocation. This section concentrates on the problems that occur here.

- How does a GPRS mobile station access the network and how does the network then allocate resources?
- How can ambiguity be avoided when accessing the network via the short access burst that only contains a few bits of actual information; or, to put it another way, how can all the mobile stations accessing the network at the same time be distinguished?

In GPRS, several variations for network access were defined. The network has the final say as to which variation is actually used. These variations are indicated by different access burst coding, which form the CHAN_REQ or PACK_CHAN_REQ message.

Here, one must consider that there are only two codings reserved for the CHAN_REQ message for GPRS access. In contrast to this, the PACK_CHAN_REQ message is used exclusively for GPRS access and there are thus accordingly more variations.

3.6.1 The Various Network Access Possibilities

The various network access possibilities are listed below. We will describe these alternatives in detail.

3.6.1.1 One-Phase Packet Access

- One phase packet access can be "proposed" by the mobile station on both the RACH (CHAN_REQ message) and the PRACH (PACK_CHAN_REQ message). This is the normal case for network access.
- We have just used the word "proposed." In fact, the PCU decides whether one-phase packet access is possible or two-phase packet access has to be enforced.

- In the unacknowledged RLC/MAC operation mode the mobile station cannot propose a one-phase packet access. The RLC/MAC operation mode results from the quality of service profile.
- The procedure is called "one phase" because the PCU allocates uplink resources to the mobile station as soon as it has received the access burst (PACK_UL_ASS).
- If the one-phase packet access is initiated via the RACH, resources can only be allocated on one time slot, because the multislot class of the mobile station is not known at the time of network access.
- If the one-phase packet access is initiated via the PRACH, resources can be allocated on several time slots, according to the mobile station's multislot class. This is possible because there are several bits available in the PACK_CHAN_REQ message in order to identify the multislot class in one-phase packet access.
- Note that a mobile station cannot be identified with absolute certainty in one-phase packet access because an access burst cannot transmit enough information (8 or 11 bits).
- For this reason, a contention resolution procedure must be carried out during the active uplink data transfer. As illustrated in Figure 3.22, the contention resolution procedure consists in the mobile station placing its temporary logical link identifier (TLLI) in the first max. $N3104_{max}$ data blocks. The TLLI identifies the mobile station with absolute certainty within an SGSN area. We will examine the TLLI in detail later.
- The contention resolution procedure is positively concluded when the TLLI is sent back to the mobile station in a PACK_UL_ACK message. The PCU will send this message as soon as the first data block coming from the mobile station has been received in perfect condition. From this point on, the mobile station can also transmit uplink data blocks without TLLI.
- If the network cannot receive any of the $N3104_{max}$ uplink data blocks, which can happen in particular when data blocks from different mobile stations collide, the mobile station stops its transmission and tries network access again.

3.6.1.2 Two-Phase/Single-Block Packet Access

- The procedure in two-phase packet access is shown in Figure 3.23. As with one-phase packet access, two-phase packet access can be

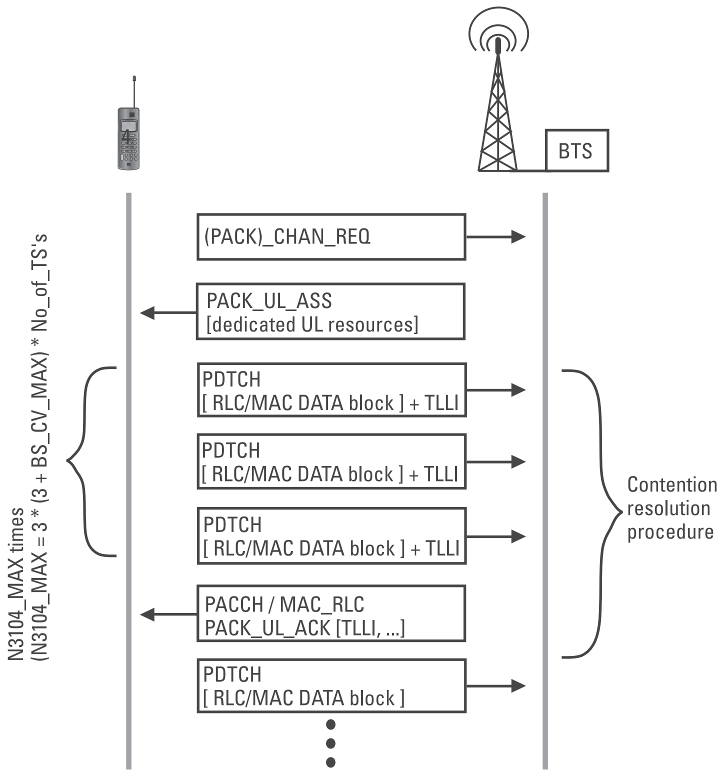


Figure 3.22 The procedure of one-phase packet access (CCCH and PCCCH).

performed on both the RACH (CHAN_REQ message) and the PRACH (PACK_CHAN_REQ message).

- The mobile station must carry out a two-phase packet access when the RLC/MAC operation mode is unacknowledged. This depends on the quality of service profile.
- In the two-phase packet access also, the mobile station can neither identify itself absolutely in an access burst, nor can it communicate exactly how many resources are required.
- For this reason, the network only allocates a single uplink block in the first phase of the two-phase packet access, which is then used by the mobile station in the second phase to send a PACK_RES_REQ message.
- The PACK_RES_REQ message primarily identifies the mobile station via the TLLI, but also indicates how many resources are needed.

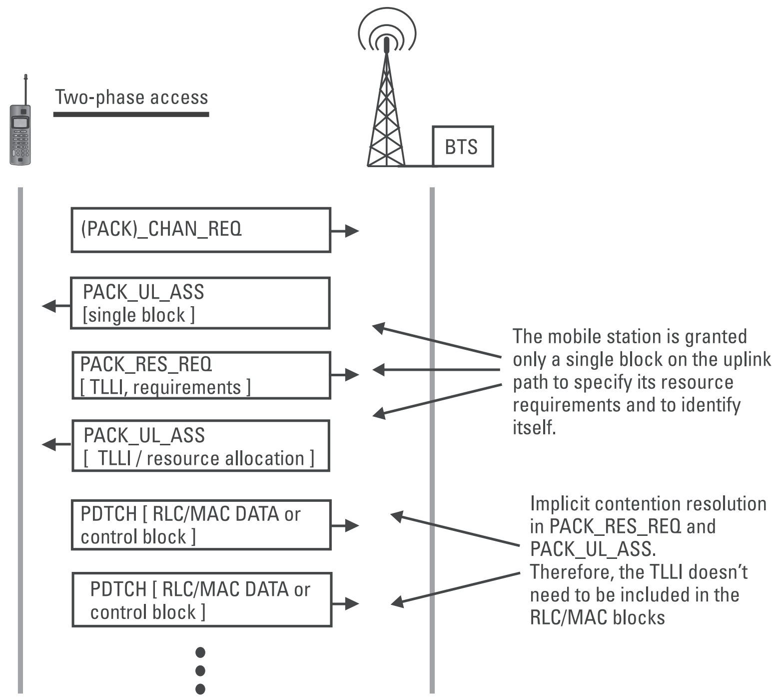


Figure 3.23 The procedures of two-phase packet access (CCCH and PCCCH).

The PACK_RES_REQ also contains relevant information on the multislot class and other radio-specific key data from the mobile station.

- The network does not allocate the necessary resources in a PACK_UL_ASS message in the second phase until it has received this PACK_RES_REQ message. Since this message contains the TLLI of the mobile stations addressed, the contention resolution procedure is also implicitly present in the two-phase packet access.
- When the mobile station uses the resources allocated, the TLLI does not have to be placed in the data blocks.

3.6.1.3 Short Access

- There are often only small amounts of data in the mobile station's transmission loop. If the amount is small enough to be transmitted with eight or fewer data blocks coded with CS-1, the mobile station should transmit a PACK_CHAN_REQ message giving *short access* as the reason instead of a one- or two-phase packet access.

- In the PACK_CHAN_REQ message for short access, there are 3 bits for naming the number of blocks required reserved for this purpose.
- Note that short access can only be initiated on the PRACH (PACK_CHAN_REQ) because only one-phase and single-block (= two phase) packet access is defined for the CHAN_REQ message on the RACH.
- In terms of procedure, short access is like one-phase packet access because a contention resolution procedure, as illustrated in Figure 3.24, also has to be carried out.

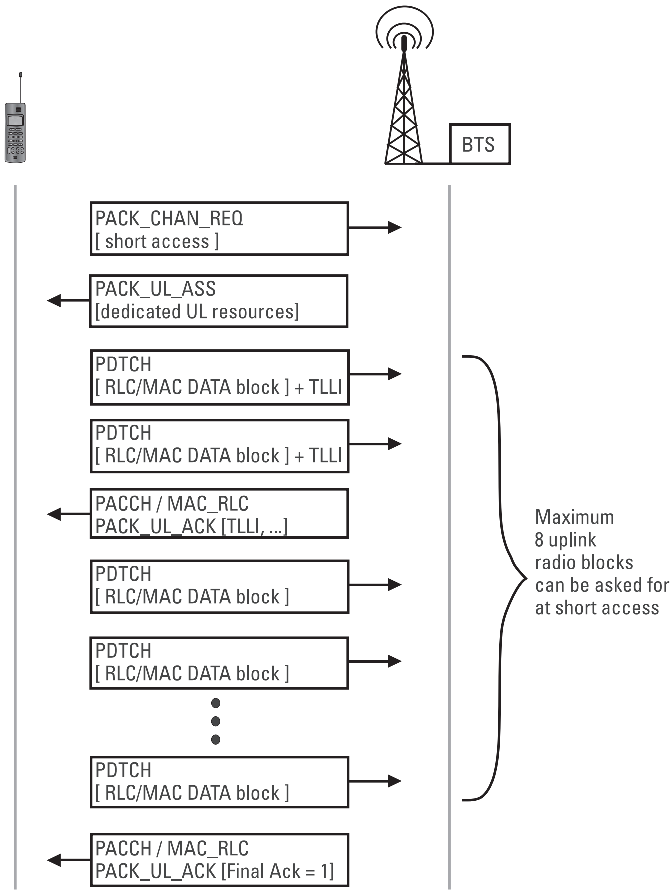


Figure 3.24 Procedures in short access (only if PCCCH is configured).

- In short access, resources can only be allocated on one time slot, because the multislot class of the mobile station is unknown.

3.6.1.4 Access for Page Response

- If a mobile station receives a paging for an imminent downlink packet data transmission via the PCU, an access burst must be sent to the network in order to transmit an empty data frame uplink first. As will become clear in the next chapter, the mobile station, by transmitting this empty data block, switches to what is called the GMM-READY state.
- If paging takes place on the PPCH—if there is also a PRACH available—the mobile station indicates *page response* as the reason for the network access in the PACK_CHAN_REQ message (Figure 3.25). Consequently, the PCU only allocates the mobile station one single uplink block.

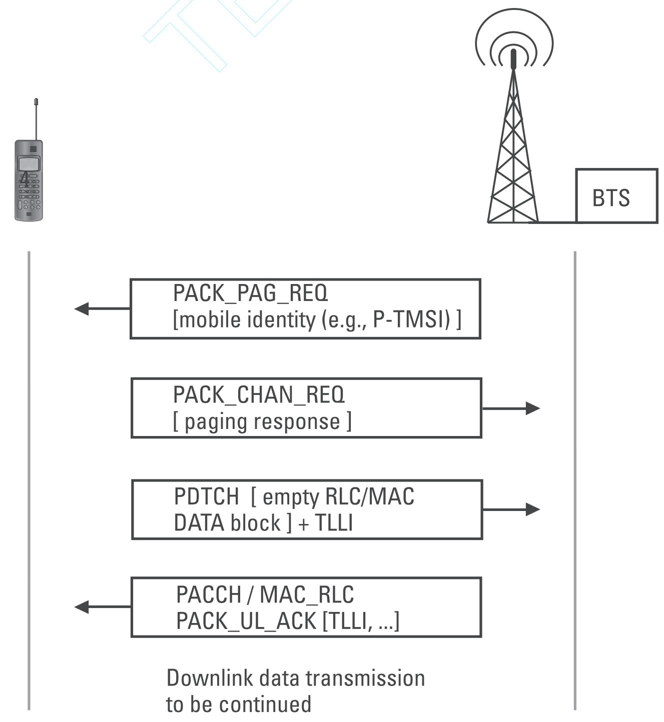


Figure 3.25 Procedure in page response (only if PCCCH is configured).

- Note that this possibility is not available on the RACH or in the CHAN_REQ message. Accordingly, the mobile station reacts there with a request for a single block two-phase packet access.

3.6.1.5 Access for GMM—Access for Cell Update

- If the mobile station has to carry out a procedure for GMM and is not in the GMM-READY state, *MM procedure* is given as the reason in the PACK_CHAN_REQ message.
- Only when the mobile station is in the GMM-READY state after choosing a new cell will *cell update* be given as the reason in the PACK_CHAN_REQ message.

3.6.1.6 Access for Single Block Without TBF Establishment

- If a mobile station intends to send a measurement report or a PACK_PAUSE message to the network, *single block without TBF establishment* is given as the reason in the PACK_CHAN_REQ message.
- The PCU then only allocated one uplink block to the mobile station.

3.7 Resource Allocation in GPRS

In Section 3.6 a number of methods were presented with which the mobile station can access the network. We have also already mentioned the fact that the network, or, to be more precise, the PCU, allocates resources to the mobile station as a reaction to the network access. This section will deal with the possibilities available for this resource allocation.

One must first differentiate between uplink and downlink resource allocation. Both directions must be regarded completely differently. We would first like to concentrate on the downlink direction.

3.7.1 Resource Allocation in the Downlink Direction

Whenever the PCU has data from higher levels in its output loop for transmission to a mobile station, a TBF in the downlink direction to this mobile station must be activated. As a rule, a PACK_DL_ASS message, as shown in Figure 3.26, is transmitted to the mobile station. This message contains at least the following information:

```
+---------+--------------------------------------------+-----------------------------------+
|BITMASK  |ID Name                                     |Comment or Value                   |
+---------+--------------------------------------------+-----------------------------------+
|20:18:35,720,119  1 -2 down  PCU-DOWN  DATA SYNC  RMAC-DOWN  PDAS                          |
|Downlink (PCU-DOWN)  DATA SYNC (= Data with Syncronisation)          |
|Data with Syncronisation                                                                   |
|00000001 |Mapping Flag                                |Data in RLC-MAC format             |
|00000000 |Spare                                       |0                                  |
|----100- |Frame type                                  |Data                               |
|---0---- |Code Field                                  |Coding Scheme 1                    |
|------0- |next expected uplink burst                  |4 normal bursts                    |
|-----0-- |Parity-even Bit of int. Block number        |0                                  |
|***b4*** |Power Control                               |15                                 |
|--111--- |Uplink State Flag                           |7                                  |
|GPRS RLC/MAC V2.4.0 Downlink (RMAC-DOWN)  PDAS (= Packet Downlink Assignment)              |
|Packet Downlink Assignment                                                                 |
|01------ |Payload Type                                |RLC/MAC Control Block, no opt.octets|
|--00---- |Relative Reserved Block Period              |TDMA frame N+13                    |
|----0--- |Supplementary/Polling                       |RRBP field is not valid            |
|-----000 |Uplink State Flag                           |0                                  |
|000010-- |Message Type                                |2                                  |
|Core Parameter                                                                             |
|------00 |Page Mode                                   |Normal Paging                      |
|0------- |Bit                                         |0                                  |
|-1------ |Bit                                         |1                                  |
|--0----- |Bit                                         |0                                  |
|**b32*** |TLLI                                        |c1c00a45                           |
|---0---- |Bit                                         |0                                  |
|----00-- |MAC Mode                                    |Dynamic Allocation                 |
|------0- |RLC Mode                                    |RLC Acknowledged Mode              |
|-------0 |Control Ack                                 |New Downlink TBF Established       |
|0------- |Timeslot 0 Allocation                       |Timeslot is not assigned           |
|-1------ |Timeslot 1 Allocation                       |Timeslot is assigned               |
|--1----- |Timeslot 2 Allocation                       |Timeslot is assigned               |
|---0---- |Timeslot 3 Allocation                       |Timeslot is not assigned           |
|----0--- |Timeslot 4 Allocation                       |Timeslot is not assigned           |
|-----0-- |Timeslot 5 Allocation                       |Timeslot is not assigned           |
|------0- |Timeslot 6 Allocation                       |Timeslot is not assigned           |
|-------0 |Timeslot 7 Allocation                       |Timeslot is not assigned           |
|1 Packet Timing Advance                                                                    |
|0------- |Bit                                         |0                                  |
|-1------ |Bit                                         |1                                  |
|--0000-- |Timing Advance Index                        |0                                  |
|***b3*** |Timing Advance Timeslot No                  |2                                  |
|1 (end of) Packet Timing Advance                                                           |
|-0------ |Bit                                         |0                                  |
|--1----- |Bit                                         |1                                  |
|2 Frequency Parameters                                                                     |
|---000-- |Training Seq Code                           |0                                  |
|------00 |Bit                                         |00                                 |
|**b10*** |ARFCN                                       |102                                |
|2 (end of) Frequency Parameters                                                            |
|--1----- |Bit                                         |1                                  |
|---00000 |Downlink TFI Assignment                     |0                                  |
|1------- |Bit                                         |1                                  |
|3                                                                                          |
|-0                                                    |alpha = 0.3                        |
|--                                                    |0                                  |
|--                                                    |1                                  |
|**                                                    |3                                  |
|--                                                    |1                                  |
|***b5*** |Gamma TN2                                   |3                                  |
|--0----- |Bit                                         |0                                  |
|---0---- |Bit                                         |0                                  |
|----0--- |Bit                                         |0                                  |
|-----0-- |Bit                                         |0                                  |
|------0- |Bit                                         |0                                  |
|3 (end of) Power Control Parameters                                                        |
|-------0 |Bit                                         |0                                  |
|0------- |Bit                                         |0                                  |
|-0101011 |Spare Padding                               |43                                 |
+----+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+
|HEX |0 |1 |2 |3 |4 |5 |6 |7 |8 |9 |A |B |C |D |E |F |
+----+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+
|0   |01|00|89|3f|40|08|58|38|01|48|a0|60|41|20|19|a0|
|10  |9a|38|c0|2b|2b|2b|2b|2b|2b|2b|2b|02|e0|ff|ff|ff|
|20  |ff|ff|ff|ff|ff|ff|ff|ff|  |  |  |  |  |  |  |  |
```

Allocation of TS 1 and TS 2 on ARFCN = 102

Which TFI shall the mobile station look for in the downlink data blocks on TS 1 and TS 2. In this case, TFI = 0 is allocated.

Repetition of the Message Content in hex.

Figure 3.26 Example of the allocation of downlink resources.

- The TFI of the allocated TBF;
- Information as to the time slots and the frequency (ARFCN) on which the mobile station should search for data packets with its TFI.

In other words, the mobile station cannot know in advance which radio blocks on the allocated time slots are actually determined for it itself. For this reason *all* the radio blocks on these time slots must be received and at least the TFI field evaluated before a decision can be made as to whether a data block has to be processed further or can be rejected.

The reader familiar with GSM will ask at this point (at the latest), what is the connection between a PDTCH and the resource allocation of the TBF/TFI? What physical basis does a PDTCH have? Before we go into detail on the uplink resource allocation, we would first like to examine this point further.

- A time slot on which PDCHs are allocated traverses the 52 multiframe with 12 radio blocks in 240 ms.
- Each of these 12 radio blocks can operate a different type of PDCH, including the PDTCH.
- No mobile station can determine in advance whether a downlink radio block is destined for it itself, another mobile station, or whether another PDCH will be transmitted as the PDTCH.
- Accordingly, a PDTCH is the possibly unperiodical sequence of radio blocks on a time slot in one direction, which contains data and is addressed to the same mobile station.
- A PDTCH can equally be defined as the possibly unperiodical sequence of radio blocks on a time slot in one direction with the same TFI.

Accordingly, the term PDTCH in GPRS relinquishes the greatest part of its physical characteristics in comparison with the TCH in GSM. It can neither be determined when a radio block belonging to a PDTCH is being transmitted, nor can the new term, TBF, be used as a clear indicator for a PDTCH.

3.7.2 Resource Allocation in the Uplink Direction

In the previous section, we discussed the case in which the network, or the PCU, wishes to transmit data blocks to the mobile station. The reverse is the case when a mobile station wishes to send data to the PCU and there is no uplink TBF active. In order to preempt the question, an uplink TBF can be active at the same time as a downlink TBF.

But back to the allocation of uplink resources. As soon as the RLC/MAC layer of the mobile station receives the request for data transmission from higher layers, the process is started as described in Section 3.6. An appropriate search for the allocation of uplink resources in the PCU is then carried out.

This section will deal with the allocation of resources. There are three different methods for the allocation of uplink resources, as Table 3.3 illustrates. Each of these three methods will be presented in the following sections.

3.7.2.1 The Fixed Allocation Method

In the fixed allocation method, the mobile station receives a resource allocation from the PCU in the form of a bitmap and a waiting time per time slot. In Figure 3.27 this method is demonstrated for one time slot. In this case, the mobile station would receive the bitmap 101 110 111 111 011 and the waiting time as given in the PACK_UL_ASS message.

Table 3.3
Support of the Different Resource Allocation Methods

	Mandatory for the Network	**Mandatory for the Mobile Station**
Fixed allocation	Yes*	Yes
Dynamic allocation	Yes*	Yes
Extended dynamic allocation	No	No, only multislot classes 22, 24, 25, and 27

*The network (PCU) must support either the fixed allocation method or the dynamic allocation method.

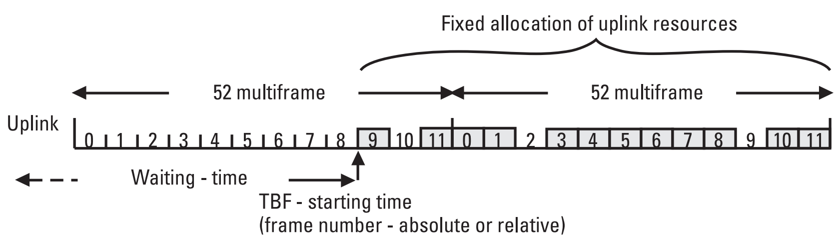


Figure 3.27 Allocation of radio blocks using the fixed allocation method.

It is to be noted that a mobile station is able to receive different bitmaps for the individual time slots to be allocated. Furthermore, a bitmap can, as illustrated, address individual blocks or entire block periods in consecutive 52 multiframes.

One question that is frequently asked in our courses is how long such a bitmap can be. This question cannot be answered precisely. Either the PACK_UL_ASS message receives the information per time slot as to how long the respective bitmap actually is or the bitmap lasts until the end of the message.

The mobile station can, of course, use one of the allocated blocks to request more resources. However, this is no longer possible in the same TBF if the network has placed the final allocation flag in the final PACK_UL_ASS message.

3.7.2.2 The Dynamic Allocation Method

While the fixed allocation method commits the network in advance with the allocation of resources for a given length of time, the method we are now going to examine chooses a much more dynamic way, which is why this method is called the dynamic allocation method.

The dynamic allocation method is, in fact, based on the USF, which always forms the first 3 bits in each RLC/MAC downlink block (see Figure 3.28). This applies equally to the RLC/MAC control and data blocks. In this way, eight different values can be distinguished via the USF.

The dynamic allocation of uplink resources functions as follows:

- First, the mobile station receives a series of time slots in a PACK_UL_ASS message and one USF per time slot. Figure 3.29 illustrates an example of this.
- As soon as the PACK_UL_ASS message is received, or after an indicated waiting time, the mobile station begins to monitor the downlink direction of the time slots allocated.
- If the mobile station recognizes its USF in a downlink block K on an allocated time slot N, the mobile station should use the uplink block ($k + 1$) on the same time slot for the transmission of an uplink RLC/MAC block.
- However, detecting its own USF on time slot n can also mean that not only the next one, but the next four uplink blocks ($k + 1$), ($k + 2$), ($k + 3$), and ($k + 4$) will be allocated to a mobile station. This is exactly what happens when the USF_GRANULARITY flag is set at 1 in the PACK_UL_ASS message.

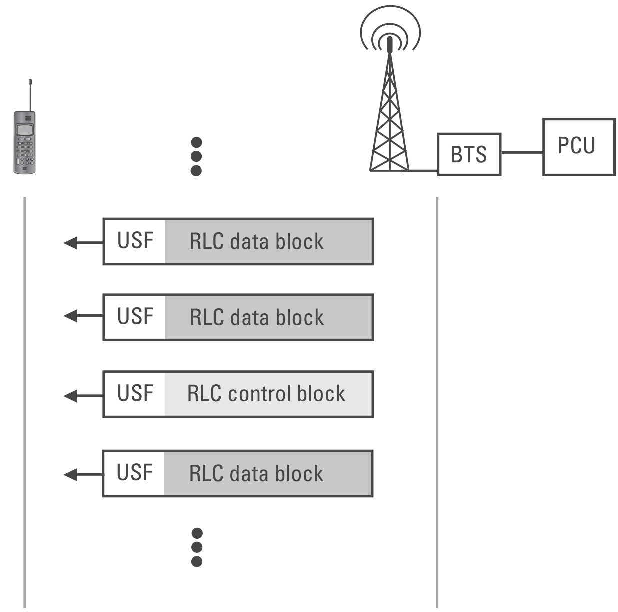


Figure 3.28 The USF always forms the first 3 bits in every downlink RLC/MAC block.

- If, however, USF_GRANULARITY is set at 0, the USF then only refers to the next uplink block (k + 1).

3.7.2.3 Further Peculiarities of the Uplink State Flag

The main function of the USF is the allocation of uplink resources to mobile stations as described above. However, the following factors and further functions must also be considered:

- *Two receivers per downlink block:* The downlink block k is, in all probability, not intended for the mobile station that is to receive the USF contained in the block. In other words, introduction of dynamic allocation means not only one, but two receivers for every downlink block. Imagine that the mobile station for which the USF is intended is a long way from the base station, while the mobile station for which the downlink block itself is intended is very close to the base station. With active downlink performance regulation, the risk of mobile stations not being able to "hear" their USF thus increases. The standard (GSM 05.08), however, clearly prescribes that in such a case, performance regulation should ensure that both mobile stations can receive the block.

```
+---------+-----------------------------------------+-----------------------------------+
|BITMASK  |ID Name                                  |Comment or Value                   |
+---------+-----------------------------------------+-----------------------------------+
|16:18:37,059,959  1 -2 down  PCU-DOWN  DATA SYNC  RMAC-DOWN  PUAS                      |
|GPRS PCU V2.0 Downlink (PCU-DOWN)  DATA SYNC                                           |
|Data with Syncronisation                                                               |
|00000001 |Mapping Flag                             |Data in RLC-MAC format             |
|00000000 |Spare                                    |0                                  |
|----100- |Frame type                               |Data                               |
|---0---- |Code Field                               |Coding Scheme 1                    |
|------0- |next expected uplink burst               |4 normal bursts                    |
|-----0-- |Parity-even Bit of int. Block number     |0                                  |
|***b4*** |Power Control                            |15                                 |
|--111--- |Uplink State Flag                        |7                                  |
|GPRS RLC/MAC V6.4.0 Downlink (RMAC-DOWN)  PUAS (= Packet Uplink Assignment)            |
|Packet Uplink Assignment                                                               |
|01------ |Payload Type                             |RLC/MAC Control Block, no opt.octets|
|--00---- |Relative Reserved Block Period           |TDMA frame N+13                    |
|----0--- |Supplementary/Polling                    |RRBP field is not valid            |
|-----000 |Uplink State Flag                        |0                                  |
|001010-- |Message Type                             |10                                 |
|Core Parameter                                                                         |
|------00 |Page Mode                                |Normal Paging                      |
|0------- |Bit                                      |0                                  |
|-0------ |Bit                                      |0                                  |
|1 Global TFI                                                                           |
|--1----- |Bit                                      |1                                  |
|---00000 |Downlink TFI                             |0                                  |
|1 (end of) Global TFI                                                                  |
|0------- |Bit                                      |0                                  |
|-01----- |Channel Coding Command                   |CS-2                               |
|---1---- |TLLI Block Channel Coding                |Use Channel Coding Command value   |
|2 Packet Timing Advance                                                                |
|----0--- |Bit                                      |0                                  |
|-----1-- |Bit                                      |1                                  |
|***b4*** |Timing Advance Index                     |0                                  |
|--010--- |Timing Advance Timeslot No               |2                                  |
|2 (end of) Packet Timing Advance                                                       |
|-----0-- |Bit                                      |0                                  |
|------01 |Bit                                      |01                                 |
|3 Dynamic Alloc Struct                                                                 |
|0------- |Extended Dynamic Allocation              |Dynamic Allocation                 |
|-0------ |Bit                                      |0                                  |
|--0----- |USF Granularity                          |Transmit 1 RLC/MAC block           |
|---1---- |Bit                                      |1                                  |
|***b5*** |Uplink TFI assignment                    |0                                  |
|-0------ |Bit                                      |0                                  |
|--0----- |Bit                                      |0                                  |
|---1---- |Bit                                      |1                                  |
|----0011 |Alpha                                    |alpha = 0.3                        |
|0------- |Bit                                      |0                                  |
|-0------ |Bit                                      |0                                  |
|--1----- |Bit                                      |1                                  |
|---100-- |USF TN2                                  |4    Allocation of:                |
|***b5*** |GAMMA TN2                                |3    => USF 4 on TS 2,             |
|---1---- |Bit                                      |1    => USF 0 on TS 3,             |
|---000-- |USF TN3                                  |0    => USF 7 on TS 4              |
|***b5*** |GAMMA TN3                                |3                                  |
|----1--- |Bit                                      |1                                  |
|---111-- |USF TN4                                  |7                                  |
|***b5*** |GAMMA TN4                                |3                                  |
|-----0-- |Bit                                      |0                                  |
|------0- |Bit                                      |0                                  |
|-------0 |Bit                                      |0                                  |
|3 (end of) Dynamic Alloc Struct                                                        |
+---------+-----------------------------------------+-----------------------------------+
```

Figure 3.29 Example of PACK_UL_ASS message in dynamic allocation with allocation to time slots 2, 3, and 4.

- *How many mobile stations can be identified per time slot via the USF?* The straightforward answer to this question would at first seem to be eight. However, the following limiting factors must also be taken into account:

    1. If the fixed allocation method is also being used on this time slot, a USF value must be reserved for those uplink blocks that are kept for mobile stations with fixed allocation. This USF value is not predetermined, but is described as USF unused value.

Accordingly, mobile stations with dynamic allocation must be prevented from regarding the next uplink block as its own. The USF unused value must also be reserved without the fixed allocation method, because the PCU must reserve some uplink blocks for the uplink PACCH of downlink TBFs on this time slot. We will go into more detail on the operation on the PACCH later.

2. If the PCCCH is operated on one or more time slots, a further USF value, the USF free value, must be reserved in order to identify PRACH blocks. In contrast to the USF unused value, the USF free value is set in the standard as 111bin (i.e., 710). Therefore, a maximum of six or seven different mobile stations per time slot can be addressed per time slot via dynamic allocation, depending on whether the PCCCH is also being operated on a time slot. Figure 3.30 shows an example of this.

3.7.2.4 The Extended Dynamic Allocation Method

The extended dynamic allocation method is simply an extension of the dynamic way of allocating more resources to a mobile station by USF allocation. In addition to dynamic allocation, the following rules apply:

- If a mobile station detects its USF in downlink block *K* on time slot *N*, this mobile station should use the uplink block (k + 1) on the same time slot *N* and all time slots with higher numbers that have been allocated.
- The USF_GRANULARITY flag also exists in extended dynamic allocation. Accordingly, the above-mentioned rule with USF_GRANULARITY flag = 1 can also refer to the next four uplink blocks.
- If a mobile station has received its USF in downlink block *K* on time slot *N*, this mobile station no longer needs to receive the USFs in the downlink blocks *K* and (k + 1) on the time slots with higher numbers that belong to the allocation.

Figure 3.31 shows a typical example. On the horizontal axis are the 12 radio blocks of a 52 multiframe; on the vertical axis are the time slots allocated (here, TS 0, 2, 3, 5, and 7). Thus, in extended dynamic allocation, time slots that only offer fixed or dynamic allocation or are used for other extended dynamic allocations can be jumped over. The example uses a USF_GRANULARITY flag = 0.

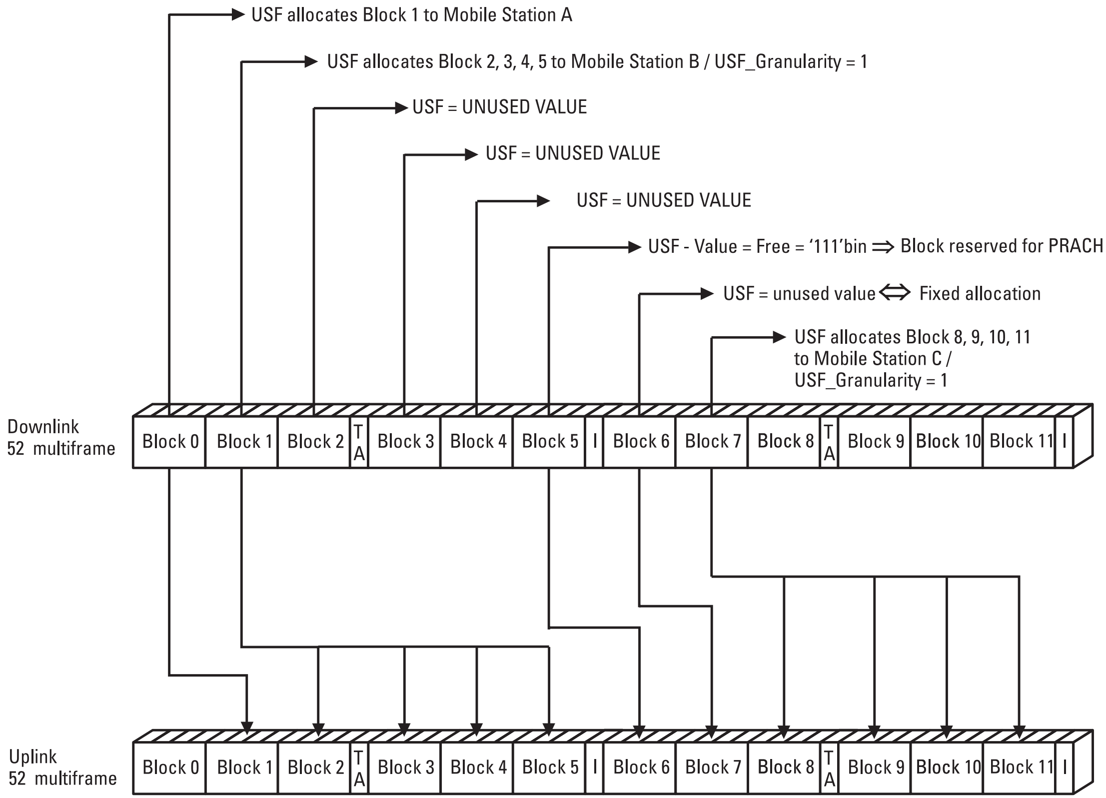


Figure 3.30 Example of the operation of the USF on a time slot with PCCCH and fixed allocation.

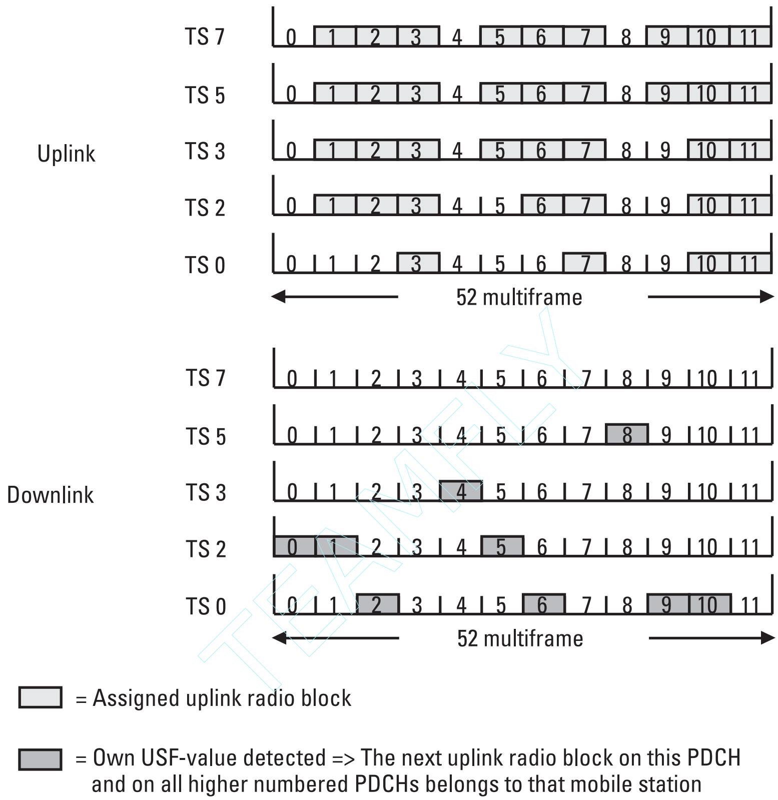


Figure 3.31 Example of the procedures in extended dynamic allocation.

- In radio block 0/time slot 2 in the downlink direction, the uplink radio block 1 is allocated on the same time slot 2 and all time slots with higher numbers that also belong to this allocation. Accordingly, the mobile station uses the uplink block 1 on time slots 2, 3, 5. and 7.
- Via the downlink radio block 1, as many resources can be allocated as in downlink block 0, maybe more, or maybe none at all. In our example, exactly the same amount of resources is allocated. Accordingly, the mobile station uses the uplink block 3 on time slots 0, 2, 3, 5, and 7.
- In downlink radio block 2, more resources are allocated because the mobile station already receives its USF on time slot 0. Accordingly, the mobile station uses the uplink block 1 on time slots 2, 3, 5, and 7.

- In downlink radio block 3, no resources at all are allocated because the network (the PCU) wishes to reduce the number of time slot to be used. Note that according to the rules stated above, the mobile station only reads downlink radio block 3 on time slot 0.
- After this "free" downlink radio block 3, the mobile station then reads the downlink radio block 4 on all allocated time slots again, at least until it has found its USF in time slot 3. Accordingly, the mobile station transmits uplink radio block 5 on time slots 3, 5, and 7.

We hope that these explanations are sufficient for understanding extended dynamic allocation.

3.8 The Operation of the PACCH in Uplink and Downlink Direction

The bidirectional PACCH has already been introduced in Section 3.2.3. Every unidirectional PDTCH is simultaneously allocated with this bidirectional PACCH. The problem is, however, that the PACCH has no resources of its own.

The operation of the PACCH will now be presented and explained.

3.8.1 The Operation of the PACCH for Downlink TBFs

In the case of a downlink TBF the mobile station listens to all allocated time slots to determine if the radio blocks transmitted by the PCU contain the TFI of its downlink TBF. Some of these radio blocks could also be control messages on the PACCH. The mobile station identifies those control messages that are addressed to it either by (1) the corresponding downlink RLC/MAC control block containing its own TFI in the optional RLC header, or (2) if there is no RLC header, investigating the control message itself and finding out if this message contains either the TFI or the TLLI of the mobile station.

This method is easy to follow. In the opposite direction, however, it becomes more difficult because, as we know, a mobile station has no uplink resources when a downlink TBF is active. Accordingly, the network must periodically or as required provide single uplink radio blocks for the transmission of control messages on the uplink PACCH during a downlink TBF. This ensues via the 2-bit *relative reserved block period* (RRBP) field in the

MAC header of downlink RLC/MAC blocks. This procedure is shown in Figure 3.32. Here an uplink radio block is referenced via the RRBP field at a set temporal distance from the current frame (TDMA frame). Due to every RRBP coding being occupied, there is, in addition to the RRBP field, the supplementary/polling (S/P) flag, which states whether or not the information in the RRBP field is valid.

3.8.2 The Operation of the PACCH for Uplink TBFs

With an uplink TBF, the mobile station can use each of the allocated uplink radio blocks for transmitting a control message on the PACCH. It could be said that in the case of uplink TNFs, the PACCH steals the resources that are actually intended for the PDTCH.

The situation becomes more interesting in the opposite direction, with an uplink TBF in the downlink direction. How can the PCU send the mobile station control messages when there is an uplink TBF? In order to answer this question, the three methods of allocating uplink resources must be differentiated:

1. In fixed allocation, the PCU informs the mobile station in the PACK_UL_ASS message which time slot is used for receiving downlink PACCH messages and must therefore be received constantly by the mobile station.
2. In dynamic allocation, the mobile station must receive the downlink direction of all allocated time slots anyway in order to decode the USF. Accordingly, there is no limit for the PCU as to which time slot can be used for transmitting downlink PACCH messages.

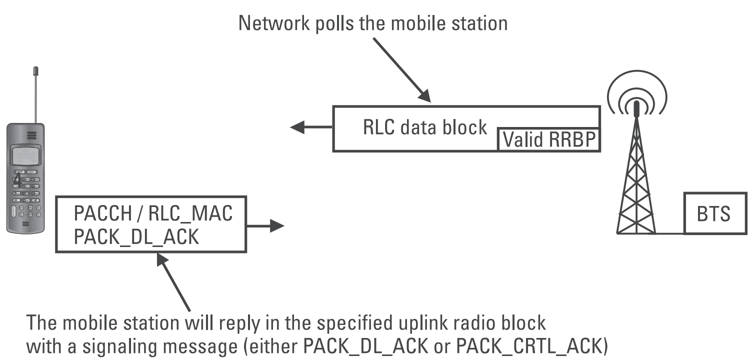


Figure 3.32 Allocation of resources for the uplink PACCH.

3. In extended dynamic allocation, the mobile station will very often not be able to receive the downlink direction of the higher time slot numbers. This situation has actually been accounted for in the definition of this method (see Section 3.7.2.4). Accordingly, the PCU may only transmit downlink PACCH messages on the lowest numbered time slot in this allocation (in our example from Section 3.7.2.4, Figure 3.31, this would be TS 0).

3.9 The Termination of a TBF—Release of Resources

In Chapter 2, we mentioned that a packet switched system requires immediate allocation and release of resources. So far we have concentrated on the allocation of these resources. The question remains of how these resources can be released again just as quickly and without excessive signaling. The termination of a TBF will be treated separately for uplink and downlink.

3.9.1 The Termination of a Downlink TBF

The termination of a downlink TBF is a simple process. As Figure 3.33 illustrates, the final block indicator (FBI) bit in the RLC header is set at 1 in the last data block transmitted by the PCU, thus showing that this is the

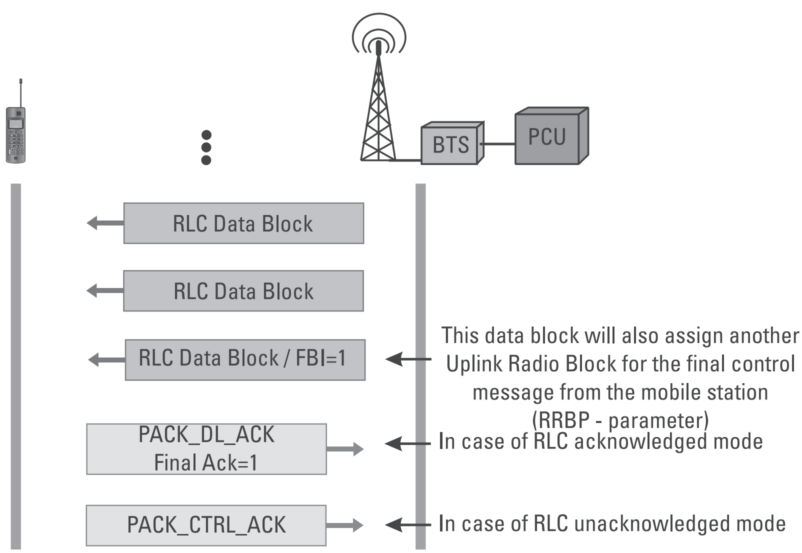


Figure 3.33 The termination of a downlink TBF via the FBI bit.

last block. The further procedure in the termination of the TBF depends on the RLC/MAC operation mode:

- *Acknowledged operation mode:* Here, perfect reception of all the data blocks must be confirmed by the mobile station via the PACK_DL_ACK, or the message to retransmit data blocks received with defects must be sent, also via the PACK_DL_ACK.
- *Unacknowledged operation mode:* Independently of whether there are any transmission errors, the mobile station sends a PACK_CTRL_ACK message to the PCU. Please note that this message can also be formatted as four access bursts (see Section 3.3.2).

3.9.2 The Termination of an Uplink TBF

The termination of uplink TBFs is usually initiated by the mobile station. In order to do this, the mobile station uses what is known as the countdown procedure. The countdown procedure is based on the 4-bit countdown value (CV), which is a parameter in the MAC header of uplink data blocks. This CV is counted down to 0 by the mobile station, which means that the last data block transmitted by the mobile station has a CV of 0, as Figure 3.34 shows.

One may now ask when the countdown procedure is started by the mobile station. In fact, this depends on the BS_CV_MAX parameter, which is sent out in the SYS_INFO messages (e.g., SYS_INFO13; see Figure 3.5). The maximum value of BS_CV_MAX is 15_{10}, but any value as low as 0 can be set by the network operator.

During an uplink TBF, the mobile station calculates the countdown value for every data block to be transmitted using the following formula:

$$\begin{array}{c}\text{Countdown procedure starts}\\ \text{BS\_CV\_MAX}\end{array} \Leftrightarrow \frac{\text{(total no. of blocks in TBF-1)} - \text{BSN'}}{\text{Number of allocated time slots}}$$

- Here, BSN' is not the block sequence number sent with every block; this BSN' ranges from 0 to (total number of blocks in TBF-1).
- As long as the result of this formula is greater than BS_CV_MAX, the countdown value in the data block to be sent will be left at 15_{10}.

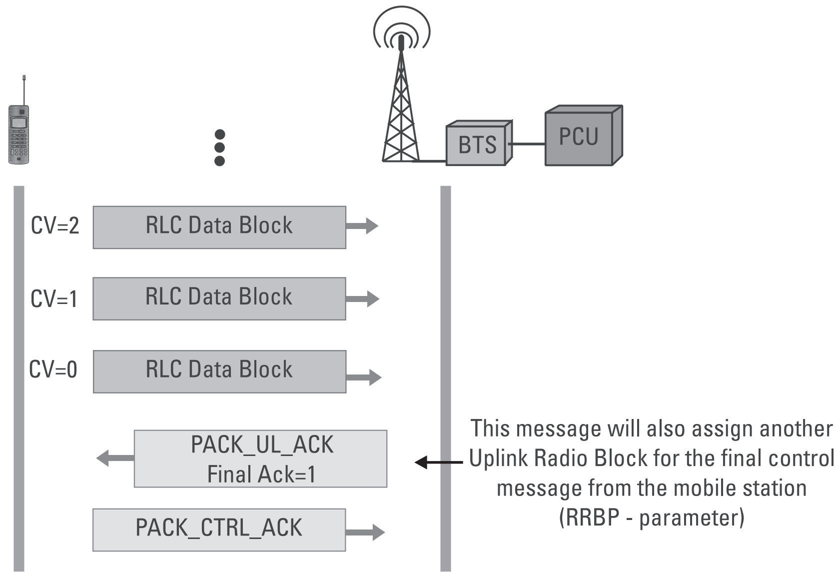


Figure 3.34 Simplified presentation of the countdown procedure.

- Only when the condition mentioned above is fulfilled is the CV decremented according to the above formula until the final block is sent with a CV of 0.

The following points are also important in understanding the countdown procedure:

- By using the countdown procedure, the PCU receives in advance the information that a mobile station will shortly no longer require any further resources. In accordance with this advance warning, the PCU can then dispose of the available uplink resources.
- The formula stated above is used for all blocks of a TBF (i.e., even after the countdown procedure has already started).
- For this reason, a mobile station cannot cancel a countdown procedure once it has started. This is especially important when additional data is placed in the mobile station's RLC/MAC output loop after a countdown procedure has begun. In such a case, the mobile station must first finish a current TBF and then set about activating a new one.

- The only exceptions to this rule are:
  - Changing the coding scheme after the countdown procedure has already started. In such a case, the mobile station must apply the formula again, which can also lead to temporary interruption of the countdown procedure;
  - The retransmission of data blocks due to transmission errors.

Figures 3.35 and 3.36 illustrate examples of the function of the countdown procedure with one and four time slots, respectively. For clarity, the other key data of these TBFs are also given.

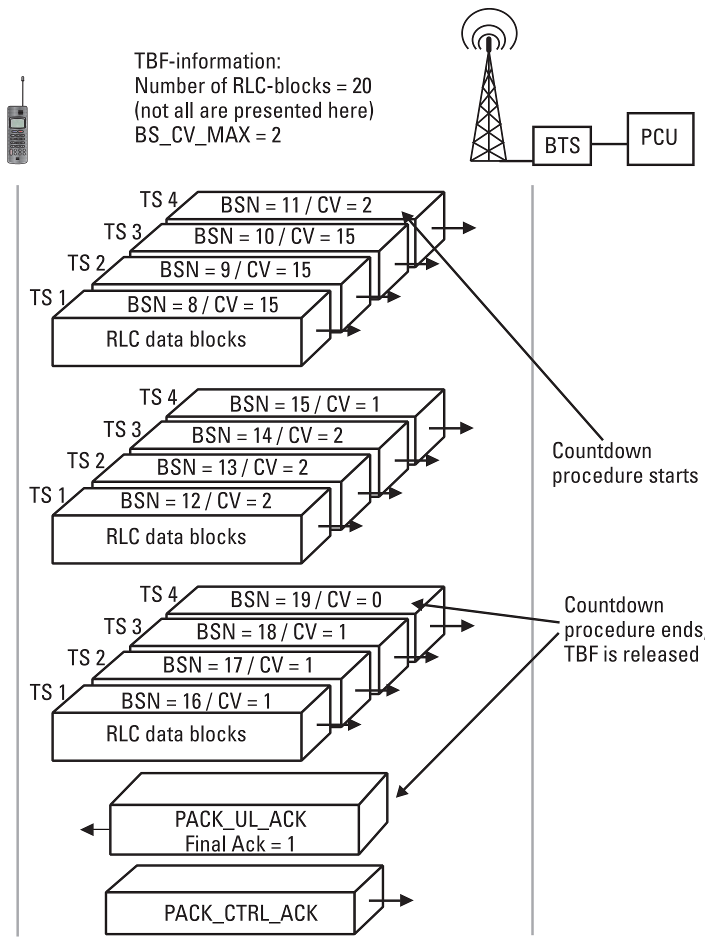


Figure 3.35 Example of the countdown procedure on one time slot.

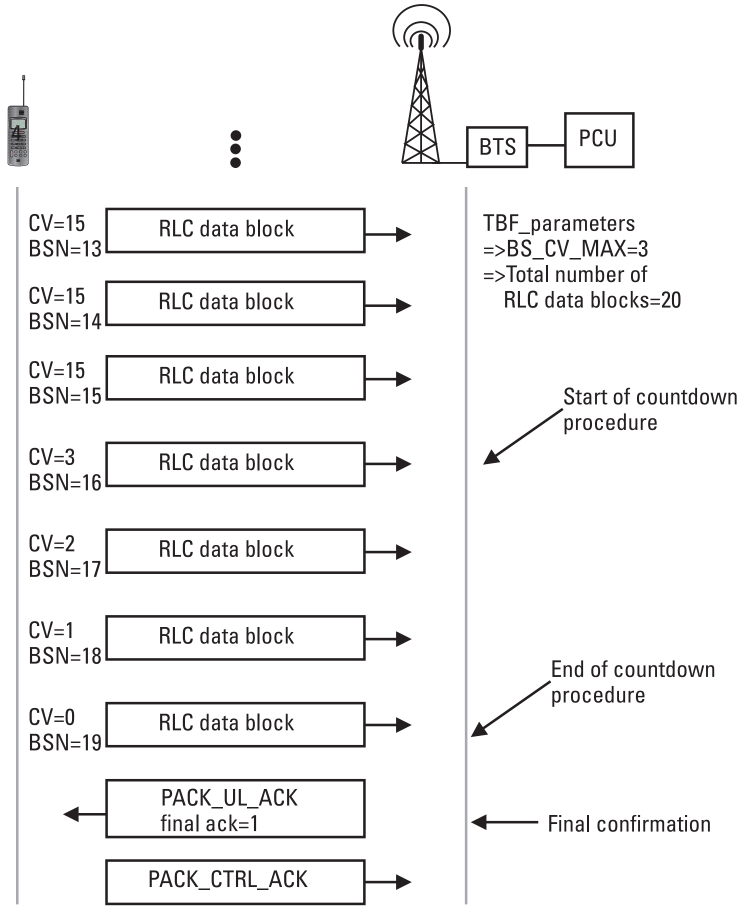


Figure 3.36 Example of the countdown procedure on four time slots.

4

QoS, Mobility, and Session Management in GPRS

4.1 GPRS Mobility Management

The capability of a mobile station to move freely inside and outside its own network is described in GSM as roaming. GSM expressly enables a subscriber to move not only in his own network, the home PLMN (H-PLMN), but also in what are called visited PLMNs (V-PLMN). The function behind roaming is mobility management.

GPRS should also allow unlimited roaming. A subscriber should then be able to access external packet data, such as the Internet, everywhere both inside and outside his H-PLMN using his GPRS mobile station.

One of the problems that needs to be solved here is the fact that in GSM, the MM functions are basically handled between the VLR and the mobile station. The VLR, however, has no function in GPRS, and the SGSN must also assume the GMM functions. Since the introduction of GPRS, there have been two databases in the network that take care of mobility management. Both of these databases (VLR and SGSN) are also in contact with the HLR in order to exchange information about the individual subscribers (Figure 4.1).

4.1.1 New Identification Parameters for GMM

In order to eliminate inconsistencies, especially between VLR and SGSN, new and additional identification parameters have been defined for GPRS mobility management.

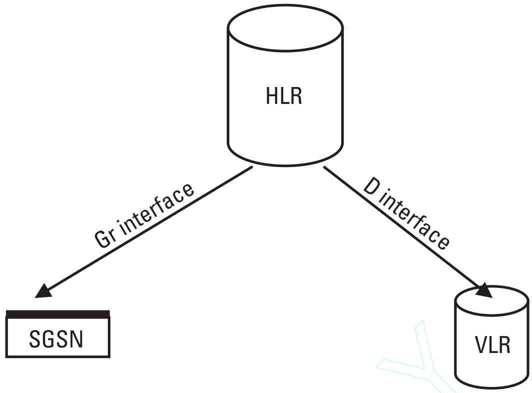


Figure 4.1 The HLR as a mutual basis for SGSN (GPRS) and VLR (GSM).

In addition to the temporary mobile subscriber identity (TMSI), there has also been the packet-temporary mobile subscriber identity (P-TMSI) and the routing area (RA) in addition to the location area (LA). As illustrated in Figure 4.2, the P-TMSI differs from the TMSI as follows:

- Bits 30 and 31 in a P-TMSI are always set at 11_2, while the same bits in a TMSI are set at 00_2 01_2 or 10_2.
- The P-TMSI is allocated by the SGSN, while the TMSI is allocated by the VLR.

A further parameter, the temporary logical link identifier, does not actually have anything to do with GPRS mobility management; however,

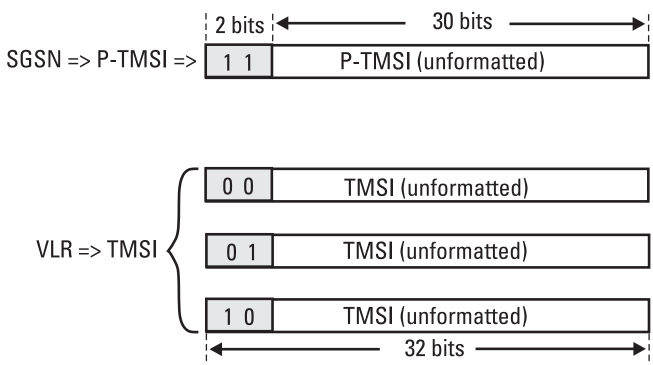


Figure 4.2 P-TMSI and TMSI.

we are introducing it here because the local TLLI is an exact copy of the P-TMSI, as shown in Figure 4.3.

Note, however, that there are four different TLLIs that are used by the mobile station or the SSGN depending on the situation or the desired procedure. Also note here that the TLLI is not a GMM parameter, but identifies the logical connection between mobile station and SGSN. For this reason, the TLLI selected in an SGSN field must be absolutely clear.

- *Local TLLI:* The local TLLI is the P-TMSI. The P-TMSI is always used by the mobile station as a TLLI when the mobile station accesses the network in the routing area in which this P-TMSI was allocated.
- *Foreign TLLI:* The foreign TLLI is always used by the mobile station for network access in the routing area update. This happens in order to minimize the probability of several identical TLLIs per SGSN field.
- *Random TLLI:* The random TLLI is always used by the mobile station for network access when there is no P-TMSI currently saved in the mobile station (see Section 4.1.2.1). Furthermore, a random TLLI is sued by the mobile station for first access in an anonymous PDP context activation, which is, however, only defined in GPRS release 97 and 98.

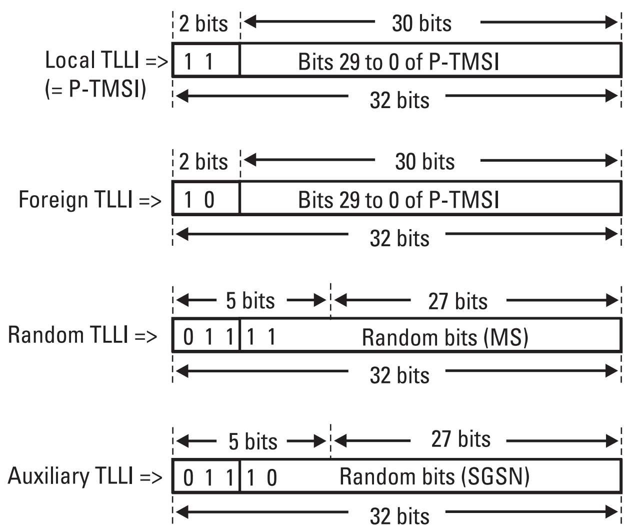


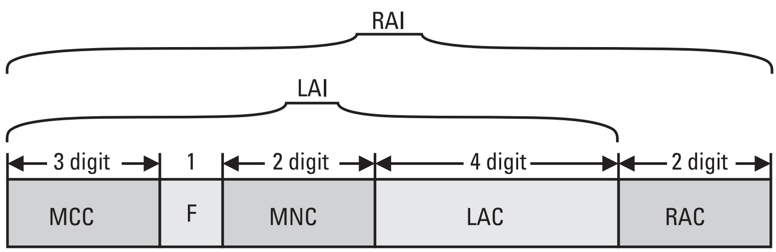
Figure 4.3 The different types of TLLIs.

- *Auxiliary TLLI:* The auxiliary TLLI is only defined in GPRS release 97 and 98 for the anonymous PDP context activation, in which a mobile station does not identify itself to the network. In such a case, the mobile station would use a random TLLI, upon which the SGSN would allocate an auxiliary TLLI in order to reduce the probability of several identical random TLLIs. From release 99 onwards, there is no longer an anonymous PDP context activation in GPRS, which is why the auxiliary TLLI is (currently) not being used.

On the network side, the RA was introduced in addition to the familiar location area. The definition of the RA is based on the LA. According to this, an RA is identified by the routing area identifier (RAI), which, in turn, is made up of the location area identifier (LAI) and the 8-bit routing area code (RAC). Figure 4.4 illustrates the RAI. An RA is thus always a part of an LA. In other words, one or more RAs (max. 256) form an LA. All the BTSs in an LA that do not support GPRS are collated in what is known as the null routing area. This is shown in Figure 4.5.

4.1.2 New Procedures in GMM

Important procedures such as IMSI attach or IMSI detach, location area updating, and authentication should be familiar from GSM. Analogously to these, GPRS performs GPRS attach, GPRS detach, and routing area updating procedures with an authentication option.



MCC = mobile country code
MNC = mobile network code
LAI = location area identification
LAC = location area code
RAC = routing area code
RAI = routing area identification

Figure 4.4 The RAI.

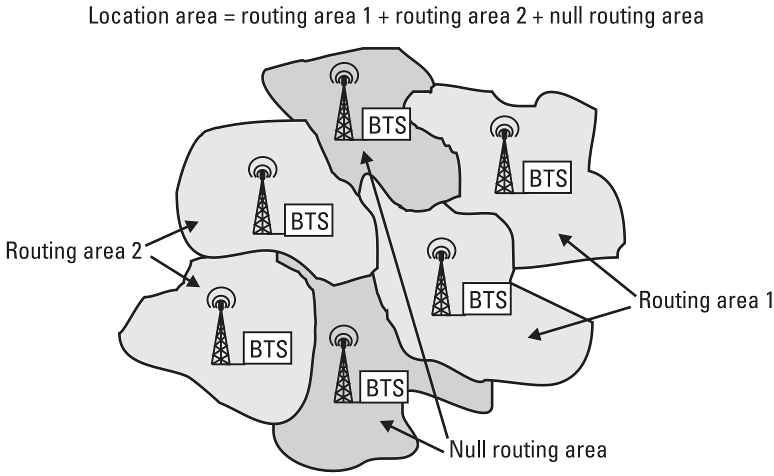


Figure 4.5 The connections between location area, routing area, and null routing area.

What is really new with GPRS is the cell update procedure, which is not defined in GSM. We shall now examine all of these procedures in detail.

4.1.2.1 Authentication and Parameter Saving in GPRS

The introduction of the new identification parameters does not basically alter any of the mechanisms for authentication in GPRS. Two special points should be noted, however GPRS combines authentication with the activation of ciphering.

- Accordingly, there are not two different messages for authentication and ciphering, as known from GSM, but only one, combined AUTH_CIPH_REQ messages.
- It is also possible, depending on the NOM, for a subscriber to be authenticated twice, once by the VLR and once by the SGSN.

Double authentication always occurs when a class A or B mobile station is situated in an SGSN field in NOM II or NOM III and wishes to use both GSM and GPRS. Accordingly, the VLR will then challenge the mobile station when registering for the network with an SRES parameter, while the SGSN, which works independently, will, in all probability, challenge the mobile station with a second RAND parameter. In any case, the SRES parameter is also determined via the A8 algorithm and the private key, Ki, in the SIM card in GPRS, as described in Section 1.1.2.1.

The only way to avoid this double authentication is to operate the network in NOM I. Let us go back to double authentication in NOM II and NOM III. The problem that the mobile station encounters with double authentication is that two SRES parameters and two ciphering keys (Kc) must now be reserved in the mobile station. This parameter, as well as the CKSN parameter, which is related to Kc, should actually be stored on the SIM card and not in the mobile device. However, GSM phase I and phase II SIM cards cannot save two different SRES or Kc values. So what must be done?

There are clearly several different ways to solve the problem. At first glance the best solution is simply to replace the old SIM cards. In fact, however, a completely different way is used because the aforementioned method will inevitably entail costs and logistical difficulties.

The solution is that the GPRS-ME must be able to save these parameters. If there is a phase I or II SIM card in the GPRS-ME, the ME must save the new parameters that have arrived with GPRS. This not only affects the previously mentioned GPRS-SRES, GPRS-Kc, and GPRS-CKSN, but also the P-TMSI or the RAC. Accordingly, such ME must check whether the same SIM card is in the device every time it is switched on. If this is not the case, the GPRS parameters saved in the device must be deleted. The mobile station, which is now without a P-TSMI, would then use a random TLLI to log on to the network.

4.1.2.2 The GPRS Attach Procedure

When a mobile station that supports GPRS is switched on, the cell selection process as described in Section 1.5.1 must be performed first. If the BTS selected supports the GPRS, the SYS_INFO13 (Figure 3.5) is also transmitted on the time slot 0 of the BCCH carrier. The mobile station then receives all relevant details on GPRS in this cell via the SYS_INFO13, or via the PBCCH.

The mobile station must then register for the GPRS service in the network. Registration means that a GMM context must be established between the mobile station and the network (especially SGSN). For this purpose, the mobile station carries out the GPRS attach scenario. Table 4.1 lists the individual components of a GMM context.

In GPRS attach, there must also be a differentiation depending on whether it is carried out via the CCCH (Figure 4.6) or, if it is configured, via the PCCCH (Figure 4.7). Both possibilities will be illustrated. The whole message procedure between mobile station and SGSN is presented. Furthermore, in Figure 4.8, we also present a GPRS attach scenario from

Table 4.1
GMM Context

1. 5 × (RAND, SRES, Kc, CKSN)	There are up to five authentication triplets, each consisting of RAND, SRES, and Kc, in the SGSN. The SGSN also supplies a CKSN in addition to each triplet.
2. DRX parameters	For discontinuous reception. The DRX settings must be known in the mobile station and the SGSN.
3. T3314 (default = 44 sec)	The ready timer (see Section 4.1.2.4 for explanations on this)
4. P-TMSI and P-TMSI signature	For identifying the mobile station or a given GMM context:
5. Position of mobile station	a) In the SGSN, in the routing area—on the BTS precisely (only in ready state) b) In the HLR, only the SGSN is known

a tracefile that is in operation, but only on the Gb interface between PCU and SGSN.

The following are some additional items of note:

- The messages shown differentiate between data content and signaling. The darker messages represent data for the respective interface.
- Each GMM message clearly requires a TBF to be set up.
- The countdown procedure (uplink TBF release) is indicated in all the scenarios as shown in Figure 4.9.
- A downlink TBF release is indicated as shown in Figure 4.10.
- The restart of the ready timer is indicated when "ready timer" appears to the left of the mobile station or to the right of the SGSN.
- Figure 4.7 illustrates that GPRS attach on the PCCCH (1) is shown when the timing advance control can be performed using polling and access bursts. Please regard this illustration as an example only and thus the PACK_CTRL_ACK message (four access bursts) as optional.

4.1.2.3 The Routing Area Updating Procedure

In Section 4.1.1, we described the routing area, a new introduction with GPRS. If a mobile station moves from one routing area to the next during operation, this mobile station performs a routing area updating procedure.

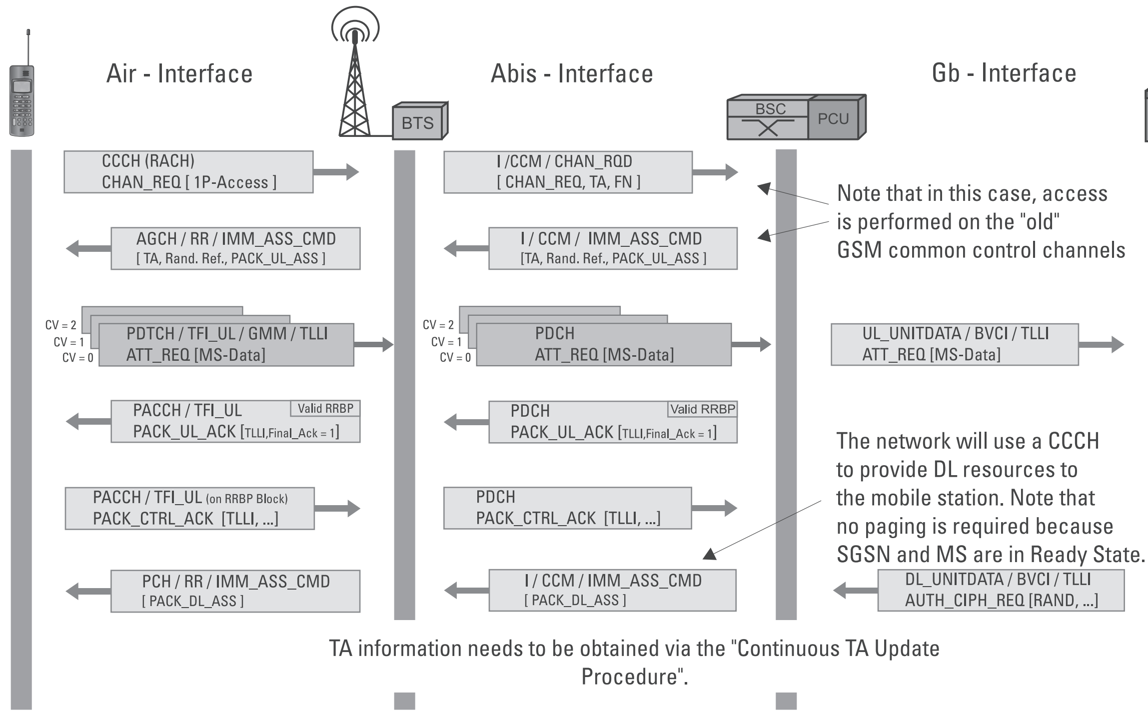


Figure 4.6 The GPRS attach via the CCCH.

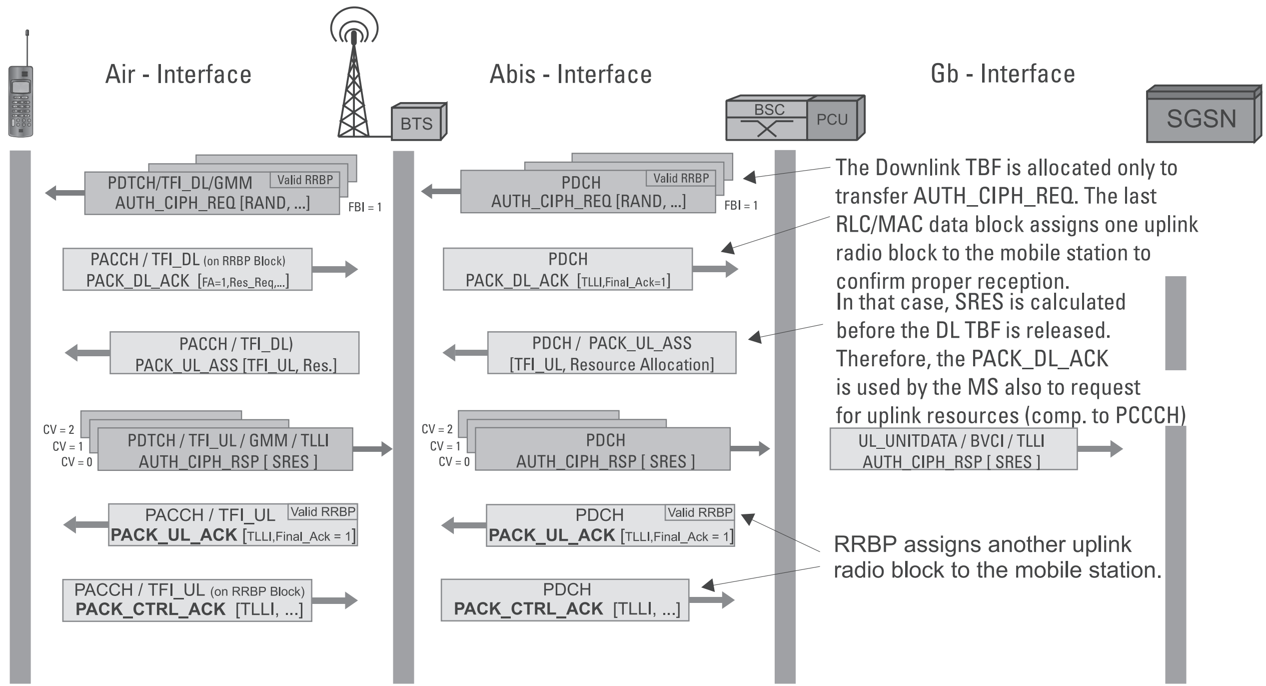


Figure 4.6 (continued).

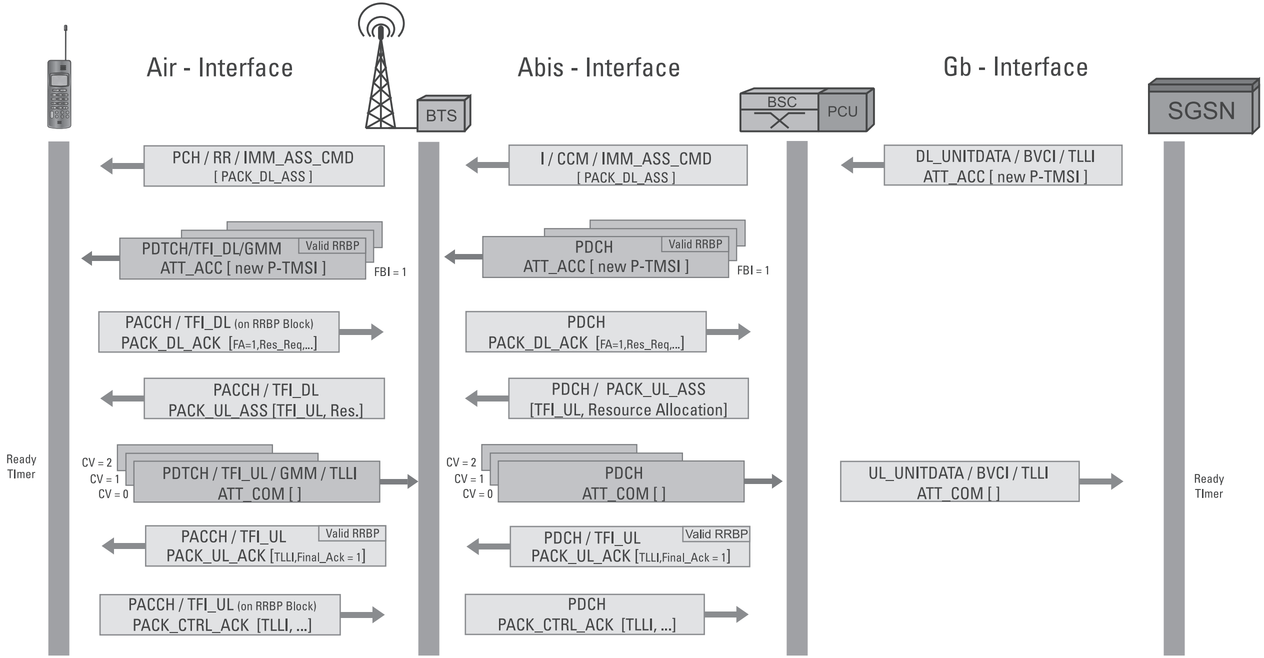


Figure 4.6 (continued).

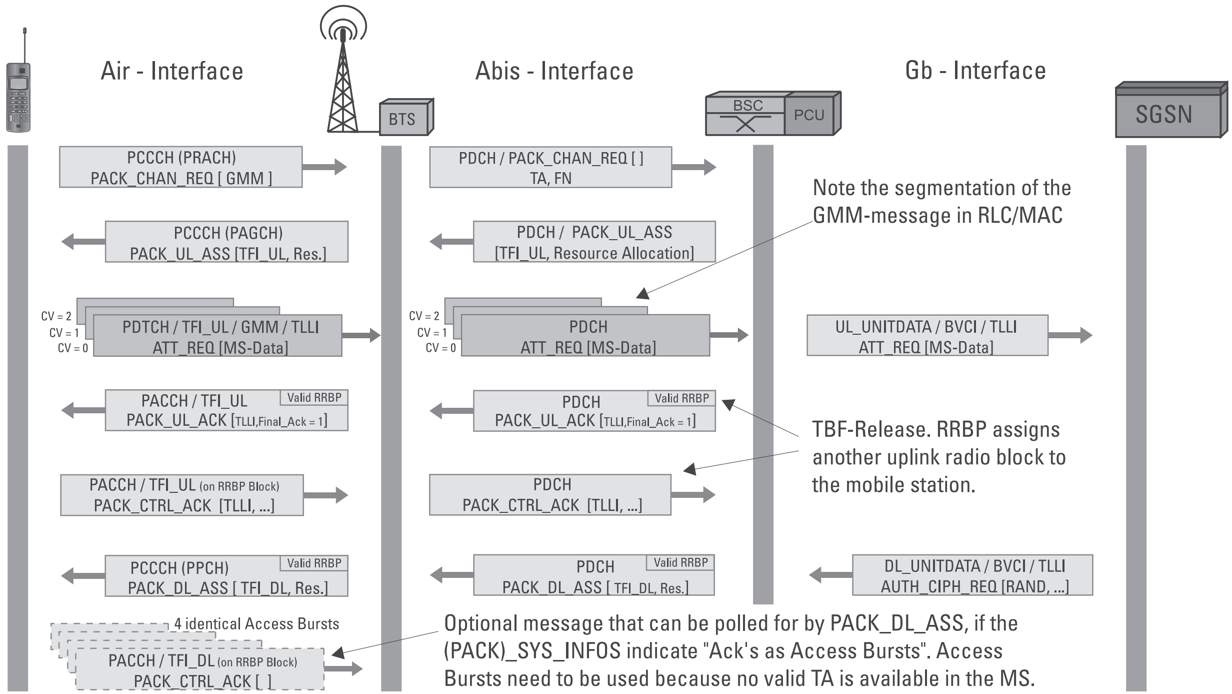


Figure 4.7 The GPRS attach via the PCCCH.

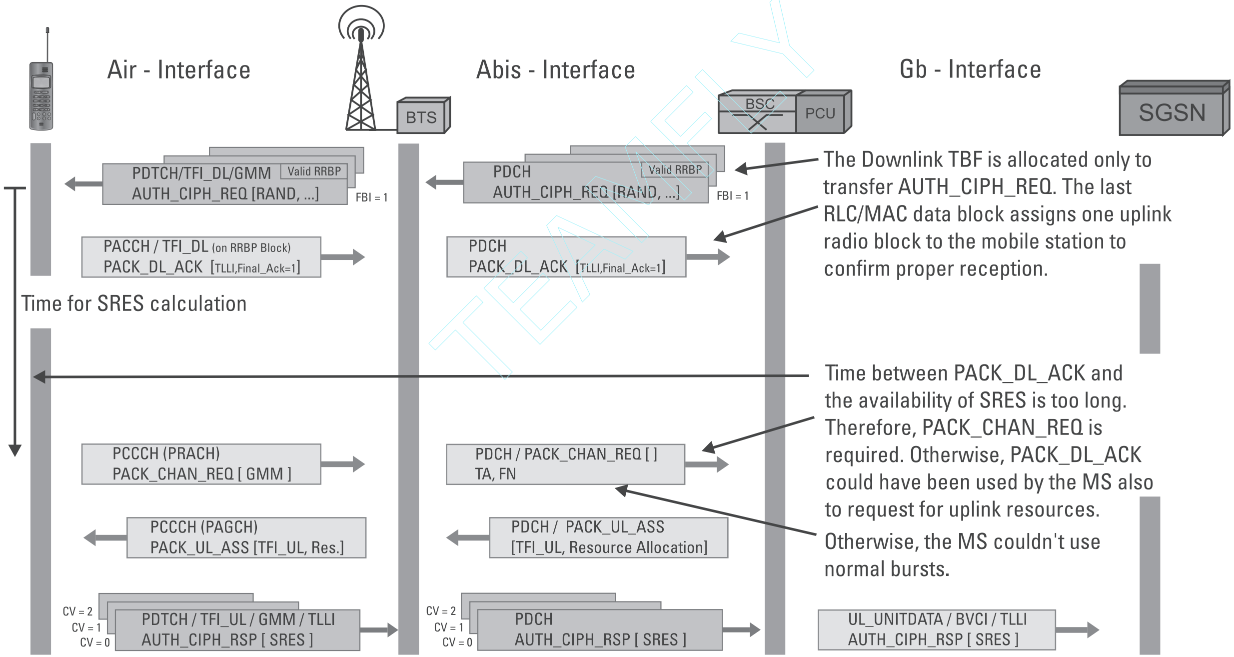


Figure 4.7 (continued).

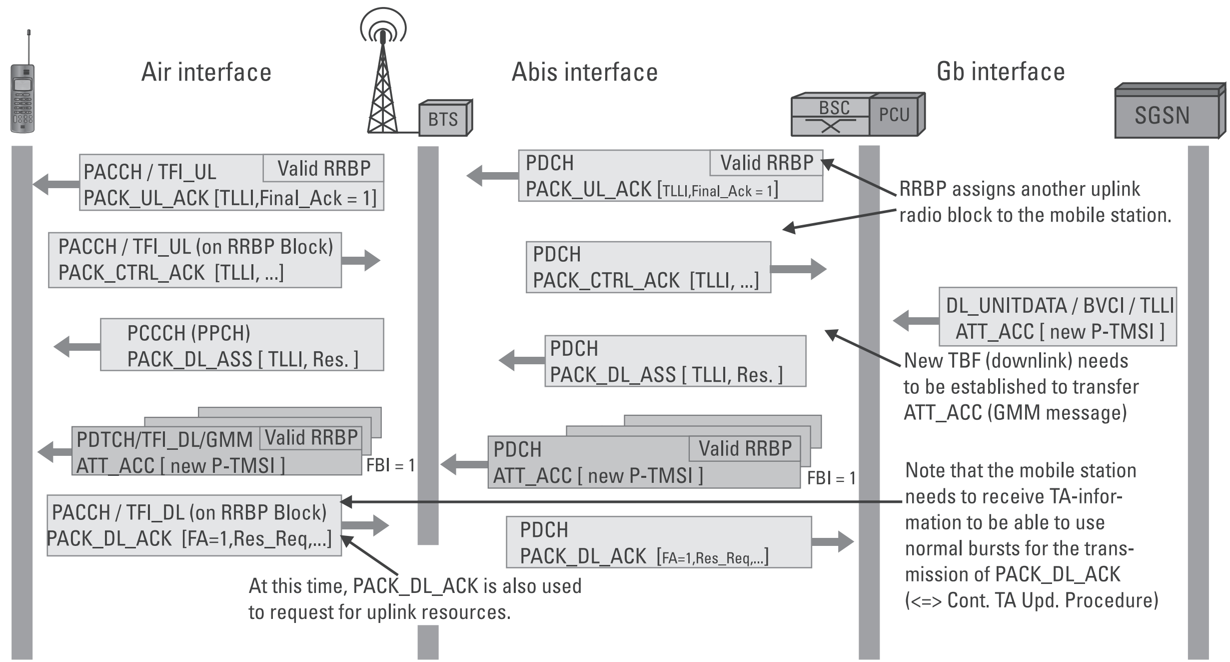


Figure 4.7 (continued).

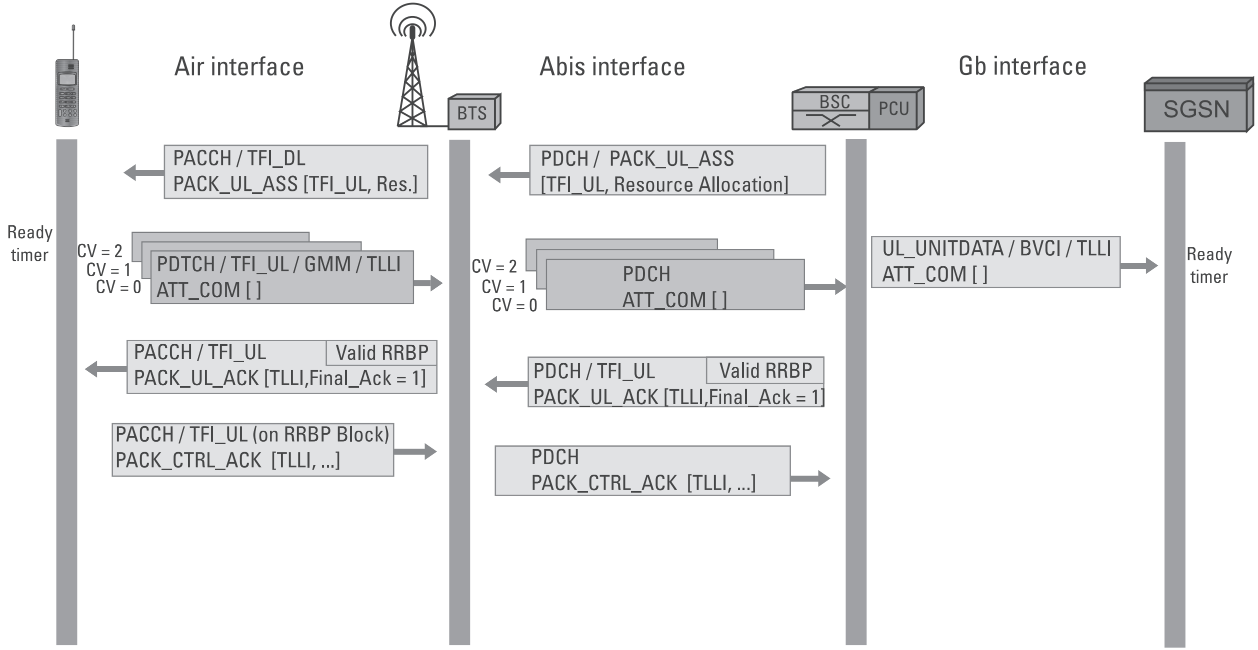


Figure 4.7 (continued).

```
+---------+-----------------------------------+-----------------------------------+
|BITMASK  |ID Name                            |Comment or Value                   |
+---------+-----------------------------------+-----------------------------------+
|GPRS Mobility/Session Management, SMG29 V6.4.2 (GMMSM642) ATRQ (= Attach request)|
|Attach request                                                                   |
|----1000 |Protocol Discriminator             |Mobility management for GPRS       |
|0000---- |Skip Indicator                     |Skip Indicator                     |
|00000001 |Message Type                       |1                                  |
|MS Network Capability                                                            |
|00000001 |IE Length                          |1                                  |
|0------- |GEA/1 algorithm supported          |GEA/1 not available                |
|-1------ |SM capabilities dedicated          |MT PTP SMS supported               |
|--0----- |SM capabilities GPRS               |No MT PTP SMS supported            |
|---1---- |UCS2 character support             |No preference                      |
|----01-- |SS Screening indicator             |Ellipsis & phase 2 error handle    |
|------00 |Spare Padding                      |0                                  |
|Attach type& CKSN                                                                |
|-----010 |Type of attach                     |GPRS only, IMSI attach             |
|----0--- |Spare                              |0                                  |
|0111---- |Key sequence                       |no key available/reserved          |
|DRX Parameter                                                                    |
|00000000 |Split PG cycle code                |0                                  |
|-----000 |Non-DRX timer                      |no none DRX mode                   |
|----0--- |SPLIT on CCCH                      |Not supported by MS                |
|0000---- |Spare                              |0                                  |
|Mobile Identity                                                                  |
|00000101 |IE Length                          |5                                  |
|-----100 |Type of identity                   |TMSI/P-TMSI                        |
|----0--- |Odd/Even Indicator                 |Even no of digits                  |
|1111---- |Filler                             |15                                 |
|***B4*** |MID TMSI                           |c0 c0 00 0d                        |
|Routing Area Identification                                                      |
|**b12*** |MCC number                         |999                                |
|1111---- |Filler                             |15                                 |
|----0000 |MNC digit 1                        |0                                  |
|0001---- |MNC digit 2                        |1                                  |
|***B2*** |LAC                                |45321                              |
|00101000 |RAC                                |07                                 |
|MS Radio Access Capability                                                       |
|00001001 |IE Length                          |9                                  |
|0001---- |1 Access Technology Type           |GSM E                              |
|***b7*** |1 Access technology type len       |33                                 |
|---100-- |1 RF Power CAP1ability             |4                                  |
|------1- |1 Encryption Algorithm Flag        |present                            |
|-------1 |1 A5/1                             |available                          |
|1------- |1 A5/2                             |available                          |
|-0------ |1 A5/3                             |not available                      |
|--0----- |1 A5/4                             |not available                      |
|---0---- |1 A5/5                             |not available                      |
|----0--- |1 A5/6                             |not available                      |
|-----0-- |1 A5/7                             |not available                      |
|------1- |1 Early classmark                  |implemented                        |
|-------0 |1 Pseudo Synchronization           |not present                        |
|0------- |1 Voice Group Call Service         |no VGCS wanted                     |
|-0------ |1 Voice Broadcast Service          |no VBS wanted                      |
|--1----- |1 MultiSlot CAP1ability flag       |present                            |
|---0---- |1 HSCSD MultiSlot Flag             |not present                        |
|----1--- |1 GPRS MultiSlot/Ext. Flag         |present                            |
|***b5*** |1 GPRS MultiSlot Class             |8                                  |
|--1----- |1 GPRS Extended Dynamic Allo. CAP1.|implemented                        |
|---1---- |1 SMS and SM Value Flag            |present                            |
|----1101 |1 Switch Measure Switch            |14/8 time slot                     |
|1000---- |1 Switch Measure                   |9/8 time slot                      |
|----1--- |1 MS RA capability Flag            |present                            |
|***b4*** |2 Access Technology Type           |GSM 1800                           |
|-0001001 |2 Access technology type len       |9                                  |
|001----- |2 RF Power CAP1ability             |1                                  |
|---0---- |2 Encryption Algorithm Flag        |not present                        |
|----1--- |2 Early classmark                  |implemented                        |
|-----0-- |2 Pseudo Synchronization           |not present                        |
|------0- |2 Voice Group Call Service         |no VGCS wanted                     |
|-------0 |2 Voice Broadcast Service          |no VBS wanted                      |
|0------- |2 MultiSlot CAP1ability flag       |not present                        |
|-0------ |2 MS RA capability Flag            |not present                        |
|--000000 |Padding                            |0                                  |
+---------+-----------------------------------+-----------------------------------+
```

Figure 4.8 Example of a GPRS attach scenario.

```
+---------+-------------------------------+------------------------------------+
|BITMASK  |ID Name                        |Comment or Value                    |
+---------+-------------------------------+------------------------------------+
|GPRS Mobility Management, SMG29 V6.4.2GMMSM642 ACRQ (= Authentication &       |
|ciphering request)                                                            |
|Authentication & ciphering request                                            |
|----1000 |Protocol Discriminator         |Mobility management for GPRS        |
|0000---- |Skip Indicator                 |Skip Indicator                      |
|00010010 |Message Type                   |18                                  |
|-----000 |Ciphering algorithm            |ciphering not used                  |
|----0--- |Spare                          |0                                   |
|-000---- |IMEISV request value           |IMEISV not requested                |
|0------- |Spare                          |0                                   |
|-----000 |Force to standby value         |Force to standby not indicated      |
|----0--- |Spare                          |0                                   |
|1111---- |A&C reference number           |15                                  |
|Authentication parameter RAND                                                 |
|00100001 |IE Name                        |Authentication parameter RAND       |
|**B16*** |RAND Value                     |cb e3 f9 67 0b d5 c3 b0 2e 47 70 ...|
|Ciphering Key Sequence Number                                                 |
|1000---- |IE Name                        |Ciphering Key Sequence Number       |
|----0111 |Key sequence                   |no key available/reserved           |
+---------+-------------------------------+------------------------------------+
```

```
+---------+-------------------------------+------------------------------------+
|BITMASK  |ID Name                        |Comment or Value                    |
+---------+-------------------------------+------------------------------------+
|GPRS Mobility Management, SMG29 V6.4.2GMMSM642 ACRE (= Authentication &       |
|ciphering response)                                                           |
|Authentication & ciphering response                                           |
|----1000 |Protocol Discriminator         |Mobility management for GPRS        |
|0000---- |Skip Indicator                 |Skip Indicator                      |
|00010011 |Message Type                   |19                                  |
|----1111 |A&C reference number           |15                                  |
|0000---- |Spare                          |0                                   |
|Authentication parameter SRES                                                 |
|00100010 |IE Name                        |Authentication parameter SRES       |
|***B4*** |Authen response signature      |b9 74 c2 6e                         |
+---------+-------------------------------+------------------------------------+
```

Figure 4.8 (continued).

It is also performed when the periodic routing area updating timer (T3312) has expired.

We have shown the routing area updating procedure in Figure 4.11 first as an intra-SGSN routing area updating procedure, while Figure 4.12 also shows the inter-SGSN routing area updating procedure. Figure 4.12 illustrates a case in which a downlink data transfer is activated during routing area updating. In Figure 4.13 we show a periodic routing area updating scenario from an active tracefile.

4.1.2.4 The Cell Update Procedure and the Ready State

We have already mentioned in Section 2.3.4 that the cell update procedure takes the place of the handover in GPRS. The term "cell update" itself indicates that this procedure is always performed between mobile station and network when the mobile station performs a cell reselection. This is not, however, entirely correct, and it would also not be very economical

```
+---------+--------------------------------+------------------------------------+
|BITMASK  |ID Name                         |Comment or Value                    |
+---------+--------------------------------+------------------------------------+
|GPRS Mobility/Session Management, SMG29 V6.4.2 (GMMSM642) ATAC (= Attach accept) |
|Attach accept                                                                   |
|----1000 |Protocol Discriminator          |Mobility management for GPRS        |
|0000---- |Skip Indicator                  |Skip Indicator                      |
|00000010 |Message Type                    |2                                   |
|-----001 |Result of attach                |GPRS only attach                    |
|----0--- |Spare                           |0                                   |
|-000---- |Force to standby value          |Force to standby not indicated      |
|0------- |Spare                           |0                                   |
|Timer                                                                           |
|---01000 |Timer value                     |8                                   |
|010----- |Timer value unit                |Value of steps : 6 min              |
|Radio Priority Level & Spare                                                    |
|-----100 |Radio priority level value      |priority level 4: lowest            |
|----0--- |Spare                           |0                                   |
|0000---- |Spare                           |0                                   |
|Routing Area Identification                                                     |
|**b12*** |MCC number                      |999                                 |
|1111---- |Filler                          |15                                  |
|----0000 |MNC digit 1                     |0                                   |
|0001---- |MNC digit 2                     |1                                   |
|***B2*** |LAC                             |45321                               |
|00101000 |RAC                             |07                                  |
|Timer                                                                           |
|00010111 |IE Name                         |Timer                               |
|---01111 |Timer value                     |15                                  |
|001----- |Timer value unit                |Value of steps : 1 min              |
|Mobile Identity                                                                 |
|00011000 |IE Name                         |Mobile Identity                     |
|00000101 |IE Length                       |5                                   |
|-----100 |Type of identity                |TMSI/P-TMSI                         |
|----0--- |Odd/Even Indicator              |Even no of digits                   |
|1111---- |Filler                          |15                                  |
|***B4*** |MID TMSI                        |c0 c0 00 0e                         |
+---------+--------------------------------+------------------------------------+
```

```
+---------+--------------------------------+------------------------------------+
|BITMASK  |ID Name                         |Comment or Value                    |
+---------+--------------------------------+------------------------------------+
|GPRS Mobility/Session Management, SMG29 V6.4.2 (GMMSM642) ACOM (= Attach complete)|
|Attach complete                                                                 |
|----1000 |Protocol Discriminator          |Mobility management for GPRS        |
|0000---- |Skip Indicator                  |Skip Indicator                      |
|00000011 |Message Type                    |3                                   |
+---------+--------------------------------+------------------------------------+
```

Figure 4.8 (continued).

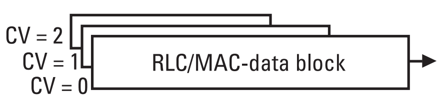


Figure 4.9 Presentation of the UL TBF release in the following scenarios.

Figure 4.10 Presentation of the DL TBF release in the following scenarios.

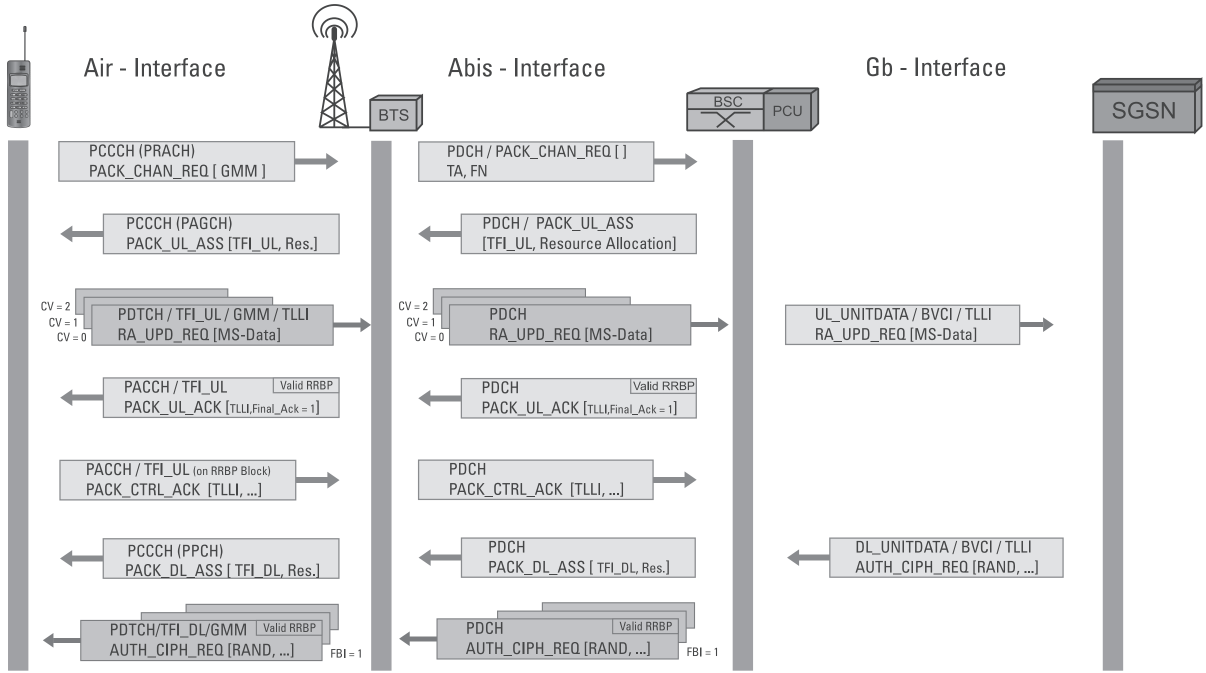


Figure 4.11 Routing area update via the PCCCH.

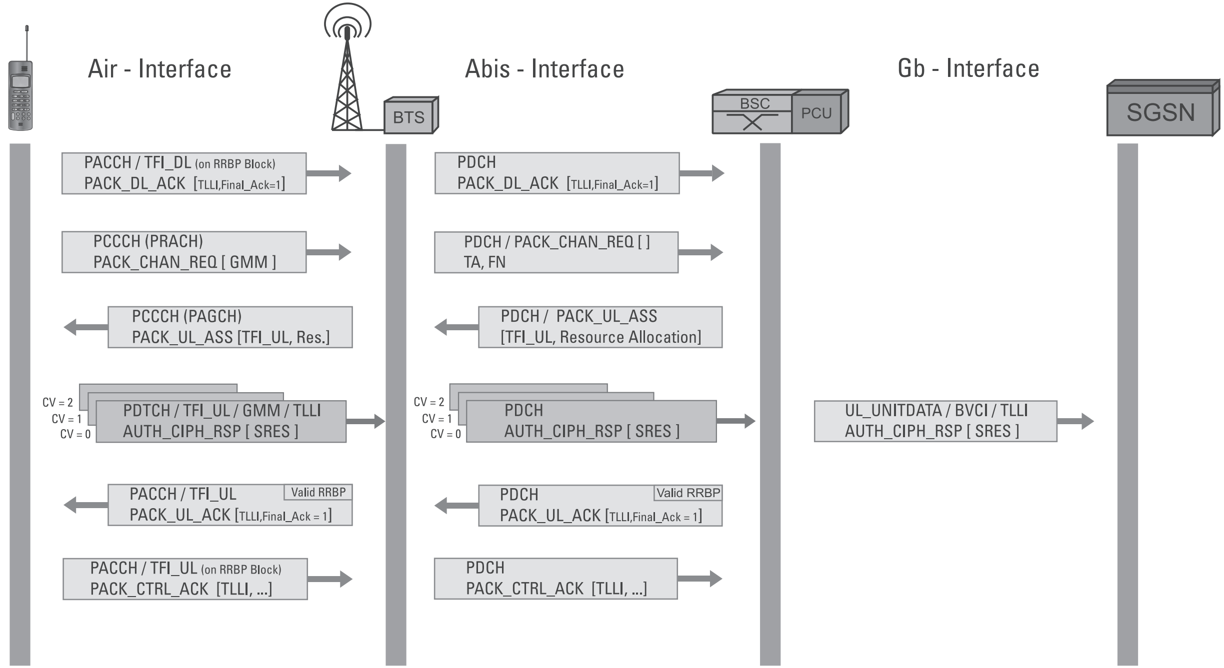


Figure 4.11 (continued).

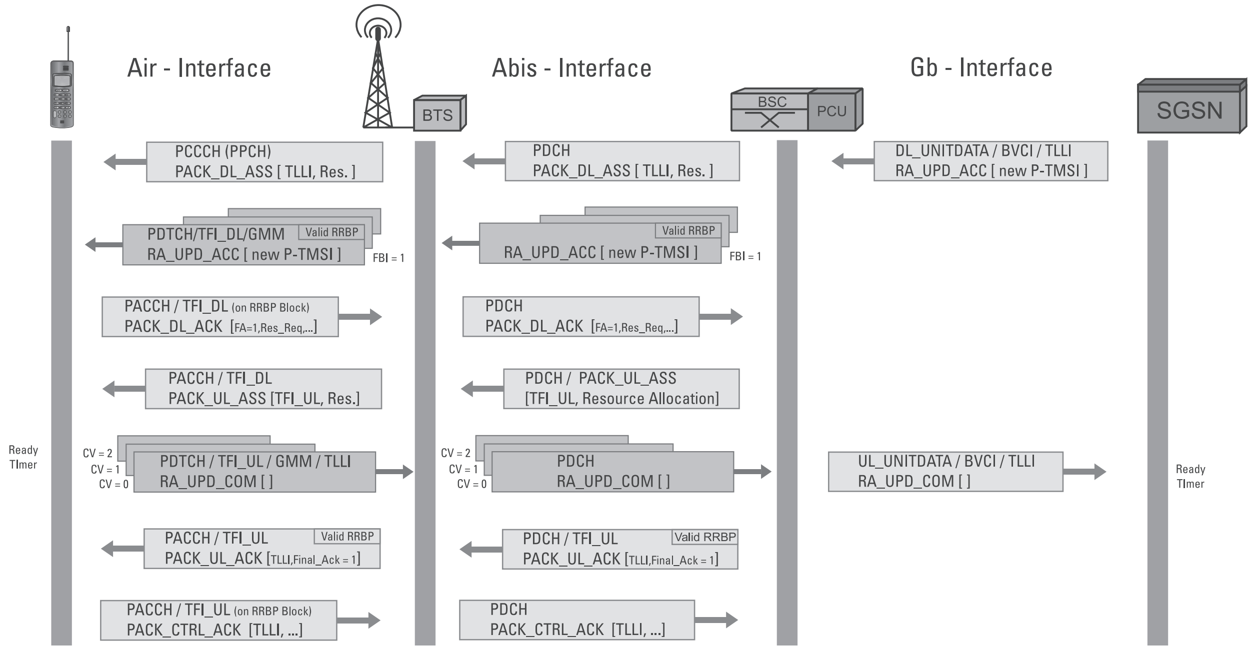


Figure 4.11 (continued).

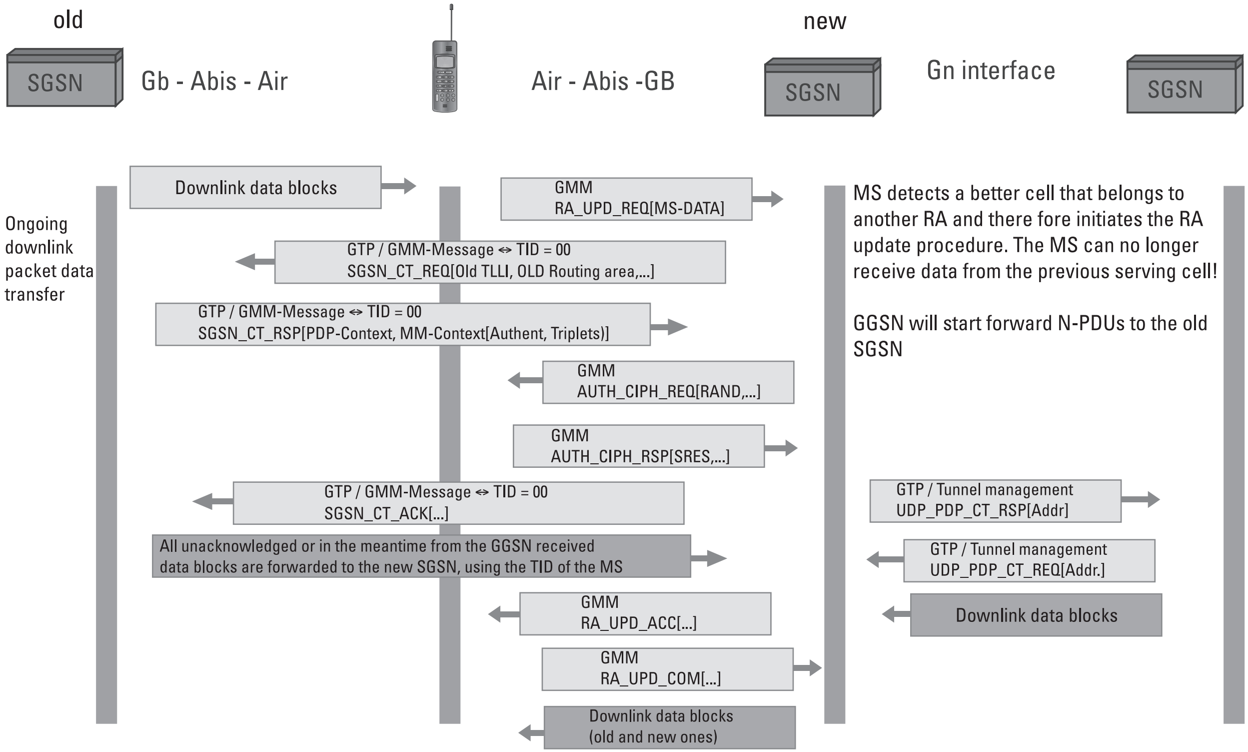


Figure 4.12 Routing area update with SGSN change during data transfer.

```
+---------+-----------------------------------+---------------------------------+
|BITMASK  |ID Name                            |Comment or Value                 |
+---------+-----------------------------------+---------------------------------+
|GPRS Mobility Management, SMG29 V6.4.2 (GMMSM642)  RARQ (= Routing area update request)|
|Routing area update request                                                    |
|----1000 |Protocol Discriminator             |Mobility management for GPRS     |
|0000---- |Skip Indicator                     |Skip Indicator                   |
|00001000 |Message Type                       |8                                |
|-----011 |Update type value                  |Periodic updating                |
|----0--- |Spare                              |0                                |
|0111---- |Key sequence                       |no key available/reserved        |
|Routing Area Identification                                                    |
|**b12*** |MCC number                         |999                              |
|1111---- |Filler                             |15                               |
|----0000 |MNC digit 1                        |0                                |
|0001---- |MNC digit 2                        |1                                |
|***B2*** |LAC                                |87666                            |
|00101000 |RAC                                |09                               |
|MS Radio Access Capability                                                     |
|00001001 |IE Length                          |9                                |
|0001---- |1 Access Technology Type           |GSM E                            |
|***b7*** |1 Access technology type len       |33                               |
|---100-- |1 RF Power CAP1ability             |4                                |
|------1- |1 Encryption Algorithm Flag        |present                          |
|-------1 |1 A5/1                             |available                        |
|1------- |1 A5/2                             |available                        |
|-0------ |1 A5/3                             |not available                    |
|--0----- |1 A5/4                             |not available                    |
|---0---- |1 A5/5                             |not available                    |
|----0--- |1 A5/6                             |not available                    |
|-----0-- |1 A5/7                             |not available                    |
|------1- |1 Early classmark                  |implemented                      |
|-------0 |1 Pseudo Synchronization           |not present                      |
|0------- |1 Voice Group Call Service         |no VGCS wanted                   |
|-0------ |1 Voice Broadcast Service          |no VBS wanted                    |
|--1----- |1 MultiSlot CAP1ability flag       |present                          |
|---0---- |1 HSCSD MultiSlot Flag             |not present                      |
|----1--- |1 GPRS MultiSlot/Ext. Flag         |present                          |
|***b5*** |1 GPRS MultiSlot Class             |12                               |
|--1----- |1 GPRS Extended Dynamic Allo. CAP1.|implemented                      |
|---1---- |1 SMS and SM Value Flag            |present                          |
|----1101 |1 Switch Measure Switch            |14/8 time slot                   |
|1000---- |1 Switch Measure                   |9/8 time slot                    |
|----1--- |1 MS RA capability Flag            |present                          |
|***b4*** |2 Access Technology Type           |GSM 1800                         |
|-0001001 |2 Access technology type len       |9                                |
|001----- |2 RF Power CAP1ability             |1                                |
|---0---- |2 Encryption Algorithm Flag        |not present                      |
|----1--- |2 Early classmark                  |implemented                      |
|-----0-- |2 Pseudo Synchronization           |not present                      |
|------0- |2 Voice Group Call Service         |no VGCS wanted                   |
|-------0 |2 Voice Broadcast Service          |no VBS wanted                    |
|0------- |2 MultiSlot CAP1ability flag       |not present                      |
|-0------ |2 MS RA capability Flag            |not present                      |
|--000000 |Padding                            |0                                |
|DRX Parameter                                                                  |
|00100111 |IE Name                            |DRX Parameter                    |
|00000000 |Split PG cycle code                |0                                |
|-----000 |Non-DRX timer                      |no none DRX mode                 |
|----0--- |SPLIT on CCCH                      |Not supported by MS              |
|0000---- |Spare                              |0                                |
+---------+-----------------------------------+---------------------------------+
```

Figure 4.13 Example of a routing area update scenario.

```
+---------+-------------------------------+-----------------------------------+
|BITMASK  |ID Name                        |Comment or Value                   |
+---------+-------------------------------+-----------------------------------+
|GPRS Mobility Management, SMG29 V6.4.GMMSM642  ACRQ (= Authentication &      |
|ciphering request)                                                           |
|Authentication & ciphering request                                           |
|----1000 |Protocol Discriminator         |Mobility management for GPRS       |
|0000---- |Skip Indicator                 |Skip Indicator                     |
|00010010 |Message Type                   |18                                 |
|-----000 |Ciphering algorithm            |ciphering not used                 |
|----0--- |Spare                          |0                                  |
|-000---- |IMEISV request value           |IMEISV not requested               |
|0------- |Spare                          |0                                  |
|-----000 |Force to standby value         |Force to standby not indicated     |
|----0--- |Spare                          |0                                  |
|1111---- |A&C reference number           |15                                 |
|Authentication parameter RAND                                                |
|00100001 |IE Name                        |Authentication parameter RAND      |
|**B16*** |RAND Value                     |5c 59 09 d2 cd ff 41 d9 71 57 6a ...|
|Ciphering Key Sequence Number                                                |
|1000---- |IE Name                        |Ciphering Key Sequence Number      |
|----0111 |Key sequence                   |no key available/reserved          |
+---------+-------------------------------+-----------------------------------+
```

```
+---------+-------------------------------+-----------------------------------+
|BITMASK  |ID Name                        |Comment or Value                   |
+---------+-------------------------------+-----------------------------------+
|GPRS Mobility Management, SMG29 V6.4.2GMMSM642 ACRE (= Authentication &      |
|ciphering response)                                                          |
|Authentication & ciphering response                                          |
|----1000 |Protocol Discriminator         |Mobility management for GPRS       |
|0000---- |Skip Indicator                 |Skip Indicator                     |
|00010011 |Message Type                   |19                                 |
|----1111 |A&C reference number           |15                                 |
|0000---- |Spare                          |0                                  |
|Authentication parameter SRES                                                |
|00100010 |IE Name                        |Authentication parameter SRES      |
|***B4*** |Authen response signature      |eb 69 ab 85                        |
+---------+-------------------------------+-----------------------------------+
```

Figure 4.13 (continued).

because such a procedure would cause a too heavy signaling burden. Indeed, another way has been chosen:

- For GPRS mobility management, the ready state is introduced as a new status in the mobile station and the SGSN.
- Every time the mobile station transmits a data block (LLC frame) to the SGSN, or rather every time the SGSN receives a data block from a certain mobile station, the ready timer (T3314) is restarted in the mobile station and the SGSN for this purpose.
- As long as the mobile station is in this ready state, a cell update scenario is carried out with a cell reselection. This is shown in Figure 4.14. If, however, the mobile station also changes the routing area during cell reselection, a routing area update is performed instead of the cell update.
- As illustrated in Figure 4.14, the cell update scenario consists of the mobile station sending an LLC frame to the SGSN from the newly

```
+---------+-------------------------------+-----------------------------------+
|BITMASK  |ID Name                        |Comment or Value                   |
+---------+-------------------------------+-----------------------------------+
|GPRS Mobility Management, SMG29 V6.4.2 (GMMSM642) RAAC (= Routing area update accept) |
|Routing area update accept                                                   |
|----1000 |Protocol Discriminator         |Mobility management for GPRS       |
|0000---- |Skip Indicator                 |Skip Indicator                     |
|00001001 |Message Type                   |9                                  |
|-----000 |Force to standby value         |Force to standby not indicated     |
|----0--- |Spare                          |0                                  |
|-000---- |Update result value            |RA updated only                    |
|0------- |Spare                          |0                                  |
|Timer                                                                        |
|---00011 |Timer value                    |3                                  |
|001----- |Timer value unit               |Value of steps : 1 min             |
|Routing Area Identification                                                  |
|**b12*** |MCC number                     |999                                |
|1111---- |Filler                         |15                                 |
|----0000 |MNC digit 1                    |0                                  |
|0001---- |MNC digit 2                    |1                                  |
|***B2*** |LAC                            |87666                              |
|00101000 |RAC                            |09                                 |
|Mobile Identity                                                              |
|00011000 |IE Name                        |Mobile Identity                    |
|00000101 |IE Length                      |5                                  |
|-----100 |Type of identity               |TMSI/P-TMSI                        |
|----0--- |Odd/Even Indicator             |Even no of digits                  |
|1111---- |Filler                         |15                                 |
|***B4*** |MID TMSI                       |c1 40 00 ad                        |
|Timer                                                                        |
|00010111 |IE Name                        |Timer                              |
|---00101 |Timer value                    |5                                  |
|001----- |Timer value unit               |Value of steps : 1 min             |
+---------+-------------------------------+-----------------------------------+
```

```
+---------+-------------------------------+-----------------------------------+
|BITMASK  |ID Name                        |Comment or Value                   |
+---------+-------------------------------+-----------------------------------+
|GPRS Mobility Management, SMG29 V6.4.2 (GMMSM642) RACO (= Routing area update complete)|
|Routing area update complete                                                 |
|----1000 |Protocol Discriminator         |Mobility management for GPRS       |
|0000---- |Skip Indicator                 |Skip Indicator                     |
|00001010 |Message Type                   |10                                 |
+---------+-------------------------------+-----------------------------------+
```

Figure 4.13 (continued).

selected cell. If the mobile station does not have any data to be transmitted in its output loop at this time, the LLC frame is empty. Depending on the last LLC operating mode, this LLC frame is either a UI frame or an RR frame.

- The significance of sending an empty LLC frame is that the SGSN also contains information about the new serving cell of the mobile station during the cell update scenario. Consequently, the SGSN can omit the paging procedure in downlink data to be transmitted and give the mobile station direct resource allocation in the serving cell. This saves time on paging and paging response.
- In fact, an explicit paging response message is replaced in GPRS by transmitting an LLC frame. If a mobile station has to be paged for

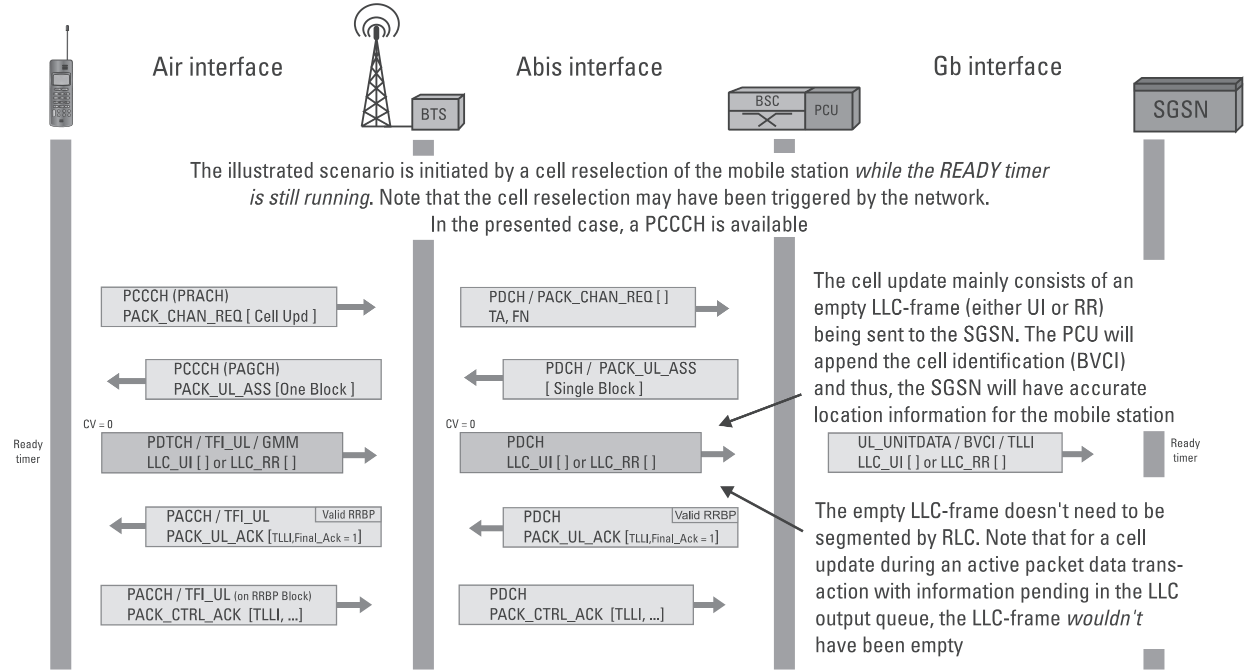


Figure 4.14 A cell update via the PCCCH.

transmitting downlink data in the standby state, the mobile station answers with an empty LLC frame and thus goes into the ready state just like the SGSN. In the ready state, direct resource allocation to the mobile station can now take place (see Figure 4.14).

- Furthermore, if a downlink TBF is active during cell update, the SGSN only receives the information that the mobile station has changed service cells via the cell update scenario. Accordingly, all RLC/MAC data blocks that have already been sent to the former serving cell but have not yet been confirmed by the mobile station must be sent again to the new cell.
- The handover function for packet switched GSM can thus be replaced by the combination of ready state and ready timer.
- The standard value of the ready timer T3314 is 44 seconds. If the ready timer is set at the maximum value, 11111_2, the standby state as featured in Figure 4.15 does not apply. The mobile station is permanently in the ready state. If the ready timer is set at 0, there is not a ready state in Figure 4.15 but only a standby state and an idle state.
- The setting or discussion of the ready timer is optional during the GPRS attach procedure and/or routing area updating.

The question remains as to whether, by introducing the ready state, it is always the mobile station that decides whether to perform a handover.

This is actually a question of the setting of the network control order system parameter. Table 4.2 shows the possible settings. One example is provided in the SYS_INFO13 in Figure 3.5. Note that the setting of the

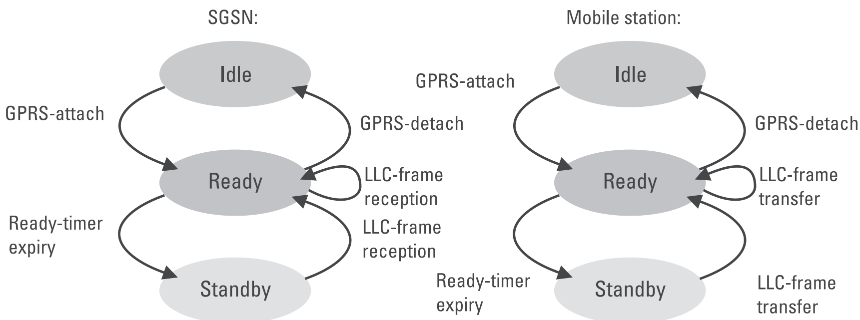


Figure 4.15 GMM status in mobile station and SGSN.

Table 4.2
The Network Control Order Regulates the Reactions in the Ready State

Network Control Order	Means
NC0	The mobile station does not transmit any measuring results to the PCU and performs a cell update scenario autonomously (i.e., without being checked by the PCU)
NC1	The mobile station transmits measuring results to the PCU but still performs a cell update scenario autonomously (i.e., without being checked by the PCU)
NC2	The mobile station transmits measuring results to the PCU and the PCU checks if and when the mobile station performs a cell update scenario
NC3	To be interpreted as NC0 (for future use)

network control order only regulates the reaction of the mobile station in ready state. In the standby state, the mobile station will still carry out a normal cell reselection without informing the SGSN.

Let us summarize:

- The GMM ready state enables direct resource allocation to a mobile station in the serving cell without having to page the mobile station in the routing area.
- As long as the mobile station is in the ready state, a cell update scenario is performed in cell reselection, through which the SGSN receives the precise cell information from the mobile station.
- The introduction of the ready state essentially saves time in the repeated, short-term allocation of downlink resources to a mobile station that is changing serving cells.

4.2 QoS in GPRS

In my first book on the subject of GSM signaling, I devoted an entire chapter to quality of service in GSM. The main point of this chapter is to present criteria and parameters for determining GSM network quality. An additional focal point is the presentation of the different measuring tools, from the protocol tester to the OMC.

The main problem in determining GSM network quality is the lack of description of quality parameters for GSM networks in the GSM standards. For this reason, presenting and comparing the quality of GSM networks is often like comparing apples and oranges.

For GPRS, parameters for determining network quality have been included in the standard from the start. It must also be made clear, however, that these parameters change with the different versions or releases of GPRS. In fact, the parameters presented in the following only apply to GPRS releases 1997 and 1998.

The new quality parameters that are also valid for UMTS, which are presented for the first time in release 1999, are still at the standardizing stage and should thus be saved for a future edition of this book.

4.2.1 Discussion of the QoS Profile

Basically, there is one or more QoS profiles stored in the HLR for every GPRS customer. Whether a GSM/GPRS network operator offers his various customer groups different QoS profiles at different rates or the same QoS profile (e.g., best effort) for all customers is not prescribed. Every network operator can decide for themselves.

Before a GPRS customer can carry out data transmission anywhere in the world, the QoS between the GPRS customer and the network (SGSN and GGSN), which has to be prepared by the network, must be discussed and agreed upon. This happens during the activation of the respective PDP context. We will discuss the activation of a PDP context in Section 4.3. In any case, it must be noted that the mobile station proposes a QoS profile during the activation of the PDP context, but the network has the final decision as to which QoS can be made available. An example of the parameter values proposed by the mobile station can be seen in Figure 4.16.

4.2.2 QoS Parameters in GPRS Releases 97 and 98

The QoS parameters listed in Table 4.3 will be explained in more detail in the following sections.

4.2.2.1 Service Precedence/Priority

The service precedence/priority states to what extent a subscriber should preferentially receive resources (Table 4.4). This obviously applies in the case of burden. For the area between mobile station and PCU (RLC/MAC), the service precedence/priority must be translated into the radio priority parameter.

```
|GPRS Mobility/Session Managment SMG29 v5.6.7 (GMMSM34) APRQ(= Activate PDP context request) |
|Activate PDP context request                                                                |
|----1010 |Protocol Discriminator                  |Session management messages             |
|-000---- |Transaction Id value                    |TI value 0                              |
|0------- |Transaction Id flag                     |message sent from orig TI               |
|01000001 |Message Type                            |65                                      |
|Network Service Access Point                                                                |
|----0101 |NSAPI value                             |NSAPI 5                                 |
|0000---- |Spare                                   |0                                       |
|LLC SAPI                                                                                    |
|----0101 |SAPI                                    |SAPI 5                                  |
|0000---- |Spare                                   |0                                       |
|Quality of Service                                                                          |
|00000011 |IE Length                               |3                                       |
|-----011 |Reliability class                       |Unack. GTP&LLC,Ack.RLC,Prot. data       |
|--100--- |Delay class                             |Delay class 4 (best effort)             |
|00------ |Spare                                   |0                                       |
|-----011 |Precedence class                        |Low priority                            |
|----0--- |Spare                                   |0                                       |
|0101---- |Peak throughput                         |Up to 16000 octet/s                     |
|---11111 |Mean throughput                         |best effort                             |
|000----- |Spare                                   |0                                       |
|Packet Data Protocol Address                                                                |
|00000010 |IE Length                               |2                                       |
|----0001 |Type of address                         |IETF specified address                  |
|0000---- |Spare                                   |0                                       |
|00100001 |Packet data protocol type               |IPv4                                    |
|Access Point Name                                                                           |
|00101000 |IE Name                                 |Access Point Name                       |
|00001101 |IE Length                               |13                                      |
|**B13*** |Access Point Name Value                 |03 90 77 73 50 38 44 66 32 2c 89 ...    |
|Protocol Configuration Options                                                              |
|00100111 |IE Name                                 |Protocol Configuration Options          |
|00010101 |IE Length                               |21                                      |
|-----000 |Options format value                    |PPP                                     |
|-0000--- |Spare                                   |0                                       |
|1------- |Extension bit                           |Octet 3 is extended                     |
|**B20*** |Address information                     |67 56 45 34 45 56 67 89 67 55 32 ...    |
```

Figure 4.16 Example of the QoS requirements of a mobile station during the activation of a PDP context.

Table 4.3
The Five QoS Parameters for GPRS Releases 97 and 98

Parameter	Means
Service precedence/priority	With what priority does a GPRS subscriber receive access to the network resources?
Delay class	What should the maximum delay time be for the transmission of a data packet in the GPRS network?
Mean throughput rate	How high should the average transmission rate (in octets/h) in the GPRS network be?
Peak throughput rate	How high should the maximum possible differential transmission rate (in octets/s) in the GPRS network be?
Reliability class	Which protocols in the GPRS protocol stack should carry out backward error correction?

Table 4.4
The Service Precedence/Priority QoS Parameter

Precedence	Precedence Name	Interpretation
1	High priority	Service commitments shall be maintained ahead of precedence classes 2 and 3
2	Normal priority	Service commitments shall be maintained ahead of precedence class 3
3	Low priority	Service commitments shall be maintained after precedence classes 1 and 2

4.2.2.2 Delay Class

The delay class parameter states the maximum delay time for the transfer of a single data packet through the GPRS network. Here, the following points have to be considered:

- Delay class refers exclusively to delay times within the GPRS network. External influences are therefore ignored.
- Table 4.5 shows, in delay class 3, a maximum permitted delay time of 375 seconds for a data packet with 1,024 bytes. This is more than 6 minutes. This value is obviously unrealistic for Internet applications. The values in Table 4.5 were not actually developed for GSM or GPRS, but stem from the ITU-T recommendations, X.135/X.140, which define QoS for packet switched networks.
- Service data unit (SDU) refers to the net data packet of an OSI-Layer X.

Table 4.5
The Delay Class QoS Parameter

	Delay (Maximum Values)			
	SDU Size: 128 Octets		SDU Size: 1,024 Octets	
Delay Class	Mean Transfer Delay (sec)	95 Percentile Delay (sec)	Mean Transfer Delay	95 Percentile Delay (sec)
1. (Predictive)	< 0.5	< 1.5	< 2	< 7
2. (Predictive)	< 5	< 25	< 15	< 75
3. (Predictive)	< 50	< 250	< 75	< 375
4. (Best effort)	Unspecified			

4.2.2.3 Mean Throughput Rate

The mean throughput rate is measured in bytes or octets per hour. Table 4.6 shows the valid table for release 1997, while Table 4.7 shows the table for release 1998. Clearly, the value for best effort has changed from 1 in release 97 to 31 in release 98.

Incidentally, for the QoS parameters valid as from release 1999, the mean throughput rate is principally mapped to best effort (i.e., the value of 31).

4.2.2.4 Peak Throughput Rate

The peak throughput rate illustrated in Table 4.8 shows the maximum throughput rates in octets per second. Note that only the first values in the table are actually relevant for GPRS. The higher throughput rates already show beginnings of the standardization of the QoS parameters for GSM and the third generation mobile communication networks.

4.2.2.5 Reliability Class

The final QoS parameter is reliability class. Attempts have already been made in ETSI to examine the QoS of different kinds of traffic using this parameter.

Table 4.6
The QoS Parameter, Mean Throughput Rate for GPRS Release 1997

Mean Throughput Class	Mean Throughput in Octets Per Hour
1	Best effort
2	100 (~0.22 bps)
3	200 (~0.44 bps)
4	500 (~1.11 bps)
5	1,000 (~2.2 bps)
6	2,000 (~4.4 bps)
7	5,000 (~11.1 bps)
8	10,000 (~22 bps)
9	20,000 (~44 bps)
10	50,000 (~111 bps)
11	100,000 (~0.22 Kbps)
12	200,000 (~0.44 Kbps)
13	500,000 (~1.11 Kbps)
14	1,000,000 (~2.2 Kbps)
15	2,000,000 (~4.4 Kbps)
16	5,000,000 (~11.1 Kbps)
17	10,000,000 (~22 Kbps)
18	20,000,000 (~44 Kbps)
19	50,000,000 (~111 Kbps)

Table 4.7
The QoS Parameter, Mean Throughput Rate for GPRS Release 1998

Mean Throughput Class	Mean Throughput Rate in Octets Per Hour
1	100 (~0.22 bps)
2	200 (~0.44 bps)
3	500 (~1.11 bps)
4	1,000 (~2.2 bps)
5	2,000 (~4.4 bps)
6	5,000 (~11.1 bps)
7	10,000 (~22 bps)
8	20,000 (~44 bps)
9	50,000 (~111 bps)
10	100,000 (~0.22 Kbps)
11	200,000 (~0.44 Kbps)
12	500,000 (~1.11 Kbps)
13	1,000,000 (~2.2 Kbps)
14	2,000,000 (~4.4 Kbps)
15	5,000,000 (~11.1 Kbps)
16	10,000,000 (~22 Kbps)
17	20,000,000 (~44 Kbps)
18	50,000,000 (~111 Kbps)
31	Best effort

Table 4.8
The QoS Parameter, Peak Throughput Rate

Peak Throughput Class	Peak Throughput in Octets Per Second
1	Up to 1,000 (8 Kbps)
2	Up to 2,000 (16 Kbps)
3	Up to 4,000 (32 Kbps)
4	Up to 8,000 (64 Kbps)
5	Up to 16,000 (128 Kbps)
6	Up to 32,000 (256 Kbps)
7	Up to 64,000 (512 Kbps)
8	Up to 128,000 (1,024 Kbps)
9	Up to 256,000 (2,048 Kbps)

This can be seen in the right-hand column of Table 4.9. This starting point will be followed up consistently in release 99 by orientating QoS to the four kinds of traffic: conversational, interactive, streaming, and background. The significance of the partial parameter, LLC data protection, will not be discussed until Chapter 5.

4.3 Session Management in GPRS

In circuit switched GSM, the call control (CC) protocol serves to establish a connection and the communication necessary for this between the mobile station and the MSC.

For packet switched GPRS the equivalent protocol is introduced, this is the session management protocol. The session management protocol is responsible for the administration of PDP contexts. Administration means, in particular, the installation, modification, and deactivation of PDP contexts.

Consequently, session management is responsible for the coordination of the service that the mobile station wishes to make use of. This includes particularly the following:

- *The choice of PDP and thus the service the mobile station wishes to use*. In release 97 and release 98, GPRS supports IP and the packet switched X.25 protocol. In contrast, GPRS release 99 still supports IP and, instead of X.25, the PPP and IHOSS. IHOSS essentially provides a bit pump between mobile station and GGSN.
- *Deciding which QoS profile the mobile station should use*. Here the mobile station finally has to accept what the network can supply or cancel the activation of the PDP context.
- *Deciding which PDP address the mobile station should use*. If the Internet is to remain the main application for GPRS, a difference must be made between dynamic IP address allocation and an already available fixed IP address.
- *The selection of an network service access point identifier (NSAPI) by the mobile station*. The NSAPI serves to differentiate several simultaneously activated PDP contexts (Figure 4.17). The mobile station can allocate NSAPI values between 5 and $15_{(10)}$.
- *As an option, the mobile station naming an APN for identifying a particular GGSN*.

Table 4.9
The QoS Parameter, Reliability Class

Class	GTP Mode	LLC Frame Mode	LLC Data Protection	RLC Block Mode	Traffic Type
1	Acknowledged	Acknowledged	Protected	Acknowledged	Non-real-time traffic, error-sensitive application that cannot cope with data loss
2	Unacknowledged	Acknowledged	Protected	Acknowledged	Non-real-time traffic, error-sensitive application that can cope with infrequent data loss
3	Unacknowledged	Unacknowledged	Protected	Acknowledged	Non-real-time traffic, error-sensitive application that can cope with data loss, GMM/SM, and SMS
4	Unacknowledged	Unacknowledged	Protected	Unacknowledged	Real-time traffic, error-sensitive application that can cope with data loss
5	Unacknowledged	Unacknowledged	Unprotected	Unacknowledged	Real-time traffic, error nonsensitive application that can cope with data loss

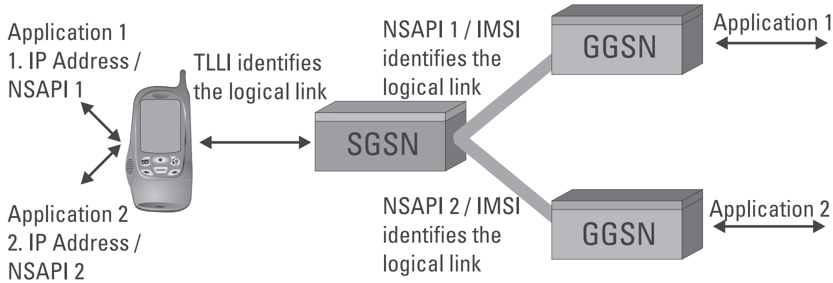


Figure 4.17 The significance of the NSAPI.

All the parameters named belong to a particular PDP context. Theoretically it is also imaginable for a mobile station to have 11 different PDP contexts active simultaneously by means of 11 possible NSAPI values.

4.3.1 The Activation of a PDP Context by the Mobile Station

Figure 4.18 shows the exchange of messages during the activation of a PDP context by the mobile station. All system interfaces between mobile station and GGSN are shown. The Abis and air interfaces are collated. Figure 4.19 shows the opposite case (i.e., when the PDP context is activated by the network).

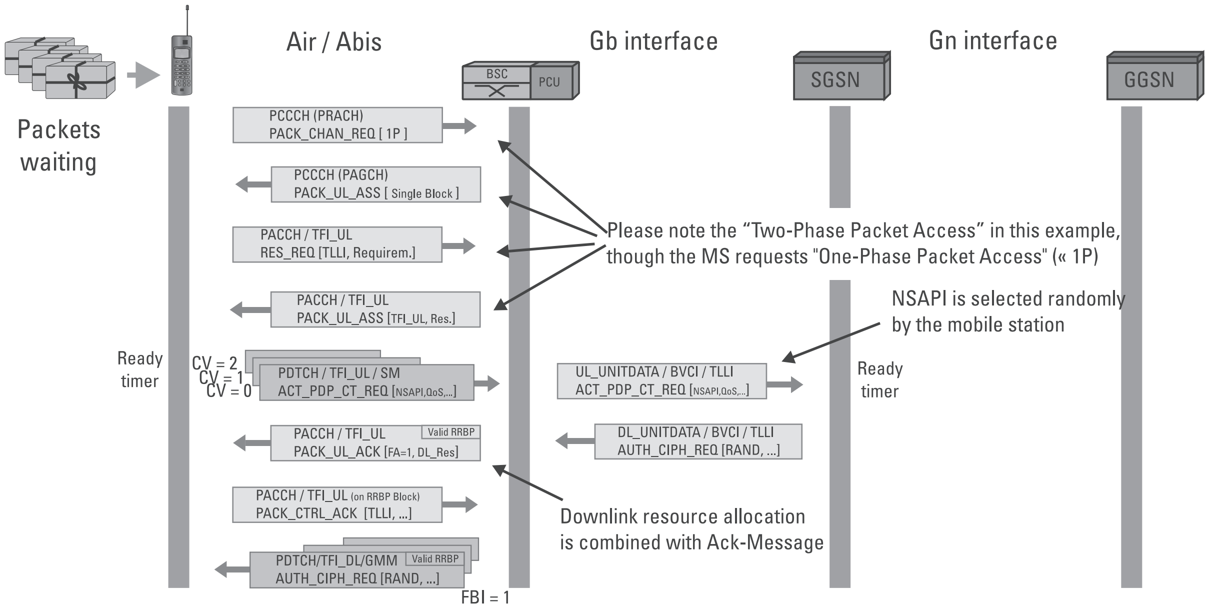


Figure 4.18 A mobile originating PDP context activation via the PCCCH.

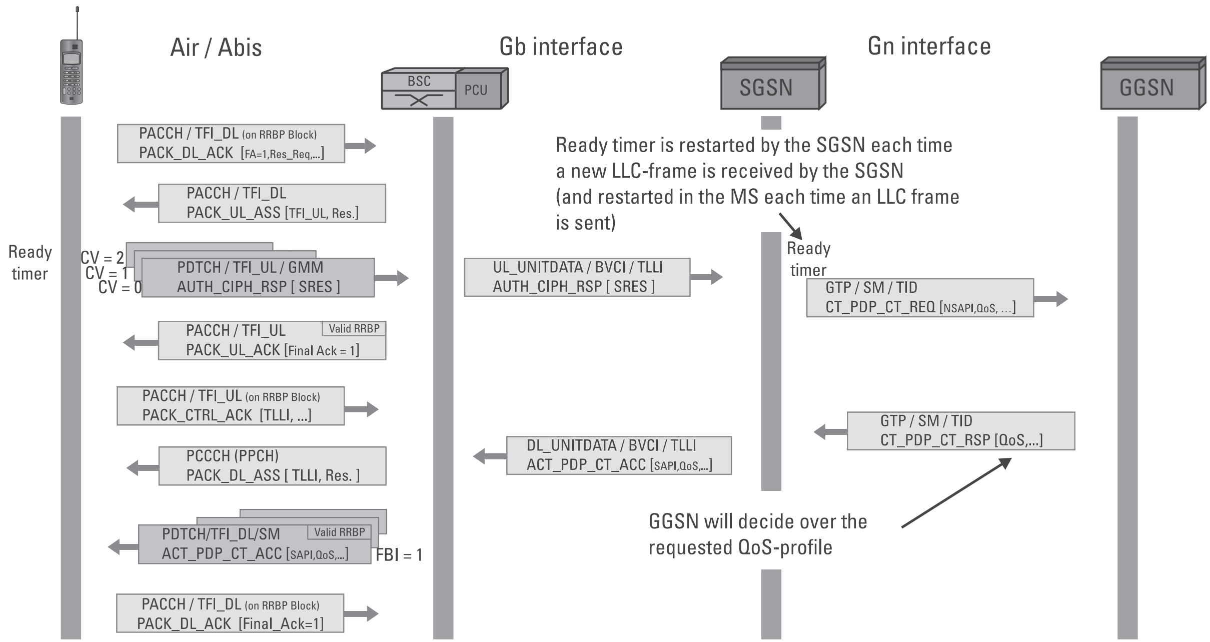


Figure 4.18 (continued).

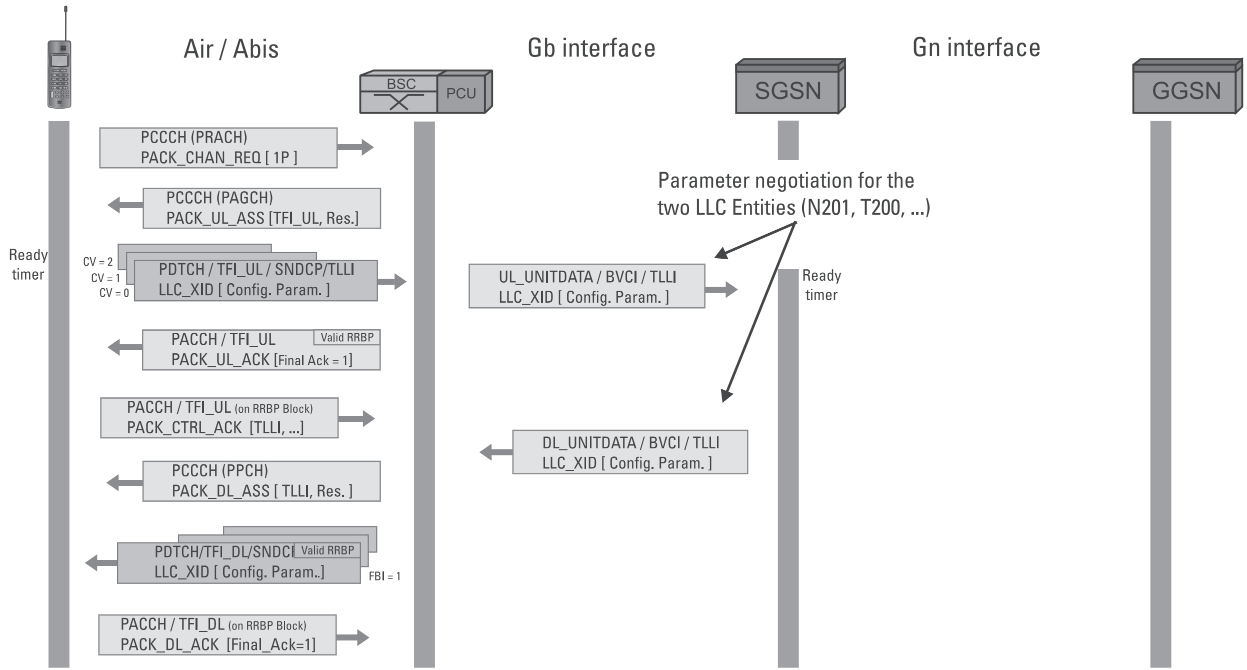


Figure 4.18 (continued).

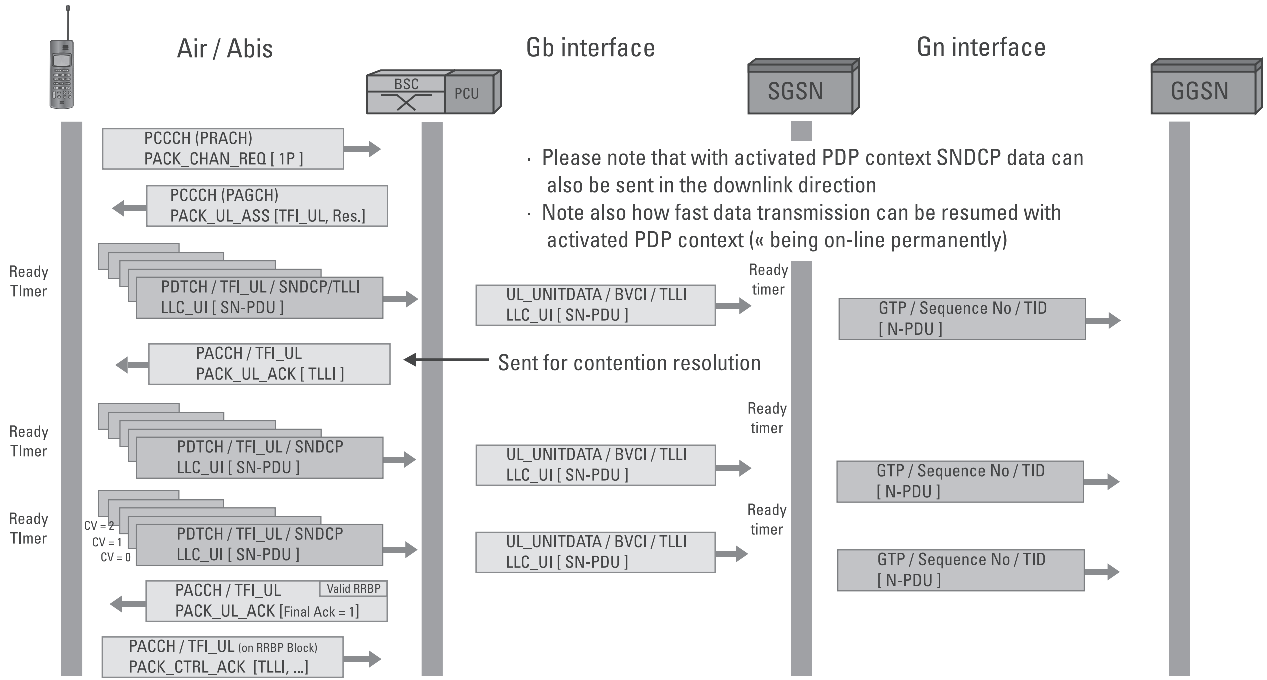


Figure 4.18 (continued).

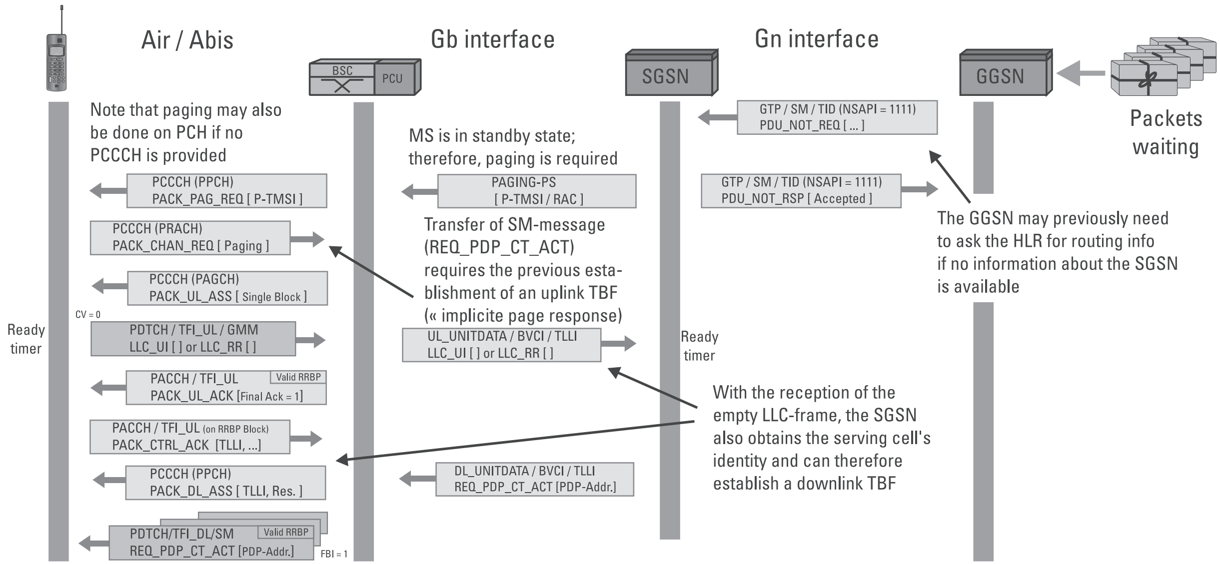


Figure 4.19 A mobile terminating PDP context activation via the PCCCH.

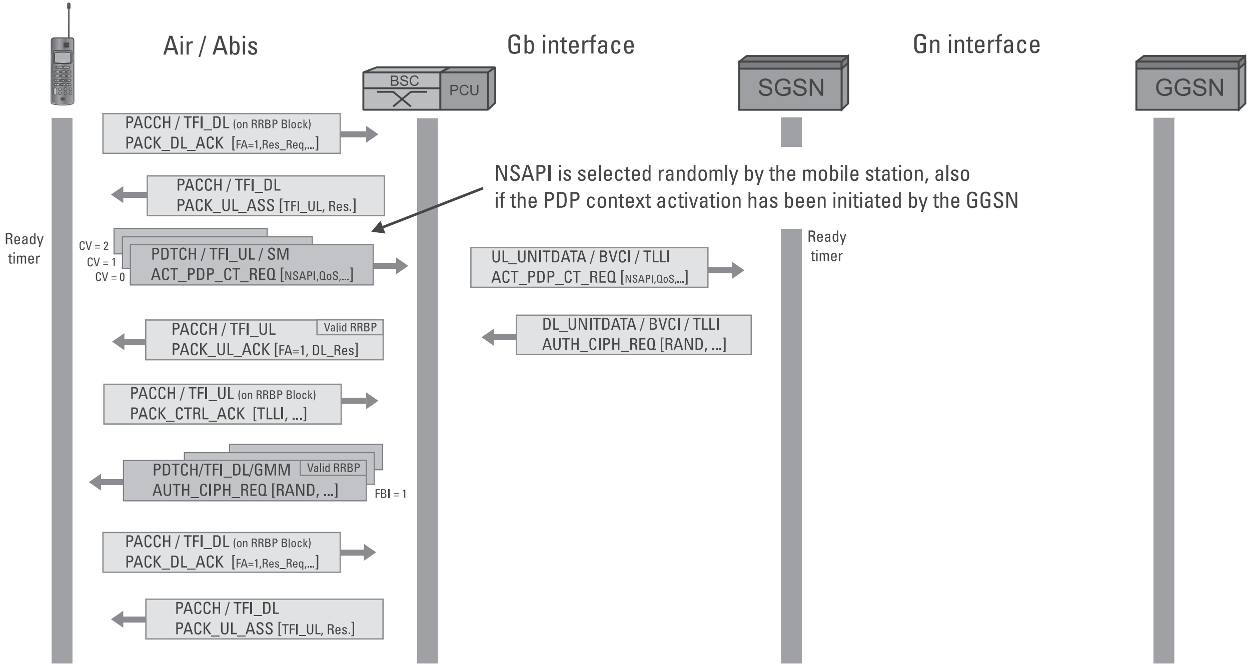


Figure 4.19 (continued).

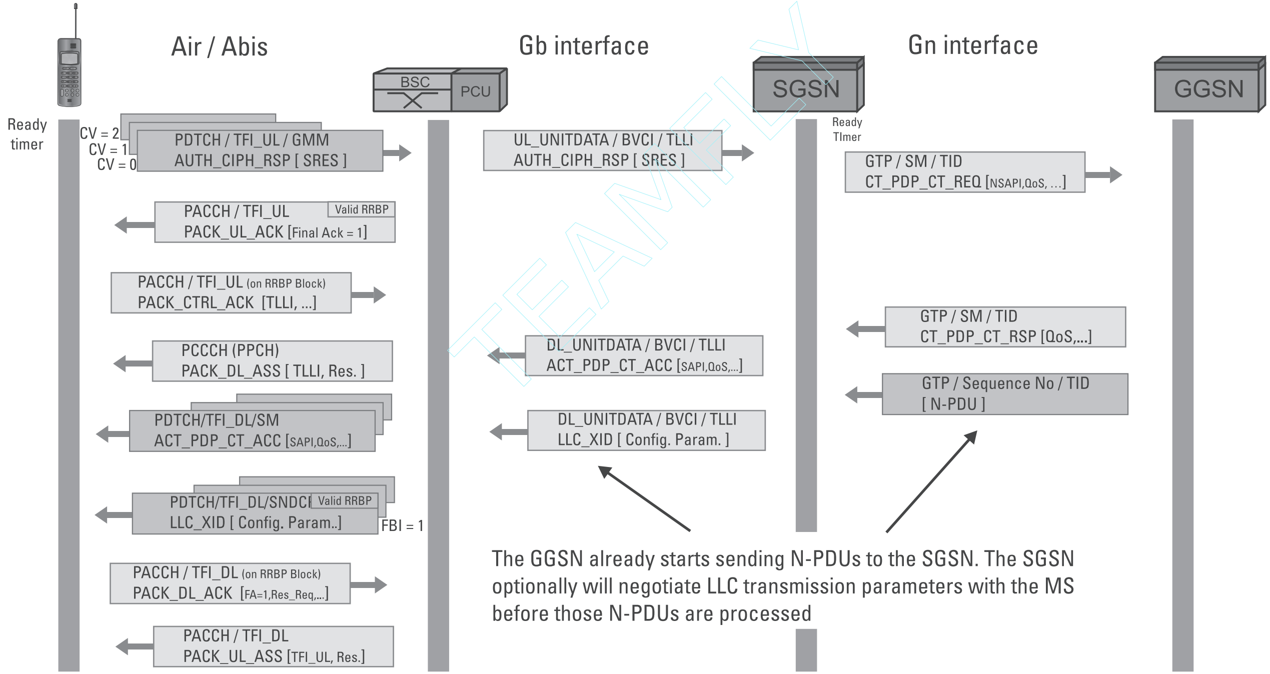


Figure 4.19 (continued).

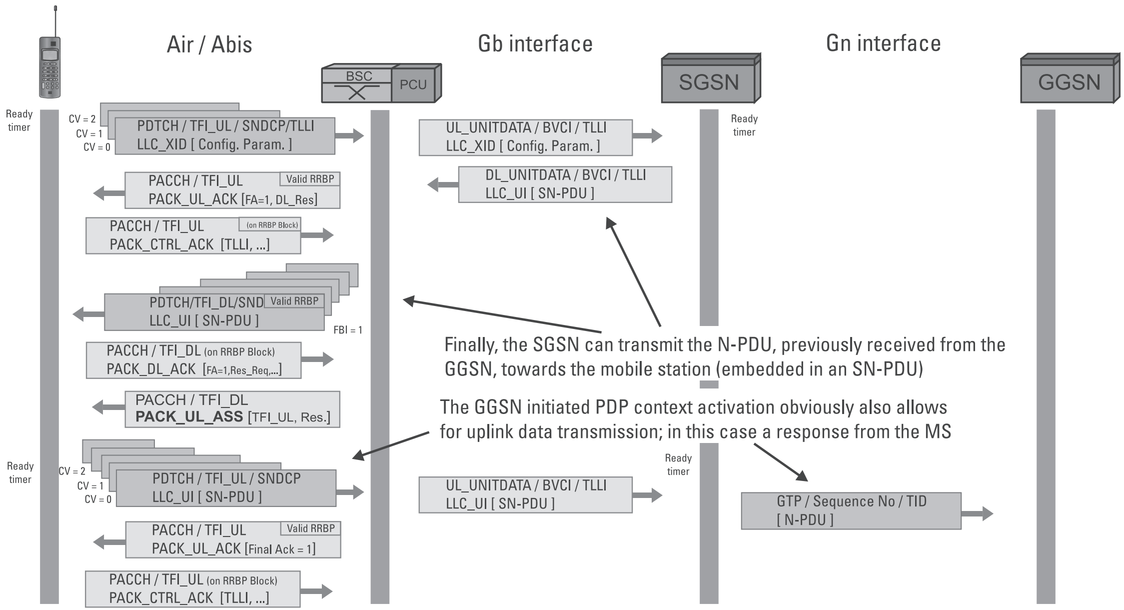


Figure 4.19 (continued).

5

The GPRS Protocol Stack

With its new capabilities and new network elements, GPRS requires new protocols. This applies to the interfaces between mobile station and BSS (PCU) and also to the GPRS core network. The GPRS protocol stack is shown in Figure 5.1. It is noticeable that there is no circuit switched MSC, and the BTS and BSC are presented completely transparently.

The following must also be noted with regard to Figure 5.1:

- Between the SGSN and the GGSN, or rather between all GPRS support nodes (GSNs), User Datagram Protocol (UDP) or Transaction Control Protocol (TCP) are shown as possibilities below the GPRS Tunneling Protocol. Both protocols are only possible with release 97 and 98. From release 99 onwards, only UDP is supported.
- In addition to SMS, the mobility management protocol (GMM) and the session management protocol (SM) are parallel to the Subnetwork Dependent Convergence Protocol (SNDCP). This SNDCP is the protocol that is used for actual data transmission (e.g., IP frames).
- All the functions discussed in Chapter 3, such as resource allocation mechanisms (fixed, dynamic, extended dynamic) belong to RLC/MAC in the protocol stack. We can then restrict ourselves in the following to the formalities of these protocols. The RLC/MAC and LLC protocols will be discussed in detail later.

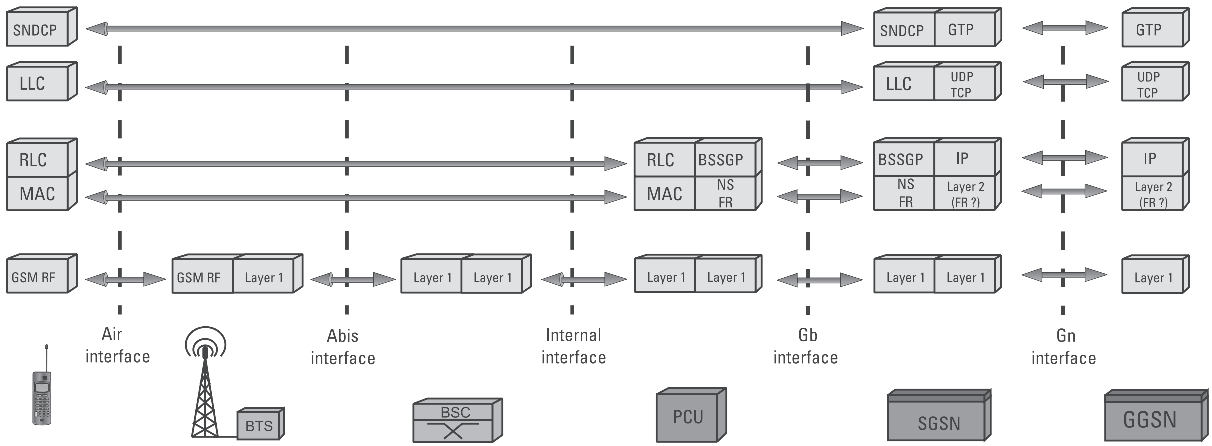


Figure 5.1 Overview of the GPRS protocol stack.

5.1 The RLC/MAC Protocol

The RLC/MAC protocol (Figure 5.2) is composed of two individual protocols, radio link control and medium access control. They are discussed here together because the GSM specification 04.60 also treats them both together. Both RLC and MAC belong to OSI layer 2.

The main functions of MAC are:

1. *Controlling network resource access (medium access):* This also includes the slotted Aloha access method on the RACH or PRACH.
2. *Distribution of available resources among several mobile stations (medium sharing):* The methods discussed in Chapter 3 for allocating resources to individual mobile stations, uplink and downlink, belong to this field.
3. *Controlling the release of network resources (medium release):* This includes the countdown procedure for releasing uplink resources and the methods carried out when releasing a downlink TBF. These were also discussed in Chapter 3.

The main functions of RLC are as follows:

1. *Setting up acknowledged and unacknowledged operating modes:* The operating mode actually required for a PDP context is determined by the QoS profile or, to be more precise, the reliability class parameter. In the unacknowledged operating mode, the transmitter does not expect any confirmation for data blocks transmitted and therefore does not have to buffer them. On the other hand, in the acknowledged operating mode, the transmitter, mobile station or PCU, expects a message of confirmation from the receiver after a

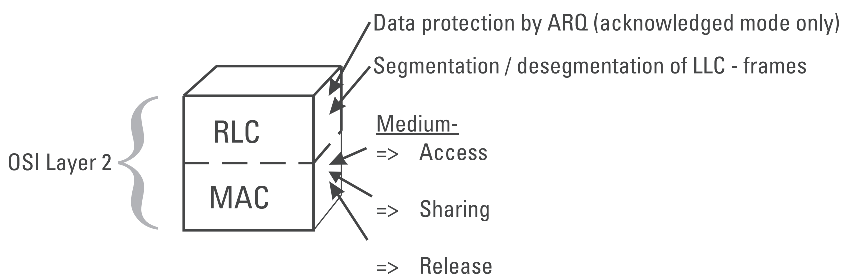


Figure 5.2 The functions of RAC and MAC.

certain number of RLC/MAC data blocks, which is a maximum of 64_{10} in GPRS. We will discuss the acknowledged operating mode in more detail later in the chapter.

2. *The segmenting of LLC frames into suitably sized RLC/MAC data blocks:* As illustrated in Figure 5.3, every LLC frame to be transmitted is divided into segments. Segment size depends on the coding scheme that is active at the given time.

5.1.1 The Acknowledged Mode in RLC—The ARQ Method

In the good old days of TCP, LAPD, and SS7, simple methods were used for BEC, as described in Section 1.6.3.1. There was one disadvantage of BEC that we did not mention in Chapter 1. In order to fill this gap, Figure 5.4 shows a modified version of Figure 1.22. The following refers to the procedures described in Figure 5.4.

1. First, the left-hand side transmits four consecutive data blocks, which are numbered 1 to 4, to the right-hand side. For unknown reasons, data block number 2 is erroneous, or is not received at all. The number of data blocks that can be transmitted consecutively and without confirmation depends on the preset or discussed window size. In the RLC/MAC protocol in GPRS, this window size is permanently set at 64_{10}.
2. At the receiving end, each individual data block is checked using the checksum field. The error in data block 2 is then noticed.
3. The right-hand side, the recipient of the data block, then sends a NACK message back to the left. This confirms data block 1 positively but shows that data block 2 was erroneous.
4. Since the NACK message in older protocols can only show the highest numbered data block that was received in perfect condition, the left-hand side must not only retransmit data block 2, but also all the following data blocks.

At first, such a method does not seem very efficient, and this impression is correct. Alternative methods, however, require every single data block to be identified separately. This would mean, using Figure 5.4 as our example, not only being able to request retransmission of data block 2, but also being able to confirm blocks 1, 3, and 4 positively. This is where automatic repeat request (ARQ) comes in.

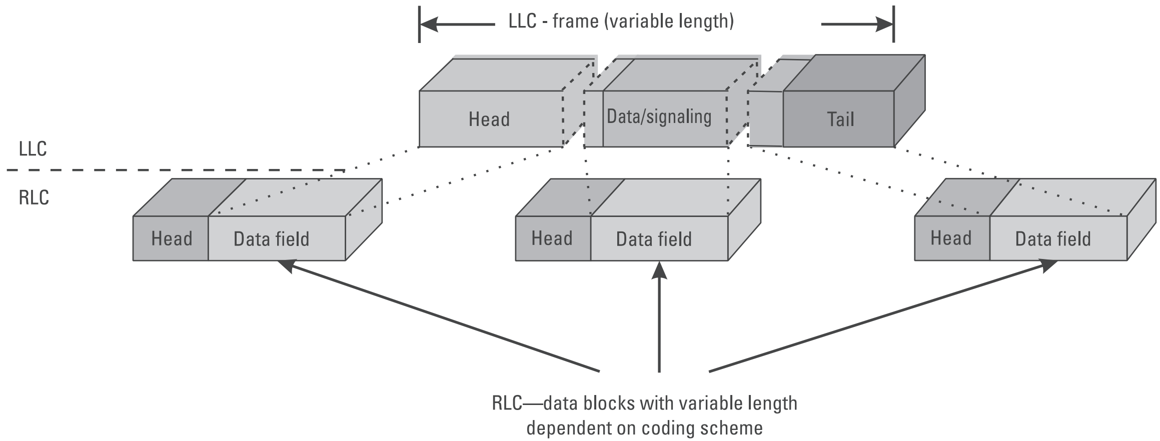


Figure 5.3 The segmenting of LLC frames by RLC.

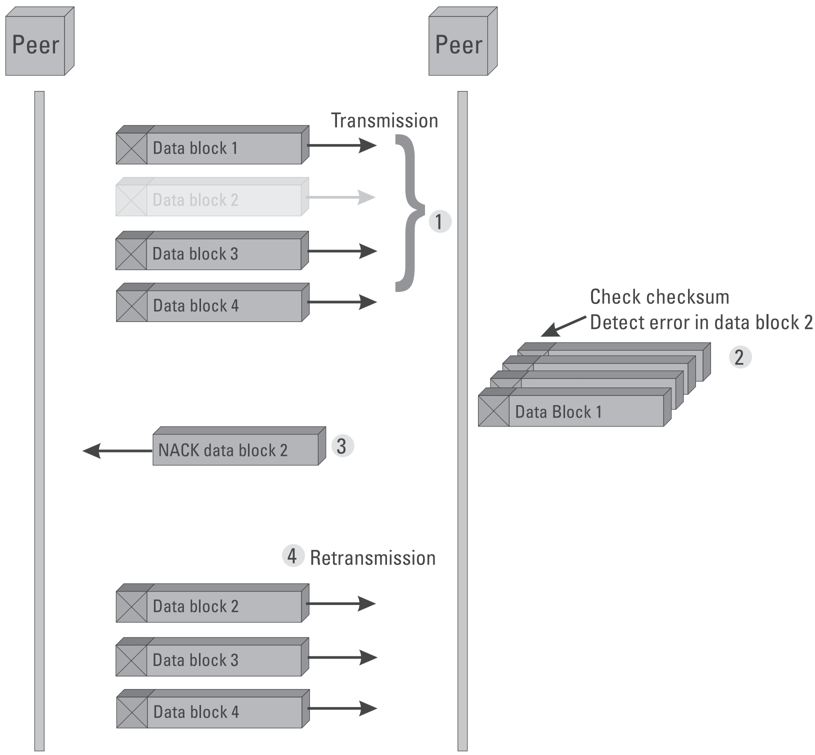


Figure 5.4 The disadvantages of BEC.

Every protocol that is to support ARQ has to be able to request retransmission of individual data blocks. Figure 5.5 shows an example of ARQ without any reference to a particular protocol.

In this case six consecutive data blocks are sent from the left to the right. Here, data blocks 3 and 5 are erroneous. Accordingly, the receiver sends the transmitter the previously mentioned NACK message. With ARQ, however, this NACK message contains a bitmap that can confirm each individual data block positively or negatively within the window size.

In our example, the bitmap displays a 1 at the bottom right for each perfectly received data block, while a 0 is displayed for each erroneous data block received.

Protocols with ARQ prepare NACK messages containing one bitmap each, in which each data block that has to be confirmed is represented by a single bit. In RLC/MAC, this is the PACK_UL_ACK message (PCU ⇒ MS) for confirming uplink data blocks and the PACK_DL_ACK message

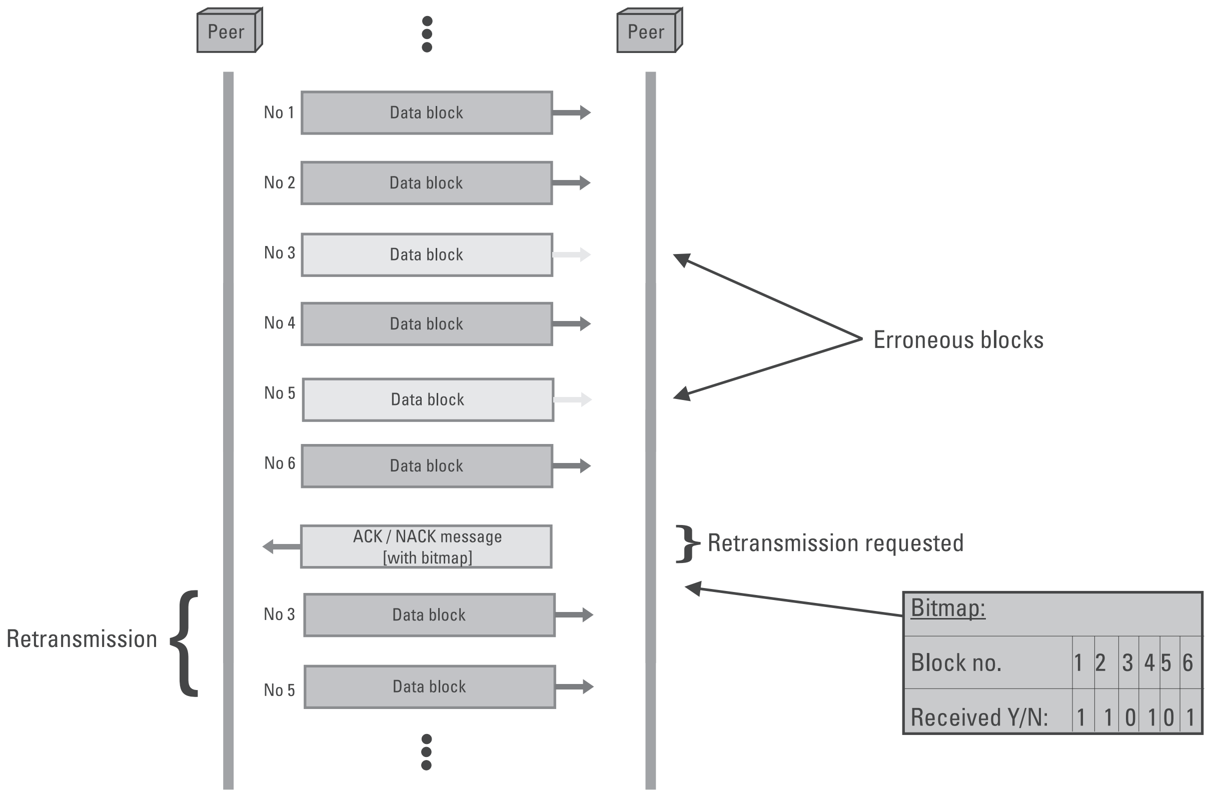


Figure 5.5 The procedures of BEC aided by ARQ.

(MS ⇒ PCU) for confirming downlink data blocks. An example of this is shown in Figure 5.6, where a PACK_UL_ACK message requests the retransmission of an erroneous block.

In GPRS, ARQ is not only used by RLC/MAC, but also by LLC.

5.1.2 The Frame Format of RLC/MAC

The RLC/MAC protocol is used in GPRS for both the transparent transfer of LLC frames and for the transmission of RLC/MAC-specific control information. Here, "transparent" means that RLC/MAC does not interpret these LLC frames, but forwards them on to LLC as soon as it receives them. For this reason, there are two different frame formats for control information and LLC frames.

```
+---------+-------------------------------------------+----------------------------------+
|BITMASK  |ID Name                                    |Comment or Value                  |
+---------+-------------------------------------------+----------------------------------+
|16:18:40,618,886  1 -1 down  PCU-DOWN  DATA SYNC  RMAC-DOWN  PUAN                       |
|GPRS PCU V2.0 Downlink (PCU-DOWN)  DATA SYNC (= Data with Syncronisation)               |
|Data with Syncronisation                                                                |
|--111--- |Uplink State Flag                          |7                                 |
|GPRS RLC/MAC V6.4.0 Downlink (RMAC-DOWN)  PUAN (= Packet Uplink Ack/Nack)               |
|Packet Uplink Ack/Nack                                                                  |
|10------ |Payload Type                               |RLC/MAC Control Block, with opt.octe|
|--00---- |Relative Reserved Block Period             |TDMA frame N+13                   |
|----0--- |Supplementary/Polling                      |RRBP field is not valid           |
|-----001 |Uplink State Flag                          |1                                 |
|0------- |Reduced Block Sequence Number              |0                                 |
|-00110-- |Radio Transaction Identifier               |6                                 |
|------1- |Final Segment                              |Current Block contains final segment|
|-------1 |Address Control                            |TFI/D octet is present            |
|00------ |Spare                                      |0                                 |
|--00000- |Temporary Flow Identifier                  |0                                 |
|-------0 |Direction                                  |TFI field ident uplink TBF        |
|001001-- |Message Type                               |9                                 |
|Core Parameter                                                                          |
|------00 |Page Mode                                  |Normal Paging                     |
|00------ |Bits                                       |00                                |
|--00000- |Uplink TFI                                 |0                                 |
|-------0 |Bit                                        |0                                 |
|01------ |Channel Coding Command                     |CS-2                              |
|1 Ack/Nack Description                                                                  |
|--0----- |Final Ack Indication                       |Retransmission requested, TBF incomp|
|***b7*** |Starting Sequence Number                   |1                                 |
|**b64*** |Received Block Bitmap                      |00 00 00 00 00 00 00 01           |
|1 (end of) Ack/Nack Description                                                         |
|--1----- |Bit                                        |1                                 |
|**b32*** |Contention Resolution TLLI                 |c1c00a45                          |
|---0---- |Bit                                        |0                                 |
|----0--- |Bit                                        |0                                 |
|-----0-- |Bit                                        |0                                 |
|------0- |Bit                                        |0                                 |
|-------0 |Spare Padding                              |0                                 |
+----+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+                                   |
|HEX |0 |1 |2 |3 |4 |5 |6 |7 |8 |9 |A |B |C |D |E |F |                                   |
+----+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+--+                                   |
|0   |01|00|89|7f|81|1b|00|24|00|40|40|00|00|00|00|00|                                   |
|10  |00|00|78|38|01|48|a0|2b|2b|2b|2b|02|e0|ff|ff|ff|                                   |
|20  |ff|ff|ff|ff|ff|ff|ff|ff|  |  |  |  |  |  |  |  |                                   |
+----------------------------------------------------------------------------------------+
```

Figure 5.6 Example of a PACK_UL_ACK message requesting retransmission of an erroneous data block.

There are two different frame formats for uplink and downlink, which means that a total of four frame formats are defined in RLC/MAC. Figures 5.7 to 5.10 show the RLC/MAC frame formats for GPRS release 1998. Please note that optional parameters are indicated with a dotted line. All the other fields are always available.

The individual parameters of the RLC/MAC frame formats will be discussed in Section 5.1.3. There is also a table of all RLC/MAC control messages in Table 5.1.

5.1.3 The Parameters in RLC/MAC Frames

5.1.3.1 The Uplink State Flag

The 3-bit USF is to be found in downlink data blocks and control messages. The function of the USF was described in detail in Sections 3.7.2.2 to 3.7.2.4.

5.1.3.2 The Retry Bit (R Bit)

The R bit is bit 0 in the MAC header of uplink control messages and uplink data blocks. When the R bit is set at 1, the mobile station has to transmit an access burst to the base station several times before resources are allocated. In this way an overload situation on the RACH or PRACH can be recognized. If several mobile stations have placed the R bit at the same time, the PCU must allocate more PRACHs.

The R bit remains constantly at 0 or 1 during an entire TBF.

5.1.3.3 The Stall Indicator Bit (SI Bit)

The SI bit indicates in uplink data blocks that the mobile station is requesting an acknowledgment even though the transmit window is not actually full yet (in GPRS, the RLC/MAC transmit window is 64 data blocks in size). Clearly, the SI bit is only significant in the RLC acknowledged mode.

5.1.3.4 The Relative Reserved Block Period Field (RRBP Field)

The function of the RRBP field is the allocation of an uplink radio block to a mobile station for the transmission of an uplink control message. Please refer to Section 3.8.1 for details. The interesting point to note here is that 2 bits are sufficient for the allocation of a given radio block; however, these 2 bits only indicate that a block is being allocated to the mobile station at a given distance from the current frame number.

Example. ($N + 13$), with N being the absolute frame number, allocates the third uplink radio block from the current position to the mobile station.

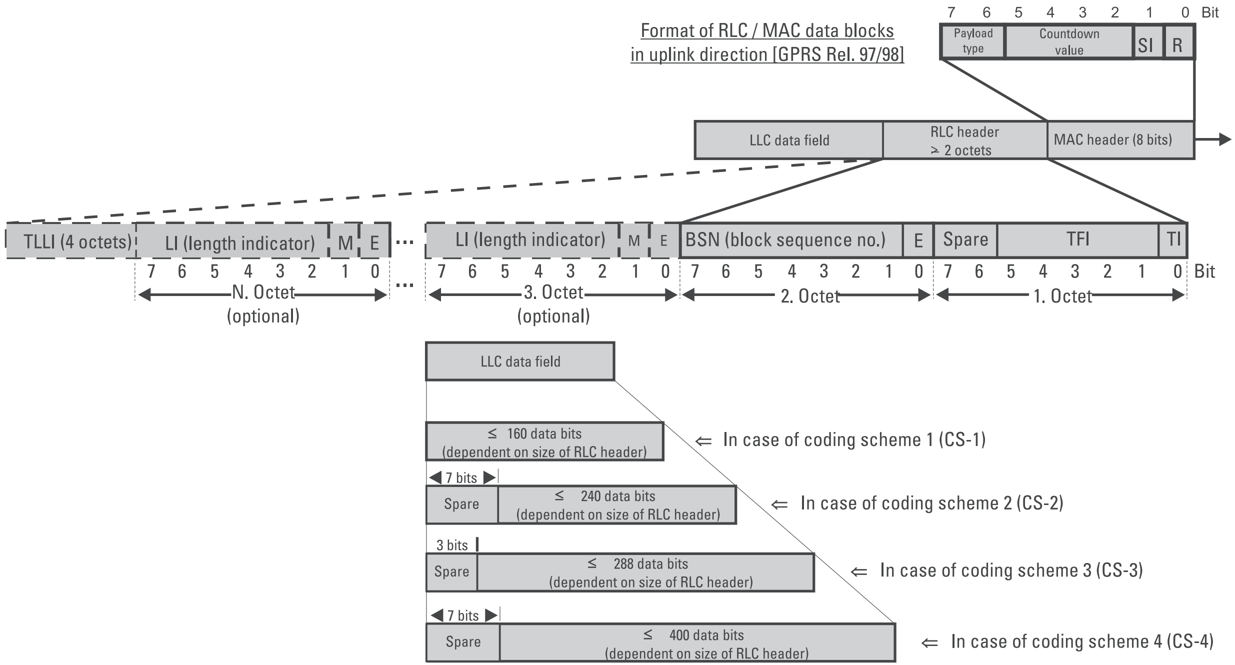


Figure 5.7 The format of the RLC/MAC frame for downlink transfer of data (LLC frames).

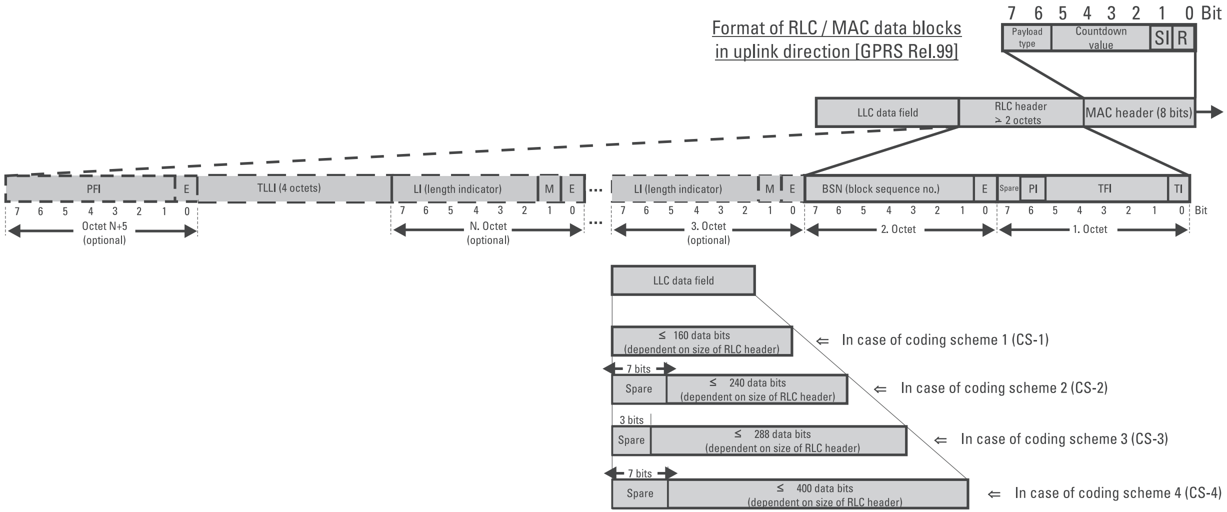


Figure 5.8 The format of the RLC/MAC frame for uplink transfer of data (LLC frames).

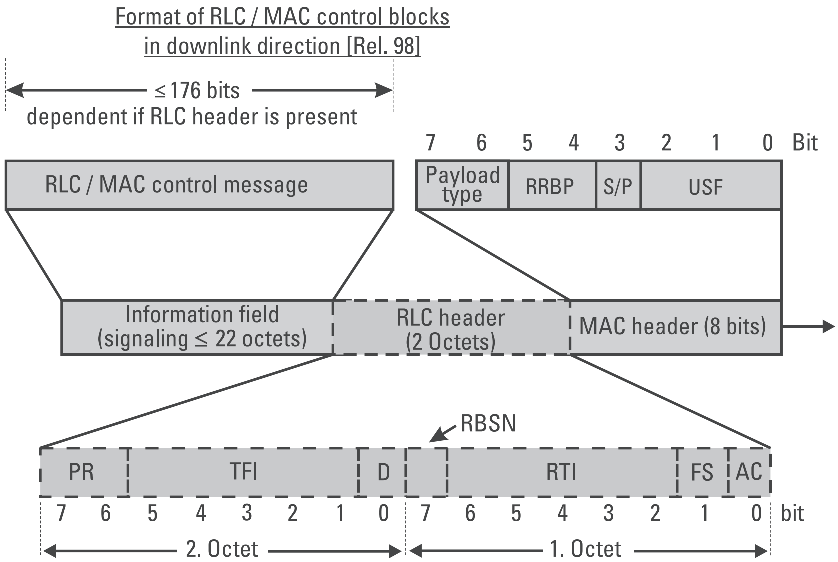


Figure 5.9 The format of the RLC/MAC frame for downlink transfer of RLC/MAC control messages.

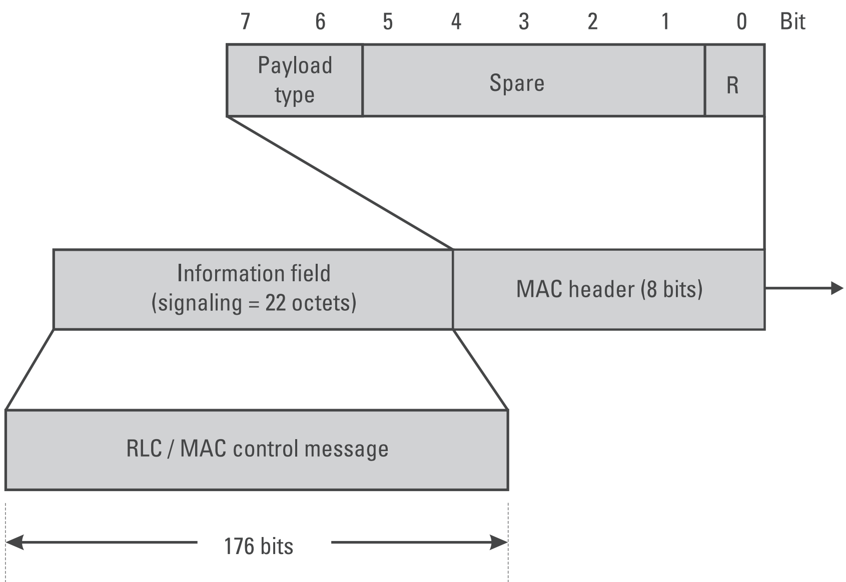


Figure 5.10 The format of the RLC/MAC frame for uplink transfer of RLC/MAC control messages.

Table 5.1
The RLC/MAC Control Messages

Message ID (bin)	Direction	Message Name
N/A	MS ⇒ PCU	PACKet CHANNel REQuest
000000	MS ⇒ PCU	PACKet CELL CHANGE FAILure
000001	MS ⇒ PCU	PACKet ConTRoL ACKnowledgement
000010	MS ⇒ PCU	PACKet DownLink ACKnowledgement
000011	MS ⇒ PCU	PACKet UpLink DUMMY ConTRoL BLocK
000100	MS ⇒ PCU	PACKet MEASurement REPort
000101	MS ⇒ PCU	PACKet RESource REQuest
000110	MS ⇒ PCU	PACKet MOBILE TBF STATus
000111	MS ⇒ PCU	PACKet PSI STATus
001000	MS ⇒ PCU	EGPRS PACKet DownLink ACKnowledgement
001001	MS ⇒ PCU	PACKet PAUSe
001010	MS ⇒ PCU	ADDitional MS RADIO ACCess CAPabilities
001010	MS ⇒ PCU	PACKet ENHanced MEASurement REPort
100001	PCU ⇒ MS	PACKet ACCess REJect
000001	PCU ⇒ MS	PACKet CELL CHANGE ORDER
000010	PCU ⇒ MS	PACKet DownLink ASSignment
000011	PCU ⇒ MS	PACKet MEASurement ORDER
100010	PCU ⇒ MS	PACKet PAGing REQuest
100011	PCU ⇒ MS	PACKet PDCH RELease
000100	PCU ⇒ MS	PACKet POLLing REQuest
000101	PCU ⇒ MS	PACKet PoWeR CONtrol/Timing Advance
100100	PCU ⇒ MS	PACKet PRACH PARAMeters
000110	PCU ⇒ MS	PACKet QUEUING NOTification
000111	PCU ⇒ MS	PACKet Time slot RECONFigure
001000	PCU ⇒ MS	PACKet TBF RELease
Message ID (bin)	**Direction**	**Message Name**
001001	PCU ⇒ MS	PACKet UpLink ACKnowledgement
001010	PCU ⇒ MS	PACKet UpLink ASSignment
100101	PCU ⇒ MS	PACKet DownLink DUMMY ConTRoL
110001	PCU ⇒ MS	PACKet SYStem INFOrmation 1
110010	PCU ⇒ MS	PACKet SYStem INFOrmation 2
110011	PCU ⇒ MS	PACKet SYStem INFOrmation 3
110100	PCU ⇒ MS	PACKet SYStem INFOrmation 3bis
110101	PCU ⇒ MS	PACKet SYStem INFOrmation 4
110110	PCU ⇒ MS	PACKet SYStem INFOrmation 5
110000	PCU ⇒ MS	PACKet SYStem INFOrmation 6
111000	PCU ⇒ MS	PACKet SYStem INFOrmation 7
111001	PCU ⇒ MS	PACKet SYStem INFOrmation 8
100111	PCU ⇒ MS	PACKet SYStem INFOrmation 13

The offset 13 accounts for the idle and/or PTCCH-TDMA frames situated in the 52 multiframe. This also applies to ($N + 17$) or ($N + 18$). Depending on the block number (0 . . . 11), both an idle frame and a PTCCH frame can be situated between the downlink allocation and the allocated uplink block.

5.1.3.5 The Supplementary/Polling Bit (S/P Bit)

The S/P bit in the MAC header of a downlink data block or a downlink control message indicates whether the RRBP field in this block/message is valid or not.

5.1.3.6 The Countdown Value Field (CV Field)

The 4-bit CV can only be found in the MAC header of uplink data blocks. During the countdown procedure, the CV indicates how many RLC/MAC data blocks are still waiting for transmission in the mobile station. When the countdown procedure has not yet started, the CV is set at 11112. More details on the countdown procedure can be found in Section 3.9.2.

5.1.3.7 The Payload Type Field

The 2-bit payload type field is a part of the MAC header in all four frame formats and indicates whether a block contains data or control information. In the case of a downlink control message, the payload type field also indicates whether the optional RLC header is available or not.

5.1.3.8 The Final Block Indicator Bit (FBI Bit)

The FBI bit is functionally a part of the MAC but is situated in downlink data blocks in the MAC header. As shown in Section 3.9.1, a set FBI bit (FBI = 1) indicates that the respective data block is the last data block in a downlink TBF.

5.1.3.9 The TLLI Indicator Bit (TI Bit)

The TI bit indicates, in uplink data blocks, whether the optional TLLI is contained in a data block or not. This is the case during the contention resolution procedure (see Section 3.6.1.1).

5.1.3.10 The Address Control Bit (AC Bit)

In the first octet of the optional RLC header of downlink control messages, the AC bit indicates whether or not this optional header is also contained in the second octet.

5.1.3.11 The Final Segment Bit (FS Bit)

In the optional RLC header of downlink control messages, the FS bit indicates whether this block is the last segment (FS = 1) of a segmented RLC/MAC control message.

5.1.3.12 The Radio Transaction Identifier Field (RTI Field)

In RLC/MAC, there can be several signaling procedures simultaneously active per PDCH. These are distinguished by the 5-bit RTI field, which is situated in the optional RLC header of downlink control messages.

5.1.3.13 The Direction Bit (D Bit)

In the optional RLC header of downlink control messages, the D bit indicates whether the TFI situated in this RLC header refers to an uplink TBF or a downlink TBF. This differentiation is necessary because identically numbered TBFs for different mobile stations are possible on each time slot in the uplink and downlink directions.

5.1.3.14 The Temporary Flow Identity Field (TFI Field)

The 5-bit TFI field is always situated in the RLC header of downlink and uplink data blocks, and it is optionally situated in the RLC header of downlink control messages. The TFI field identifies the TBF in each direction. There are more details on this in Section 3.5.1.

5.1.3.15 The Power Reduction Field (PR Field)

The 2-bit PR field is a component of downlink control messages and data blocks. The PR field signals how much weaker the respective block was transmitted in comparison to the BCCH. The PR field is only significant when the downlink performance regulation is activated.

5.1.3.16 The Reduced Block Sequence Number Bit (RBSN Bit)

The RBSN bit is a component of the optional header in downlink control messages. As shown in Figure 5.11 RBSN toggles the value of the maximum of two RLC/MAC control blocks that are possible for a downlink RLC/MAC control message between 0 and 1. The first segment is therefore always transmitted with RBSN = 0.

5.1.3.17 The Length Indicator Field (LI Field), the *E* Bit, and the *M* Bit

- The 6-bit LI field primarily serves delimitation of the segments of several LLC frames in the same RLC data block.

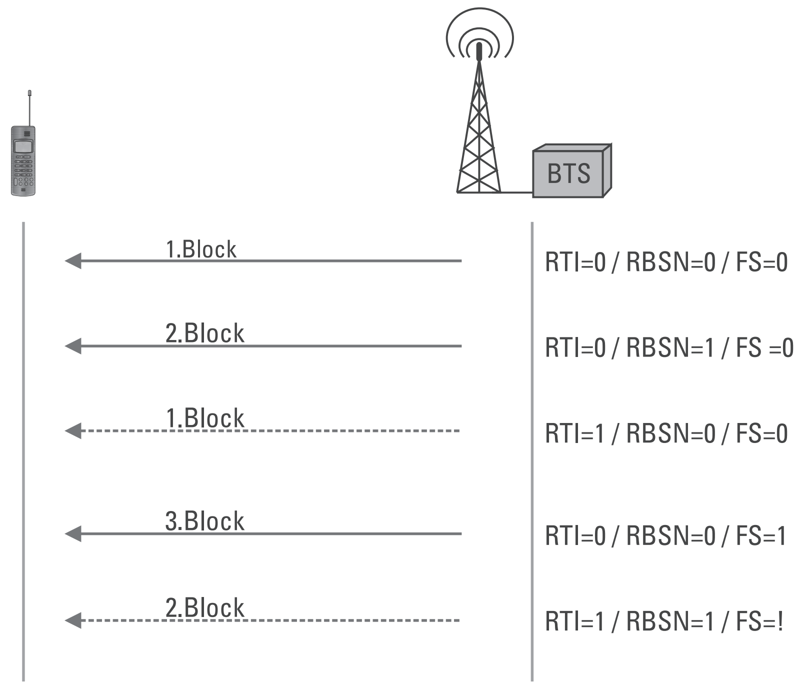


Figure 5.11 The function of RBSN in conjunction with the RTI field and FS bit.

- The basic rule is that the last segment of an LLC frame requires an LI field.
- A special case occurs when the last segment of the last LLC frame of a TBF would completely fill up the data field of an uplink and downlink data block but an additional RLC data block is also required for what is then the last octet of the LLC frame due to the addition of the LI field. In this case, the two final segments, and thus the two final RTC data blocks of the last LLC frame, contain an LI field. In the penultimate RLC data block, the LI field is set at 0, while it is set at 1 in the final RLC data block. In order to indicate this special case to the receiver, the penultimate RLC data block also has its *E* bit set at 1 and the *M* bit at 0.

The LI field can thus be present for two or three reasons and therefore is an optional information element in uplink and downlink data blocks.

The *E* (extension) and *M* (more) bits have already been addressed above. The *E* bit also exists in the mandatory part of the RLC header of uplink and downlink data blocks. If the *E* bit is set at 0, the presence of a third RLC header octet is indicated. If the *E* bit = 1, there is no further octet for the RLC header and the data segment starts immediately after this octet.

In the optional part of the RLC header, the field, the *E* bit and the *M* bit must be regarded together. The mutual significance of the field, the *E* bit, and the *M* bit is best explained with an example (Figure 5.12).

- The LI field no. X always shows the length of the Xth LLC segment in an RLC data block. If there are several LLC segments, there is always at least one LI field also present. However, one must consider that no LI field is indicated for the last LLC segment in an RLC data block if this LLC segment only contains the start of an LLC frame, and not an entire one. This case is valid for the "First segment of LLC-PDU 3" in Figure 5.12.
- The presence of an LI field for LLC-PDU 2 in Figure 5.12 results from the above-mentioned rule that the last LLC segment of an LLC frame requires a LI field. In the case of the LLC-PDU 2 in Figure 5.12, the entire LLC-PDU fits into one segment, this then being the last one.
- Notice the coding of the *E* bit and the *M* bit. In the first optional RLC header octet, the *E* bit = 0, which means that the next octet still contains RLC headers.
- *M* = 1 in the first optional RLC header octet, which indicates that an additional LLC-PDU (no. 2) and no filler octets is contained in this RLC data block.
- In the second optional RLC header octet, the *E* bit = 1, which means that no further header octet and data only is coming. Nonetheless, the *M* bit = 1 because the LLC-PDU 3 also begins in this RLC data block after the LLC-PDU 2.

The case presented is only intended as an example and to explain the function of *E* bit, *M* bit, and LI field.

5.2 The LLC Protocol

As shown in Figure 5.1 and Figure 5.13, the LLC protocol is defined between SGSN and mobile station. In the GSM standards LLC is classified under

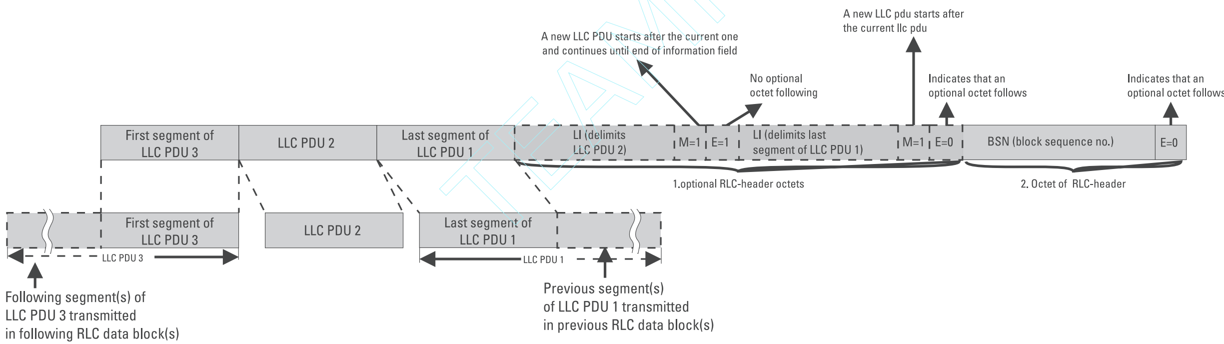


Figure 5.12 Example of the interaction of LI field, *E* bit, and *M* bit.

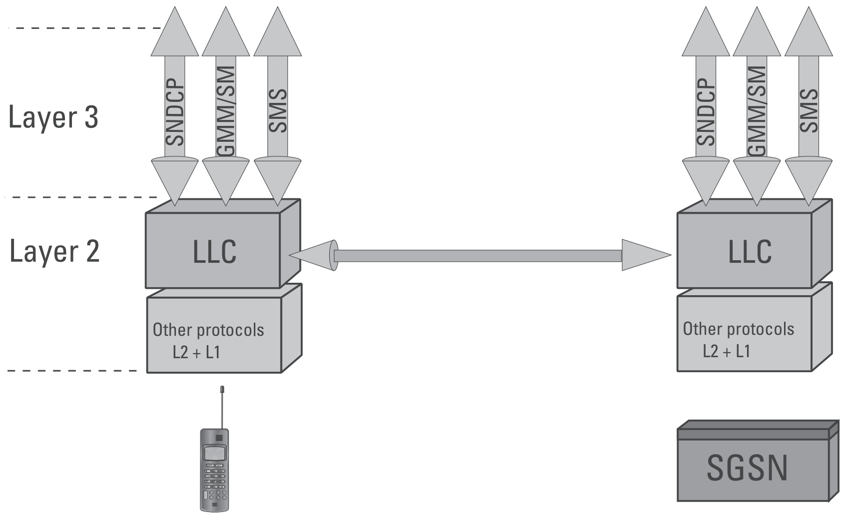


Figure 5.13 The LLC protocol between mobile station and SGSN.

OSI Layer 2, although the BEC mechanisms prepared by the LLC extend across several network nodes, which is more indicative of Layer 4.

In the mobile station, the LLC communicates with RLC/MAC, or rather uses RLC/MAC data blocks for the transport of LLC frames. In the SGSN, the BSSGP protocol is used as a relay between SGSN and PCU.

5.2.1 Functions of LLC

The functions of LLC in GPRS are introduced in the following sections.

5.2.1.1 Transfer of PDUs of Higher Layers

As Figure 5.13 shows, LLC transports the PDUs from all the higher layers. The users of LLC include the GMM and SM protocols; however, the transfer of short messages and PDUs of the various PDPs such as IP frames takes place embedded in LLC frames. The PDUs of the packet data protocols are, however, preprocessed in the SNDCP protocol and prepared for transfer via LLC. These PDUs are generally called SN-PDUs and carry, for instance, IP datagrams. The only LLC frame types for the transport of PDUs of higher layers are the unconfirmed information (UI) and information and supervisory (I + S) frames, which we shall discuss in detail in the following section.

Note that all the higher layers, SMS, SM/GMM and SNDCP, forward their PDUs to LLC in such a way that one LLC frame is sufficient for the transfer of such a PDP in each case (Figure 5.14). The data field of LLC frames must therefore have a variable length that must not exceed a maximum value (N201 = 1,520 octets). The actual value of N201 to be used can be discussed by SGSN and mobile station before a data transfer, but must not exceed the stated 1,520 octets.

5.2.1.2 Transmission in Asynchronous Disconnected Mode and Asynchronous Balanced Mode

The asynchronous disconnected mode (ADM) and the asynchronous balanced mode (ABM) are known from ISDN and APD. The functioning of both modes of transmission is illustrated in Figures 5.15 and 5.16, respectively.

What is decisive is that in ABM a virtual connection is established between SGSN and the mobile station before the actual transmission of information. As in LAPD, the ABM also serves transmission in acknowledged mode in LLC. For this reason, numbered I+S frames are used for data transmission, each of which must be confirmed by the other side. Note that in LLC, or rather in GPRS, this acknowledged mode (ABM) is only supported for the transfer of payload, but may not be used for the transmission of short messages or GMM/SM PDUs.

In contrast, the unacknowledged mode (ADM) can also be used for the transfer of payload but is particularly prescribed for the transmission of GMM/SM PDUs and short messages.

The question remains as to who decides whether ADM or ABM should be used for the transmission of payload and when it should be used. This is actually a question of the QoS profile or, to be more precise, the reliability class parameter, which was described in Section 4.2.2.3. If you refer back to earlier chapters, you will see that "LLC Frame Mode Acknowledged" is only possible for reliability classes 1 and 2.

In both ADM and ABM there is an optional agreement between SGSN and mobile station regarding the transmission parameters to be used via exchange identification (XID) frames. This includes the previously mentioned

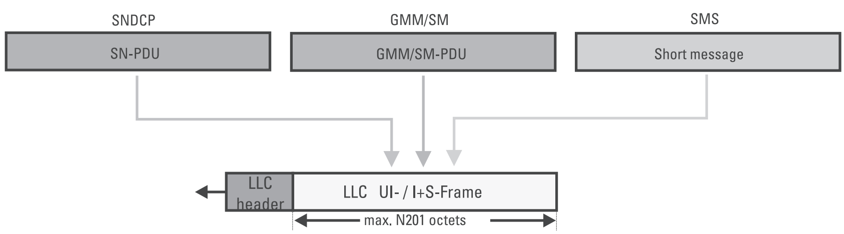


Figure 5.14 The higher layers supply LLC with preprocessed PDUs.

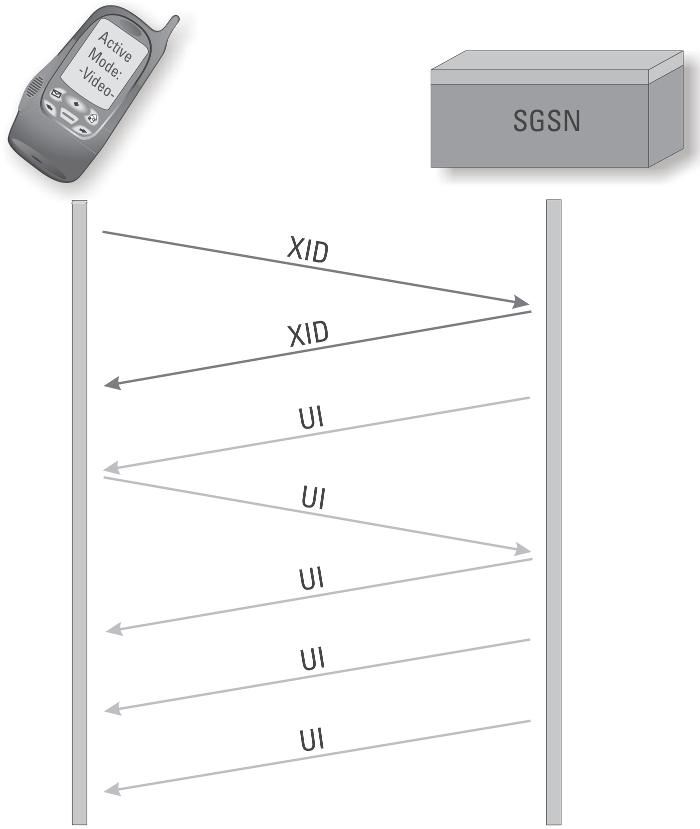


Figure 5.15 The ADM.

N201 (maximum number of octets in the information fields of I + S and UI frames) and also includes agreement on the offset variable for the encryption of I + S frames (see Figure 5.17), the counter N200 (maximum number of retransmissions) and other parameters.

5.2.1.3 Data Encryption (Ciphering)

The range of encryption in GPRS is considerably higher than that in GSM. This is because in GSM, ciphering is only activated between mobile station and BTS, but all data and speech on the terrestrial interfaces remain unencrypted. As illustrated in Figure 5.18, all LLC frames between the SGSN and the mobile station can be selected to be transmitted encrypted in GPRS.

The process of encryption itself is illustrated in Figure 5.17. At present only the GPRS encryption algorithm 1 (GEA/1) is available. The Kc value

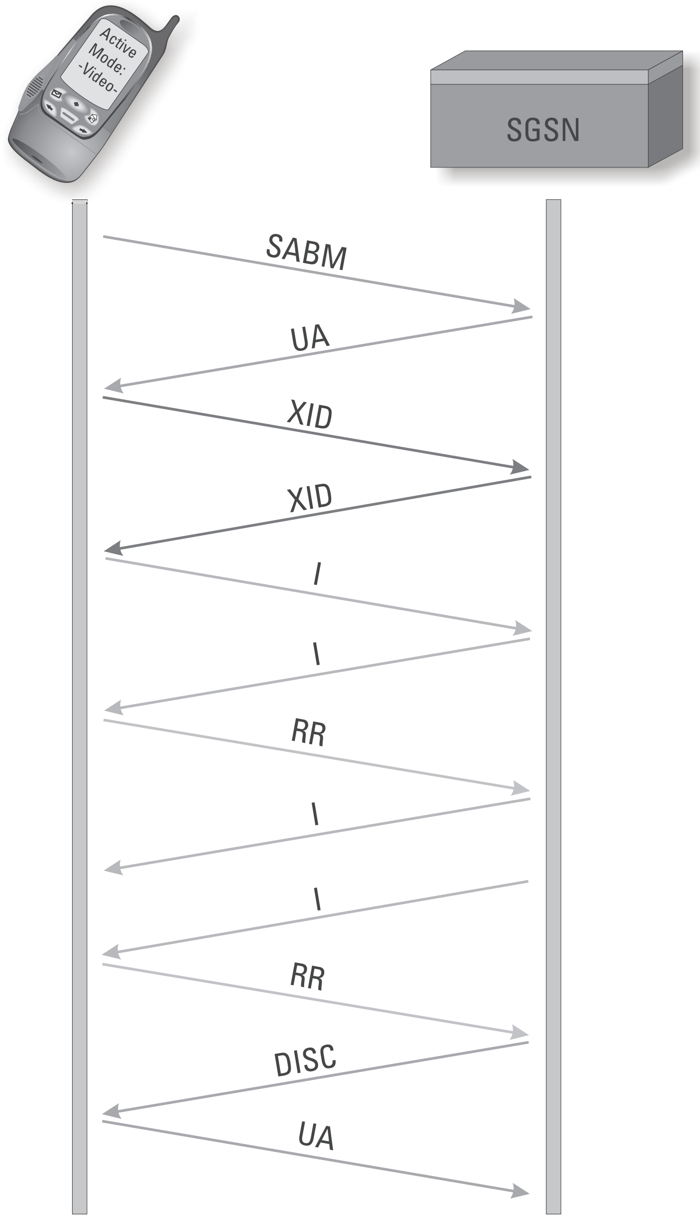


Figure 5.16 The ABM.

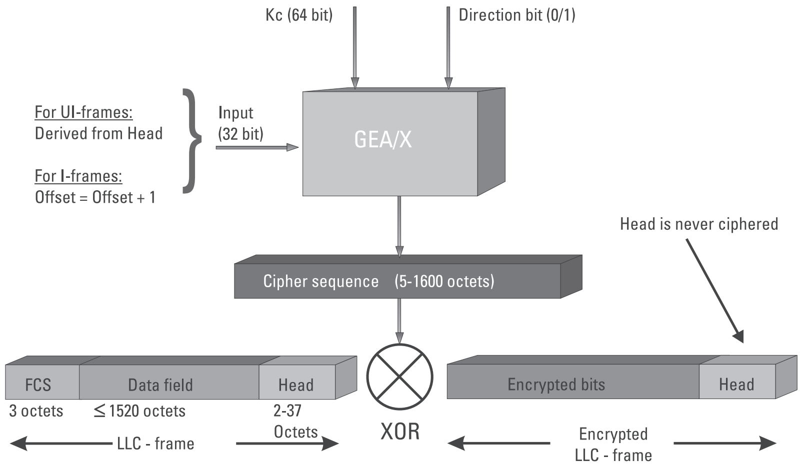


Figure 5.17 The process of encryption in GPRS/LLC.

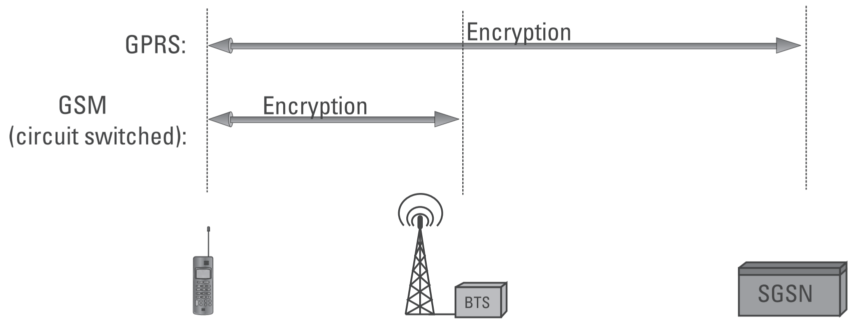


Figure 5.18 The range of encryption in GSM and GPRS.

indicated is the Kc already known from GSM, with the restrictions as described in Section 4.1.2.1.

The offset variable is either derived using a formula (GSM 04.64) from the header of the UI frame or is determined for I + S frames in advance in the XID frame discussion by the SGSN.

5.2.2 The Frame Format of LLC

The frame format of LLC is shown in Figure 5.19. For a detailed description of the functions of the individual frame types, refer to my book on GSM or the CD-ROM, *GPRS, EDGE, HSCSD and the Path to 3G* (Artech House, 2000).

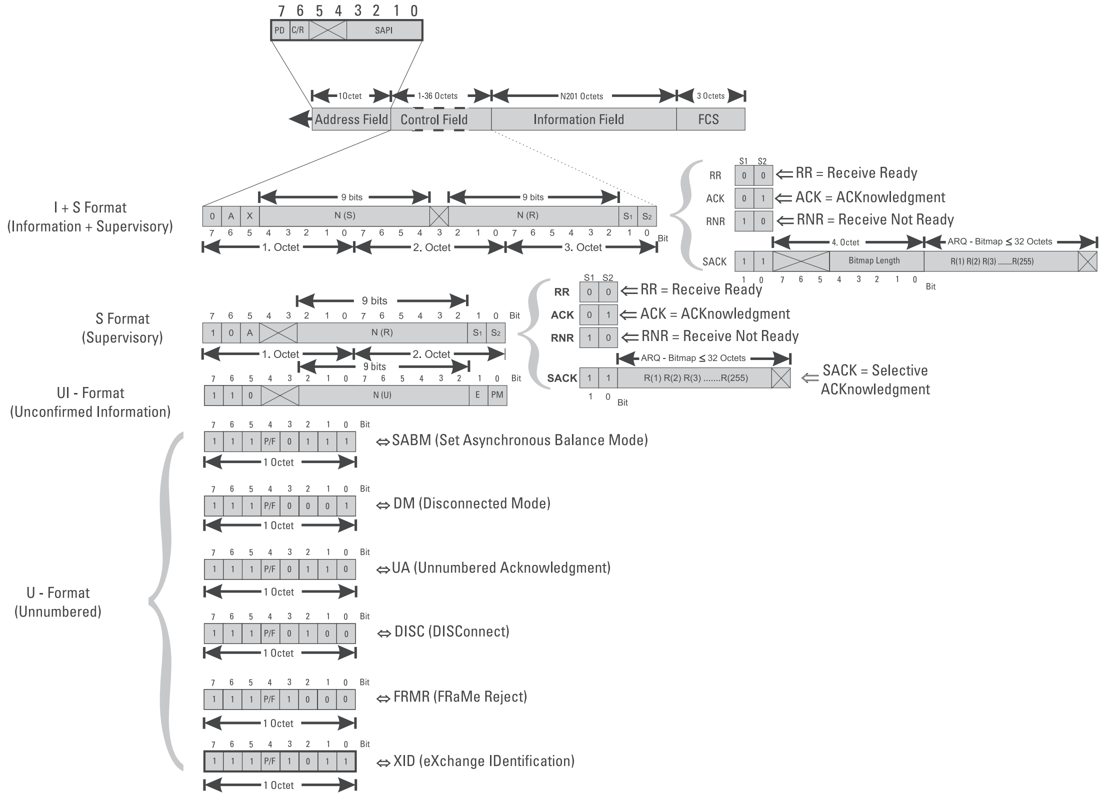


Figure 5.19 The LLC frame format.

6

Introduction to HSCSD

6.1 Overview

High-speed circuit switched data expands the 4.8-Kbps, 9.6-Kbps, and 14.4-Kbps GSM data channels to multislot configurations by means of channel bundling (Figure 6.1). Up to six time slots of the same type can be bundled on a TDMA frame. However, the maximum synchronous data rate is limited to 64 Kbps. Higher data rates, such as 86.4 Kbps with 6 × 14.4 Kbps, are not possible. This restriction is a result of interworking with ISDN and the fact that the terrestrial interfaces of the PLMN, such as the A interface, are based on ISDN technology (i.e., PDH architecture with 64-Kbps channel structure). Even with interworking with the PSTN and V.90 modems no more than 57.6 Kbps is possible in the downlink and 33.4 Kbps in the uplink.

With transparent data service and asynchronous transmission, the maximum data rate is 38.4 Kbps which is enabled by using 3 × 14.4 Kbps or 4 × 9.6 Kbps. In practice, however, many operators prefer the nontransparent data service and offer 4 × 14.4 Kbps or 4 × 9.6 Kbps as the largest multislot configuration. The nontransparent data service is based on the Radio Link Protocol (RLP). RLP is a Layer 2 protocol in which data is saved (data link layer) between the mobile station and the IWF on the fixed net junction. There is also the possibility of data compression with V.42bis. Here, the data rate can be increased further with optimum wireless and line conditions. However, when the network burden is heavier, it will be difficult to have the necessary resources for four free time slots on a carrier with a maximum

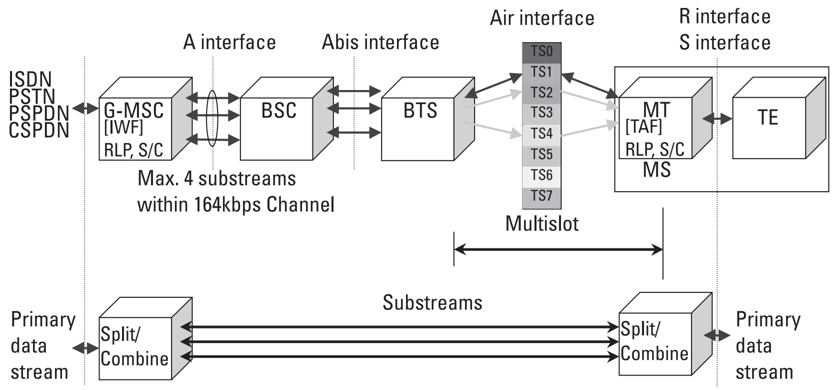


Figure 6.1 Channel bundling in the GSM PLMN.

of eight time slots, or a maximum of six in the case of the noncombined BCCH, constantly available and to guarantee mobility with high reliability (handover). Mobile stations available (at the beginning of 2001) can bundle a maximum of four channels and therefore offer 57.6 Kbps (nontransparent 4 × 14.4 Kbps, without V.42bis) at best. For this reason, we will frequently restrict ourselves to describing configurations with a maximum of four channels. In practice, voice bearers will have resource preemption over HSCSD and with GPRS supported in parallel, the priority dilemma is perfect and gives one more reason why only a few infrastructure vendors have chosen to implement HSCSD.

The familiar ISDN channel bundling using the MLPP protocol (RFC 1717) is not directly supported by the GSM PLMN and is not used to offer the above-mentioned 6 × 14.4 Kbps. Apart from the principle of channel bundling, these two applications are independent from one another. Outside the PLMN, the multislot configuration is not visible and, at the junction to other networks (exPLMN), is always limited to a single channel, which is not bundled and has a maximum of 64 Kbps. This does not mean, however, that MLPP is impossible as an end-to-end solution.

6.1.1 Asymmetrical Connections

What is new in HSCSD is asymmetrical connections within the GSM PLMN. They provide a higher data rate in both the downlink and uplink and therefore take into account characteristics, which are specific to applications such as Web browsing and downloading e-mail. One considerable advantage is the conservation of battery energy in the mobile station. Since RF power is emitted on fewer time slots in the uplink and within the RF amplifier, which remains off during the idle slots, less power is consumed overall. Idle slots give the mobile station more time to monitor neighboring cells, without extra technological effort and thus avoiding increased costs (e.g., for a further receiver or synthesizer/VCO). The opposite configuration, with a higher data rate in the uplink than in the downlink, is not foreseen; typical applications do not seem to call for this kind of service. Asymmetrical connections are only supported together with the nontransparent service. The reason for this is the handling of the unused uplink resources on the terrestrial interfaces between the base station and the junction in which the data flow is combined again (combine function). The unused resources are wasted in any case because they cannot be used for any other purpose. Presumably, the subscriber will bear the costs of this and will not be able to profit financially from the asymmetry.

6.1.2 Transparent and Nontransparent Connections

In their functions and service features, the transparent and nontransparent connections are identical to those for the line transmitted data services (GSM phase 2), which are not bundled. The essential feature of the nontransparent connections is saving data using the ARQ method and optional V.42bis data compression. However, the multislot configuration is limited to four subchannels and, without data compression, offers 4×14.4 Kbps at best (14.4 Kbps are transmitted in 16-Kbps channels on the terrestrial interfaces, and four of those occupy a 64-Kbps channel). The RLP has been expanded to version 2 especially for HSCSD, and therefore supports channel bundling (numbering of subchannels).

Transparent connections do not have data-saving capacities and thus have a variable bit error rate that depends greatly on the interface channel conditions. Transmission quality decreases particularly in switching procedures during handover. Advantages in comparison with the nontransparent service are the shorter delay and the more constant data rate. Note here that the delay due to interleaving after channel coding always contributed heavily to the overall delay, both for transparent and nontransparent services (size approximately 100 ms, 22 interleaving = 22×4.615 ms); see also Section 1.6.4.

A radio link adaptation (RLC) as offered by GPRS does not exist in HSCSD, which means that it is not possible to adapt the coding on the air interface to the conditions of the wireless channel and change it dynamically.

6.1.3 Types and Classes of the Mobile Stations

Mobile stations are divided into types and classes. There are two types of device that differ in their ability to transmit and receive simultaneously. Type 1 devices cannot transmit and receive simultaneously, as opposed to type 2, which can. This class determines how the time slots of a TDMA frame can be combined to form a multislot configuration.

6.1.4 Data Rates

With HSCSD, the general bearer services (GBS) were introduced. These offer the following:

1. Synchronous payload channels, bearer service #30:
   - Transparent, up to 64 Kbps with 5 × 14.4 Kbps or the realistic limitation to four subchannels: 56 Kbps with 4 × 14.4 Kbps, and 38.4 Kbps with 4 × 9.6 Kbps;
   - Nontransparent, up to 57.6 Kbps with 4 × 14.4 Kbps, or 38.4 Kbps with 4 × 9.6 Kbps.
2. Asynchronous payload channels, bearer service #20:
   - Transparent up to 38.4 Kbps with 4 × 9.6 Kbps, or 3 × 14.4 Kbps;
   - Nontransparent up to 57.6 Kbps with 4 × 14.4 Kbps, or 38.4 Kbps with 4 × 9.6 Kbps.

There are three parameters to be differentiated:

1. *Fixed network user rate (FNUR):* Data rates at the fixed network junction. By using analog modems (e.g., at the junction to the PSTN), these are mostly integer multiples of 600 bps.
2. *Air interface user rate (AIUR):* Not all the FNUR data rates are supported in the GSM PLMN. This is the reason for the existence of the AIUR, which is based on the channel coding of the interface:
   - TCH/F4.8;
   - TCH/F9.6;
   - TCH/F14.4.

   Together with various rate adjustments, almost all the FNUR rates can be realized as AIUR data rates. Please note the following exceptions:

- 7.2 Kbps;
- 12 Kbps;
- 24 Kbps.

  In transparent connections, the FNUR and AIUR are always identical; in nontransparent connections they can differ. When, for instance, the V.42bis service is only used in the GSM PLMN but not at the fixed network junction, the FNUR must be greater than the AIUR.

3. *Wanted air interface user rate (WAIUR):* The WAIUR controls asymmetrical connections (nontransparent data service). If it is greater than the mobile station's maximum possible symmetrical AIUR, the network will try to establish an asymmetrical connection. The decisive factors are the characteristics of the mobile station itself, or the mobile station characteristics set by the subscriber that are communicated to the network when the connection is established (see Section 6.2.3). These include:
   - mTCH—maximum number of TCH/F channels for channel bundling;
   - Acceptable channel codings (ACC)—one or more of the three above-mentioned full-rate channels (TCH/F4.8, 9.6, and 14.4).

Table 6.1 shows data rates for both FNUR and AIUR. Note that with the nontransparent service and interworking to the PSTN, higher asynchronous data rates than 38.4 Kbps are possible if V.90 modems are used. This is used in the example in Section 6.3.

6.2 The Essential Innovations in HSCSD

6.2.1 Split/Combine Function

From the transmitter side, the division of the primary datastream takes place using the split function into k secondary substreams with the same data rate; we will refer to these as subchannels in the following. On the air interface, each subchannel is transmitted in a full-rate traffic channel of the same type (e.g., TCH/F9.6) and then recombined into a primary datastream by the receiver using the combine function. The subchannels are numbered from 0 to $k - 1$. Allocation to the time slots on the air interface can be arbitrary. However, in the first channel allocation, the assignment must take place

Table 6.1
FNUR/AIUR Data Rates

FNUR (Kbps)	AIUR (Kbps)	Transparent	Nontransparent
9.6	9.6	Asynchronous/ synchronous	Asynchronous/ synchronous
14.4	14.4	Asynchronous/ synchronous	Asynchronous/ synchronous
14.4	19.2	—	Asynchronous/ synchronous
19.2	19.2	Asynchronous/ synchronous	Asynchronous/ synchronous
28.8	28.8	Asynchronous/ synchronous	Asynchronous/ synchronous
38.4	38.4	Asynchronous/ synchronous	Asynchronous/ synchronous
43.2	43.2	—	Synchronous
48.0	48.0	Synchronous	Synchronous
56.0	56.0	Synchronous	Synchronous
57.6	57.6	—	Synchronous
64.0	64.0	Synchronous	—

from the lowest subchannel and the lowest time slot progressively up to the highest subchannel and highest time slot (e.g., 0 ⇒ TS1, 1 ⇒ TS2, 2 ⇒ TS4). Deviations can occur when the network carries out a reconfiguration (network initiated modification) due to the resource situation, such as during a handover when a 3 × 9.6-Kbps configuration becomes a 2 × 14.4-Kbps configuration, without changing the primary data rate of 28.8 Kbps. This happens provided that the mobile station has indicated that it supports both channel types (TCH/F9.6 and TCH/F14.4) while establishing the connection. In nontransparent connections and as long as it is agreed on in the establishment of the connection, the mobile station can also request a change of configuration in order to increase or decrease the data rate [user initiated modification indication (UIMI)]. If the number of subchannels is increased, the part of the configuration that is not changed is taken over directly and the part that is added is allocated according to the same rule as in the first allocation.

On the network side, the split/combine function is usually found in the gateway MSC (G-MSC), particularly in the IWF; however, it can be found elsewhere. The same principle applies here as with the transcoder: there is more than one possible place in the PLMN. For the maximum data rate of 64 Kbps, the place should really be in the BTS (see Section 6.4.3 for further details). On the mobile side, the split/combine function is a part of the terminal adapter function (TAF; see GSM 07.01/07.02). This means that the subchannels exist between the mobile station and the place in the PLMN in which the split/combine function is situated. Outside the PLMN, channel bundling is neither recognizable nor controllable.

First we will describe the split/combine function in general. Then we will look at the application in the various service features and channel types with the peculiarities that arise from the use of different frame sizes and formats.

The functional differences of the split/combine function are as follows:

1. Transparent service with TCH/F9.6 or TCH/F4.8 on the basis of the 80-bit V.110 frame;
2. Transparent service with TCH/F14.4 and the 290-bit frame of the air interface;
3. Nontransparent service with TCH/F9.6 or TCH/F4.8 and 240-bit RLP frame;
4. Nontransparent service with TCH/F14.4 and 576-bit RLP frame.

6.2.1.1 Split Function

The frames of the primary datastream at the split function input (n, $n + 1$, $n + 2$, . . . ; see Section 6.2) are distributed cyclically into the number of subchannels k, starting with subchannel 0, ascending to the highest channel, $k - 1$: $n \Rightarrow 0$, $(n + 1) \Rightarrow 1$, $(n + 2) \Rightarrow 2$, . . . , $(n + k - 1) \Rightarrow k - 1$. The cycle begins again, with the frame $n + k$ in the subchannel 0: $(n + 4) \Rightarrow 0$, $(n + 5) \Rightarrow 1$, . . . , $(n + 2 * (k - 1)) \Rightarrow k - 1$.

The split function, however, is not able to distribute the primary datastream into subchannels of different bandwidths—that is, all subchannels must have the same data rate and the same frame type, both on the air interface (TCH/F4.8, TCH/F9.6 or TCH/F14.4) and on the terrestrial interfaces (intermediate rate: 8, 16, 32, or 64 Kbps).

6.2.1.2 Padding

If the sum of the subchannels has a higher data rate than the primary datastream, padding is necessary (Figure 6.2). In the highest subchannel ($k - 1$), the bandwidth that is not required is filled out with a set pattern. This, however, only applies to transparent data connections. In nontransparent connections, it is the task of the RLP protocol to balance out the unused

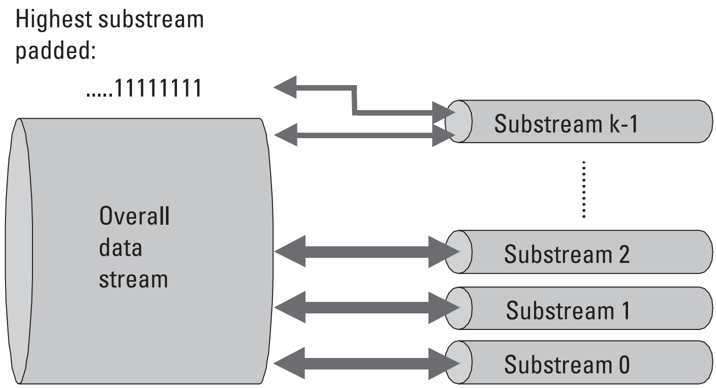


Figure 6.2 Padding in the highest subchannel.

bandwidth with supervisory frames. The greater bandwidth has a positive effect on the efficiency of the protocol since parts of the protocol overheads are compensated. Both sides of the connection (i.e., the split function and the combine function) know whether and in what form padding is to be carried out on account of the channel configuration and the connection parameters. Table 6.2 shows data rates with padding.

6.2.1.3 31-Bit Frame Sync Pattern

Due to different signal transmission times, the frames in the subchannels can lose their chronological context on the transmission path to the combine function and can become disordered by the combine function if they exceed approximately one frame length (or ± half frame length). For example, the frame length of the 80-bit V.110 frame in a 16-Kbps subchannel is 5 ms. This value can easily be exceeded with radio link channels or satellite connections on the transmission path. In order to enable greater transmission time differences, a 31-bit frame sync pattern is transmitted in each subchannel. The split function ensures that the group of frames from the primary datastream receives the same bit position from the 31-bit pattern. The primary datastream is the same as the number of subchannels and is thus distributed simultaneously into the subchannels. Figure 6.3 illustrates this.

Frame sync pattern: 0000 1001 0110 0111 1100 0110 1111 0101 (from left to right).

6.2.1.4 Resynchronization and Combine Function

The position of the frame can be determined with every combination of 5 consecutive bits of the 31-bit frame sync pattern. A maximum shift of 15

Table 6.2
Data Rates with Padding

Data Rate (Kbps)	Rate Without Padding (Kbps)	Fill Rate (Kbps)	Fill Bits/Frames, Bit Positions	Payload Bits/ Frames	Frame Type	Frame Duration (ms)
14.4	19.2 (2×9.6)	4.8	24, D25..48	48	80 bit V.110	5
38.4	43.2 (3×14.4)	4.8	96; see b)	$288 = 8 \times 36$	290-bit radio Itf	20
48	57.6 (4×14.4)	9.6	192; see d)	$288 = 8 \times 36$	290-bit radio Itf	20
56	57.6 (4×14.4)	1.6	32; see a)	$288 = 8 \times 36$	290-bit radio Itf	20
64	72 (5×14.4)	8	160; see c)	$288 = 8 \times 36$	290-bit radio Itf	20
64	67.2 (6×9.6)	3.2	16, D41..56	$56 = (48 + 8)$	Modified 60-bit frame	5

a–d): Padding of the TCH/F14.4 channel type with 290-bit air interface frame (letters refer to those in Figure 6.4).

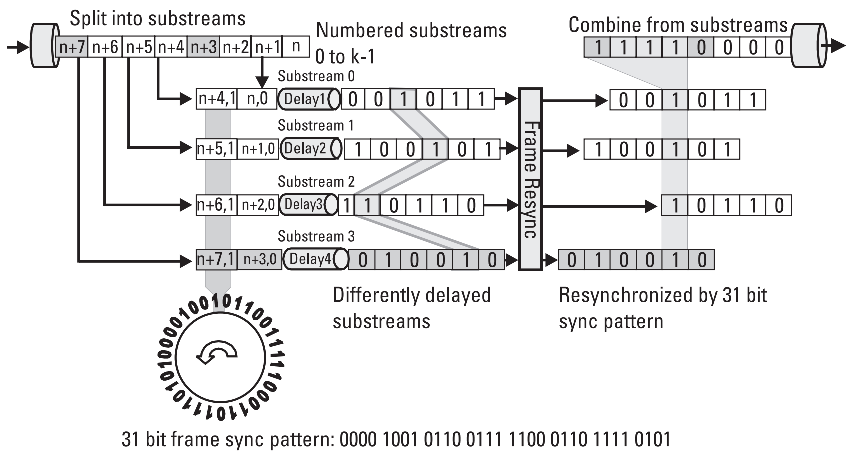


Figure 6.3 Split/combine function.

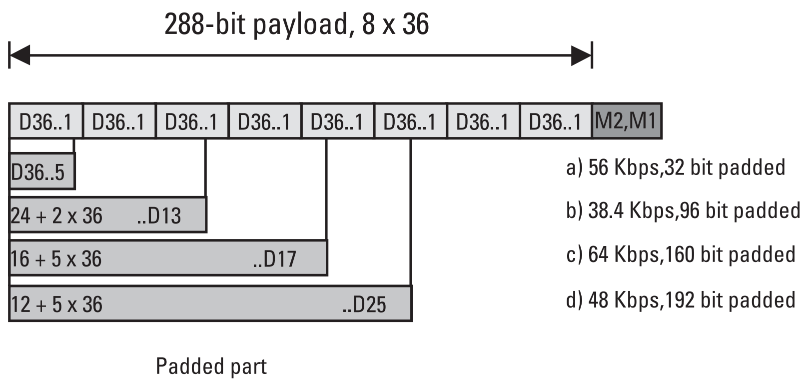


Figure 6.4 Padding in the 290-bit frame.

bits in one direction and 16 in the other can therefore be recognized when comparing positions. When the delay is 15 bit positions or fewer, the frames belong to the same group and are resorted by the combine function into the primary datastream, starting with subchannel 0 and ascending to the highest subchannel, $k - 1$. The cycle begins again at subchannel 0 with the first frame of the next group.

6.2.1.5 Transmission of the Subchannel Number and the Frame Sync Pattern

The subchannel number ($k = 0...7$) is coded with 3 bits. A fourth bit transmits the frame sync pattern. Depending on the data rate of the primary datastream and the channel codings used on the air interface (TCH/F4.8, TCH/F9.6, and TCH/F14.4), the frames used vary, as does the way in which the 4 bits are transmitted into the frame.

The following four cases are to be differentiated:

1. Transparent with TCH/F9.6 or TCH/F4.8, 80-bit V.110 frame;
2. Transparent with TCH/F14.4, 290-bit frame;
3. Nontransparent with TCH/F9.6 or TCH/F4.8, 240-bit RLP frame;
4. Nontransparent with TCH/F14.4, 576-bit RLP frame.

Transparent with TCH/F9.6 or TCH/F4.8, 80-Bit V.110 Frame

The 3-bit subchannel number overwrites the status bits S1, S3 and the first X-bit (between D12 and D13) in the 80-bit V.110 frame (Figure 6.5). The status bit S4 is overwritten for the frame sync pattern. The status bits S6, S8 (SA group) and S9 (SB group) and the second X-bit retain their original function.

Transparent with TCH/F14.4, 290-Bit Air Interface Frame

With the 290-bit air interface frame, a payload rate of 14.4 Kbps can be transmitted on the terrestrial interfaces at 16 Kbps (Figure 6.6). By way of comparison, 32 Kbps would be necessary with the standard 80-bit frame. In the 290-bit frame there are 288 payload bits available, organized in 8 × 36 bits. The two remaining bits are the header bits M1 and M2, which transfer the following information in the multiplex:

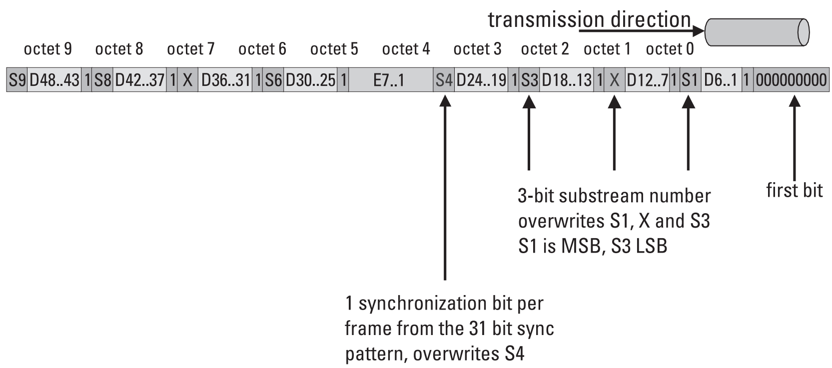


Figure 6.5 80-bit V.110 mapping, frame sync pattern, and subchannel number.

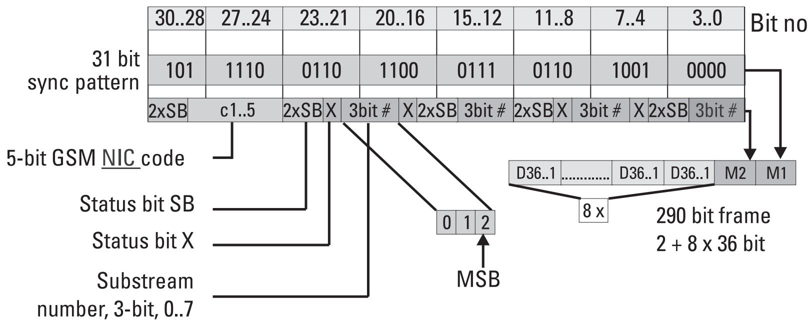


Figure 6.6 290-bit frame mapping, frame sync pattern, and subchannel number.

- M1: The duration of transmission for the 31-bit frame sync pattern is 31 × 290 bit/16 Kbps = 561,875 ms and is packed in a 320-bit A-TRAU frame: 31 × 320 bit/16 Kbps = 620 ms.
- M2: The status bits SB and X (analogous to 80-bit V.110 frame); 5-bit code for the network independent clocking (NIC); and 3-bit subchannel number (0. . .7) of channel bundling.

Nontransparent with TCH/F9.6 or TCH/F4.8, 240-Bit RLP Frame

In nontransparent data connections, it is the task of the split/combine function to hold the four 80-bit V.110 frames that form a 240-bit RLP (see Section 6.4.4.3) together and transmit them in the same subchannel. The frame sync pattern and the subchannel number are not used in this case (i.e., the 80-bit V.110 frame is not modified). In the combine function, the transmission sequence number N(S) of the RLP header is used for reconstructing the frame sequence.

Nontransparent with TCH/F14.4, 576-Bit RLP Frame

Here a complete RLP frame of 576 bits with two 290-bit frames of the 14.4-Kbps channel is also transmitted in a subchannel (see Section 6.4.4.3). The reconstruction of the frame sequence also takes place with the help of the transmission sequence number N(S) of the RLP header.

In order to keep the transmission delay for the long RLP frame as small as possible, the split function has free choice in distributing the frames into the subchannels. It is therefore not bound to the cyclical distribution rule as prescribed in the TCH/F4.8 and 9.6 channels.

6.2.2 Channel Bundling on the Air Interface

Several full-rate channels of the same type are always bundled on the air interface. A mixture is not possible:

- N × TCH/F4.8;
- N × TCH/F9.6;
- N × TCH/F14.4.

Theoretically, N is eight, but it is specified in the GSM recommendations with a maximum of six and, with the mobile stations that are currently available, is limited to a maximum of four. Since the full-rate channels are transmitted on the terrestrial interfaces in 16-Kbps subchannels and an HSCSD connection is limited to 64-Kbps on the A interface, channel bundling remains limited to four subchannels as long as the split/combine function is in the MSC (normal status). Furthermore, the RLP protocol version 2 currently supports a maximum of four subchannels (although it is equipped for eight) in nontransparent data services.

Table 6.3 shows the data rate on the air interface.

When analyzing the table, one poses the question of whether a transparent data rate of 64 Kbps with 6 × 9.6 Kbps (57.6 Kbps) and the application of padding (i.e., the filling out of unusable bandwidth) can be correct or not. The answer is that it is correct. In order to explain this, it is necessary

Table 6.3
Data Rate on the Air Interface (AIUR)

AIUR	TCH/F14.4				TCH/F9.6				TCH/F4.8			
(Kbps)	#	NT	T	P	#	NT	T	P	#	NT	T	P
9.6	—				1*	✓	✓		2	✓	✓	
14.4	1*	✓	✓		2		✓	✓	3	✓	✓	
19.2	—				2	✓	✓		4	✓	✓	
28.8	2	✓	✓		3	✓	✓		—			
38.4	3		✓	✓	4	✓	✓		—			
43.2	3	✓			—				—			
48.0	4		✓	✓	5		✓		—			
56.0	4		✓	✓	5		✓		—			
57.6	4	✓			—				—			
64.0	5		✓	✓	6		✓	✓	—			

* No channel bundling.
T = Transparent data service
NT = Nontransparent data service. The AIUR is the maximum possible AIUR, no V.42bis data compression
P = Padding is the filling out of unusable bandwidth in the highest subchannel with a constant data pattern when the data rate of the bundled full-rate channels is greater than the data rate of the fixed network payload channel (FNUR). Based on analog modem standards and ISDN terminal adapter definitions, the payload rates are always multiples of 0.6 Kbps and thus do not always fit the n × 4.8, 9.6, or 14.4 Kbps of the air interface.

to become more familiar with the payload rates on the air interface. This is dealt with in the following section.

6.2.2.1 Principles of Air Interface Data Rates

The usable data rate (radio interface rate; Figure 6.7) of the traffic channels TCH/F4.8, TCH/F9.6, and TCH/F14.4 is always greater than the data rate actually used in the air interface (AIUR). Figure 6.7 illustrates this concept.

6.2.2.2 Calculation of the Radio Interface Rate

With GMSK modulation, a TDMA burst contains 114 coded payload bits. A burst via a TDMA frame and with a duration of 4.615 ms (48 * 1,250/13 MHz), produces a data rate of 24,700 bps. The frame structure, in which only 24 of the 26 TDMA frames are available for the traffic channel (the 26 multiframe has been discussed in Chapter 1), has to be considered. One frame is idle and one is reserved for signaling (SACCH). The data rate is thus reduced to 22,800 bps. With rate 1/2 channel coding, there is a data rate of 11,400 bps, which is converted at the channel coder input. With the rate 1/3 coder, this rate is 7,600 bps. Puncturing can reduce the data rate at the output of the coder by multiples of 200 bps (1 bit/4.615 ms * 24/26). This gives 100 bps for the rate 1/2 coder and 66.67 bps for the rate 1/3 coder when transmitted to the input. One must also consider that all rate adjustments take place in multiples of four bursts (the length of a TRAU frame is equivalent to four bursts, and signaling blocks also have this length). The data rate can therefore be reduced by puncturing in steps of 400 bps (4*100) or 266.67 bps (4*66.67), respectively.

TCH/F9.6

Figure 6.8 illustrates the path from the 80-bit V.110 frame on the terrestrial interfaces of the PLMN in subchannels of 16 Kbps to the TDMA burst on

Channel Type	Radio Interface Rate (Kbps)	FEC Rate	Punctured Bits per TDMA Burst	AIUR (Kbps)
TCH/F14	14.5	1/2	33	14.4
TCH/F9.6	12.0	1/2	8	9.6
TCH/F4.8	6.0	1/3	0	4.8

Figure 6.7 Radio interface rate. (FEC = forward error correction; rate 1/2 means that the channel coder generates 2 output bits per input bit. Accordingly, at rate 1/3, 3 output bits per input bit are generated.)

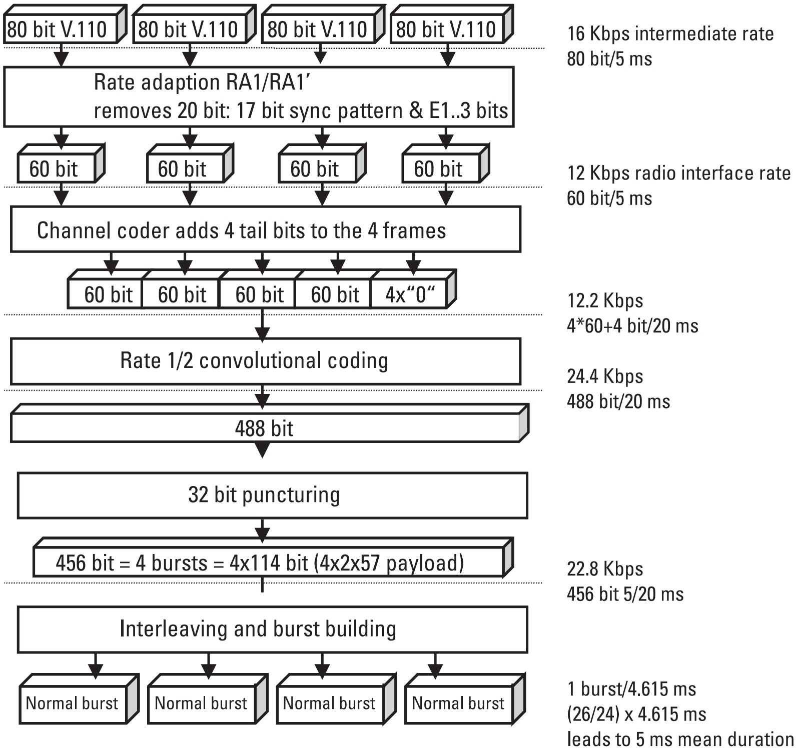


Figure 6.8 80 TCH/F9.6, V.110 frame air interface.

the air interface. The transport of the V.110 frames into data TRAU frames was ignored for this diagram. This applies if the transcoder is not integrated into the BTS (as is the rule). The illustration is also valid for the mobile station. However, all the functions are usually integrated in the device and are not externally accessible. The external data interface is often configured according to the V.24/RS232 standard, and the terminal adapter function can be connected to the channel coder and decoder without being diverted via V.110 frames (the 17-bit sync pattern in the V.110 frame is superfluous, and the information about the data rate can also be obtained elsewhere, E1..3 bits).

Let us return to the explanation of why 6 × 9.6 Kbps enable an AIUR of 64 Kbps and padding is still used.

At a payload rate of 64 Kbps, it is no longer possible to use a V.110 frame on the terrestrial interfaces (i.e., the channel is available to the subscriber as a raw channel). Rate adaptation therefore does not take place, and there are 6 × 12 Kbps = 72 Kbps (radio interface rate) available on the interface. The data rate must then be reduced from 72 Kbps to 64 Kbps by padding.

TCH/F4.8

With rate 1/3 channel coding, the TCH/F4.8 channel offers the highest coding protection/gain as compared to the TCH/F9.6 and 14.4, both of which, as described in Figure 6.7, use a rate 1/2 coder (Figure 6.9).

The 80-bit V.110 frame is transmitted on the terrestrial interfaces in the PLMN in an 8-Kbps subchannel. As with the TCH/F9.6 channel, the remote transcoder is disregarded in the following diagram. If the transcoder is not integrated into the BTS, the V.110 frames are transported in data TRAU frames, (i.e., on 16 Kbps subchannels). The diagram is also valid for the mobile station. However, usually all the functions are integrated into the device (i.e., the same applies as described in the previous section): the V.110 frame is presumably not available.

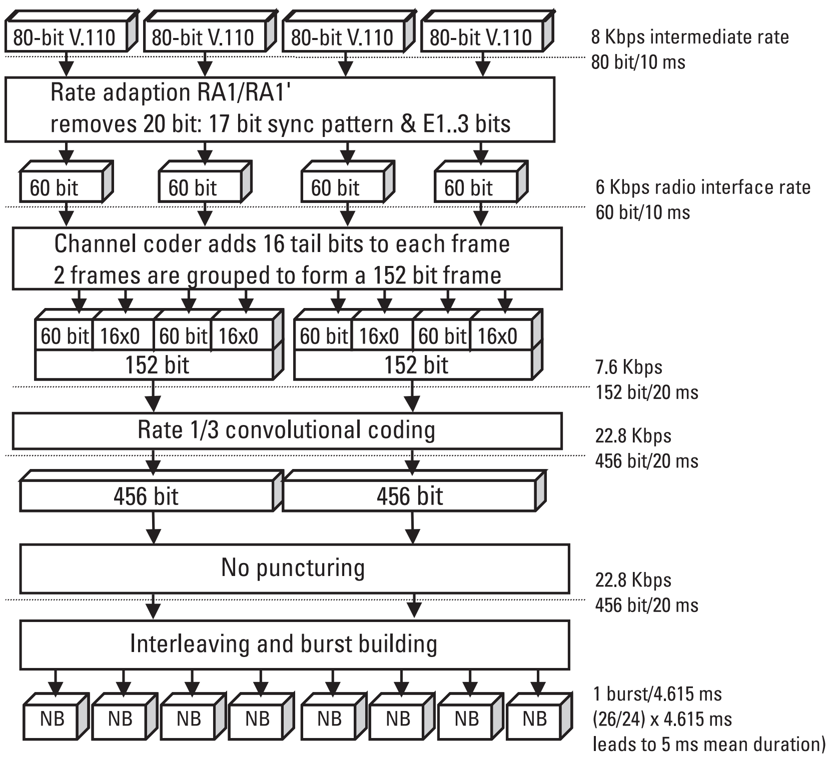


Figure 6.9 TCH/F4.8, V.110 frame air interface.

TCH/F14.4

The TCH/F14.4 channel was introduced with HSCSD (Figure 6.10). In the specification, a lot of work went into transmitting the 14.4-Kbps payload rate into 16-Kbps channels on the Abis and A interfaces, and also supporting data TRAU frames between the transcoder and the BTS (Abis and Ater interfaces). As a result of this, several things are different in the 14.4-Kbps channel from the "simpler" 9.6 and 4.8-Kbps channels. With the standard 80-bit V.110 frame, 14.4 Kbps can only be transmitted in a 32-Kbps channel. Special E-TRAU and A-TRAU frames have been defined to solve the problem and thus introduce new rate adaptation functions. The 14.4-Kbps channel is also special on the air interface: a high puncturing rate of approximately 22% is necessary (132 of 588 bits) in order to adapt the payload rate to the 11.4 Kbps of the air interface.

Finally, rate adaptation in the mobile station and the BTS are different because the mobile station can use the standard 80-bit V.110 frame, since this is, as already mentioned in the last section, merely an internal interface. With nontransparent connections, the RLP frames are passed on to the channel coder directly as 290-bit frames, without being diverted via V.110

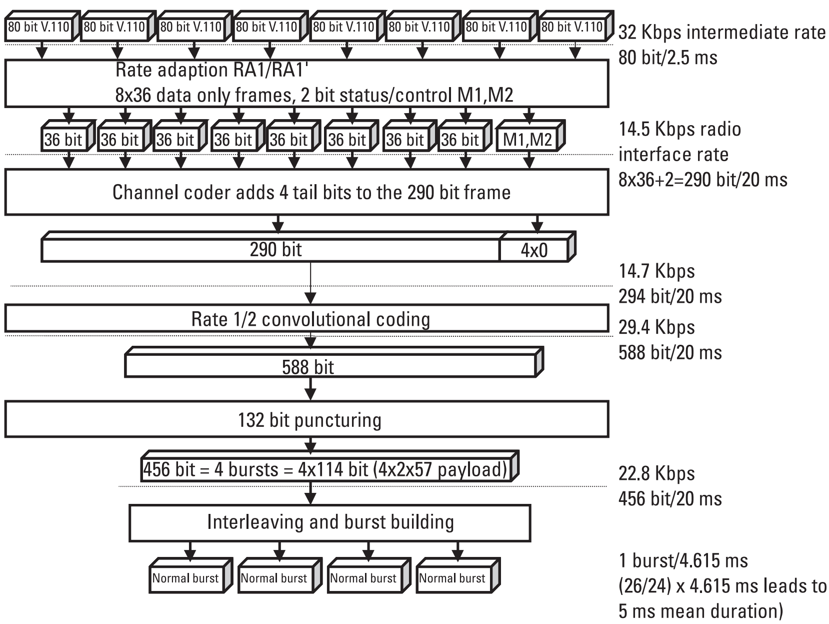


Figure 6.10 TCH/F14.4, V.110 frame air interface.

frames (shown below as $8 \times 36 + M1$ & M2). The coder merely adds 4 tail bits. In the next illustration, the implementation of the mobile station was used for transparent connections. For the base station, the RAA/RAA′ rate adaptation must be added to the RA1/RA1′ rate adaptation in order to take the conversion to 16 Kbps into account.

Note on Interleaving. For reasons of clarity, the effects of interleaving were not illustrated in Figures 6.8 to 6.10. With interleaving, a block of 456 coded bits would be distributed among 22 TDMA bursts. If one considers a single burst, only 24 bits would then belong to the same 456-bit block (see also Section 1.6.4).

6.2.2.3 Asymmetrical Configurations

For the asymmetrical channel configurations, a new channel type was created: the unidirectional full-rate downlink channel. The channel consists of the following:

- Traffic channel full-rate downlink (TCH/FD) payload channel;
- Slow associated control channel multislot downlink (SACCH/MD) signaling channel.

Each asymmetrical configuration consists of at least one bidirectional channel, which carries the signaling in both directions as a main channel, and 1. . .7 unidirectional channels, of which 0. . .6 can also be bidirectional.

In nontransparent connections, the BSS must fill the unused subchannels with V.110 or TRAU idle frames. This ensures that the split/combine in the MSC-IWF does not make an RLP frame out of them.

6.2.2.4 Signaling and Wireless Channel Measurements

The unidirectional downlink channel (TCH/FD) does not support a fast associated control channel (FACCH), so the FACCH signaling (e.g., for a handover) is performed via the FACCH of the main channel. Since the measuring results of the mobile station for the unidirectional downlink channels (RXLEV/RXQUAL) cannot be transmitted in the uplink, the bidirectional main channel is used. Not all measuring results are transmitted, however, but only the values of the worst TCH/FD channel. Additional bidirectional channels use their own SACCH channel in order to transmit the RXLEV and RXQUAL results uplink. Measuring results of the neighboring cell monitoring are transmitted redundantly on all available uplink signaling channels.

6.2.2.5 Hopping

Since HSCSD channel bundling is limited to one carrier, there is, from the mobile station's point of view, no difference between a configuration with one channel or several in frequency hopping (i.e., the frequency does not change for the duration of a TDMA frame—this does not include the frequency duplex between RX and TX that makes a frequency change in the TDMA frame necessary when the mobile station switches between RX and TX).

All subchannels have the same hopping parameters:

- The same training sequence code (TSC).
- The same mobile allocation (MA)—set of frequencies from the cell allocation used in the hopping sequence, with a maximum of 64 frequencies.
- The same mobile allocation index offset (MAIO). The range of values is the same as the number of frequencies in the mobile allocation. The index differentiates all mobile stations that occupy the same mobile allocation and the same time slot in the TDMA frame. In this way the mobile stations are distributed over the frequencies available.
- The same hopping sequence number (HSN). Cyclical hopping (HSN = 0) and random hopping (HSN = 1. . .63) are differentiated.

If the parameters were not the same, especially the MAIO parameter, the mobile station would possibly be forced to change frequencies within a TDMA frame—perhaps even several times in the worst case. Since the transient period of the synthesizers is usually greater than the guard period between two active TDMA bursts, a second synthesizer would be required. This would be programmed to the next frequency while the first was active. This solution is not used for reasons of cost. In single channel connections, the transient behavior was accounted for by a set time shift of three time slots between the receiving and the transmitting directions. In this way, the mobile station has sufficient time for the duplex control and reprogramming of the synthesizer. The problems, therefore, do not arise in frequency hopping. With channel bundling, as the number of transmitting and/or receiving time slots increases, the time for the duplex frequency change and the neighboring cell monitoring decreases.

6.2.2.6 Channel Bundling of the Mobile Station, Classes, and Types

The multislot features of the mobile station are general features and are essentially valid for both HSCSD and GPRS. There is a description of this in Section 2.5.1.

There is the restriction, however, that of the 29 defined device classes, only groups 1 to 18 are recognized by HSCSD. Devices from higher classes (mostly more than six RX time slots) must use a corresponding value from the group named.

6.2.2.7 Encryption, A5.x

The HSCSD bundled channels can also be encrypted on the air interface using the A5.x algorithm. There is basically no difference from a single channel configuration with encryption; the only exception is that the different time slots each have their own ciphering key. Encryption parameter signaling takes place via the main channel, but perfect synchronization of the individual channels is not guaranteed.

6.2.3 Features of the Mobile Station

What is required to make a mobile station HSCSD capable? In short, the following:

- The mobile station must be able to transmit and receive on more than one channel. Because of the introduction of asymmetrical connections, it is sufficient to receive on more than one TDMA time slot. Device class and type determine these features. To control several channels, all signaling expansions (RR/CC/MM) and Layer 1 functions must be available.
- At least one of the three elementary full-rate channel codings must be supported:
  1. TCH/F4.8;
  2. TCH/F9.6;
  3. TCH/F14.4.
- The terminal adapter function (TAF) has the split/combine function and thus realizes the channel bundling. For nontransparent data service, this is closely coupled with the RLP, which must be available in version 2 to support the transmission of frames in subchannels.
- The TAF offers an external interface with a sufficient data rate as an entry point. Channel bundling is no longer visible here. The corresponding interfaces are:

1. V.24/RS232 interface, also as infrared interface (IrDA);
2. PCMCIA (or PC card) interface, such as with the Nokia Card-Phone;
3. Bluetooth as the most modern version;
4. A USB interface is also possible.

- Implementation of signaling between GSM signaling and inband or outband signaling of the external data interface—for example, modem AT commands to establish or terminate a connection, receive a call or set connection parameters (e.g., speed or data compression on/off), or display/request connection status. In the 3GPP standard 27.007 (AT command set) HSCSD-specific commands have also been defined:
  - *HSCSD device parameters, +CHSD:* indicates the HSCSD features of the mobile station—device class (multislot class), maximum number of receiving and transmitting time slots (RX/TX), the sum of both, which can also be smaller than RX+TX, and the channel codings available (ACC: TCH/F4.8, TCH/F9.6, TCH/F14.4);
  - *HSCSD transparent call configuration, +CHST:* provides parameters for configuring the transparent data service (e.g., transmission speed, number of receiving time slots desired, and channel codings available);
  - *HSCSD nontransparent call configuration, +CHSN:* provides parameters for configuring the nontransparent data service, such as desired data rate on the air interface (wanted AIUR), number of receiving time slots desired, and channel codings available;
  - *HSCSD current call parameters, +CHSC:* provides information on the current HSCSD connection (e.g., number of receiving and transmitting time slots, data rate on the air interface (AIUR), and channel codings used;
  - *HSCSD parameters report, +CHSR:* report function that sets the HSCSD parameters agreed upon with the network during an HSCSD connection establishment (i.e., before the CONNECT):

    1. Number of receiving and transmitting time slots;
    2. Data rate on the air interface (AIUR);
    3. Channel coding used.

    The function can be switched on or off using the CHSR command.

- *HSCSD automatic user initiated upgrading, +CHSU*: determines whether, in nontransparent data connections, the data rate can be increased using the UP bit in the frame header of the radio link protocol (see Section 6.4.4.4) (service level upgrading). Consequently, the number of receiving and transmitting time slots and/or the channel coding used can change.

6.3 Connection Chain Using Nontransparent/ Asymmetrical Data Service as an Example

As an example, a complete nontransparent HSCSD connection will be described below. A configuration that is common in practice is a PPP connection from a PC or notebook via the mobile station to the Internet provider with a modem pool at the PSTN.

The configuration is as follows (see also Figure 6.11):

- PC/notebook at the mobile station as terminal adapter and network access point (R interface) with asynchronous interface (e.g., V.24);
- Internet provider with analog modem pool at the PSTN, V.90 compatible;
- PPP data transmission connection as basis for TCP/IP Internet connection;
- Mobile station with three RX time slots and one TX time slot (3 + 1), corresponding to device class 4 and type 1 (i.e., no simultaneous transmitting and receiving);
- Nontransparent general bearer service with TCH/F14.4 channel coding—this results in the following theoretical connecting speeds:
  - Downlink (to the MS): 43.2 Kbps (3 × 14.4 Kbps);
  - Uplink (from the MS): 14.4 Kbps (1 × 14.4 Kbps).
- V.42bis data compression.

With the nontransparent service and the bundling of three TCH/F14.4 channels, the signal chain as illustrated in Figure 6.12 is produced.

6.3.1 Function Blocks in the Signal Chain

6.3.1.1 MS/MT

R Interface. The mobile station is the reference access point for the PC. Each of the three asynchronous interfaces shown above is suitable:

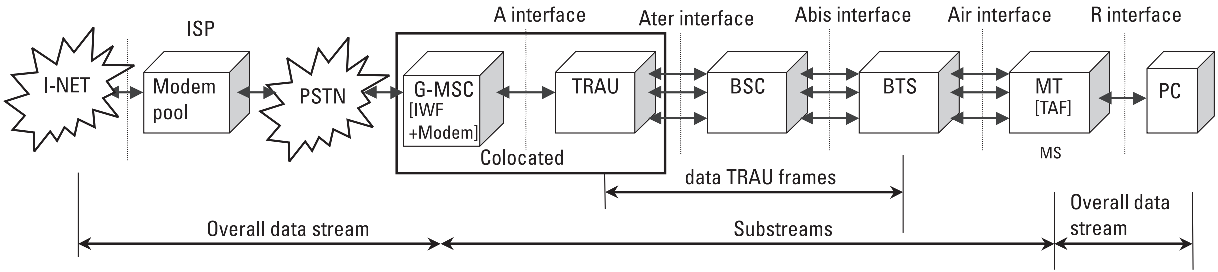


Figure 6.11 Configuration example.

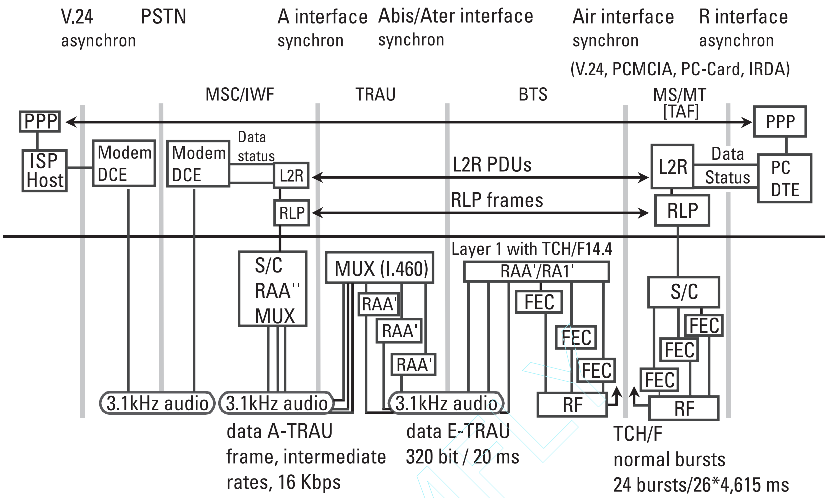


Figure 6.12 Signal chain.

- V.24;
- PCMCIA/PC-Card;
- IrDA.

TAF. The terminal adapter function of the mobile station imitates a modem and can be controlled with AT commands (the HSCSD-specific commands have already been introduced in Section 6.2.3). In practice, however, the manufacturer should provide SW drivers and applications that take over the configuration of the connection parameters and interface settings and integrate themselves into the data transmission concept ("aspects" may be a better word) of the operating system.

L2R. The Layer 2 relay establishes transparent connection to the modem in the IWF of the MSC as if the PC were connected directly, as data terminal equipment (DTE), to the modem in the MSC as data communication equipment (DCE). All layers below this are hidden. The Layer 2 relay assumes flow control. There is usually the choice between hardware flow control (RTS/CTS), also known as outband flow control, and software flow control (Xon/Xoff), also known as inband flow control. Both flow controls use the X-bit in the status octet of the L2R-PDU for transmitting. There is a more detailed description of the L2R-COP PDU and the status octet in Section 6.4.5.1. The L2R-COP PDU is 66 octets long and forms the data part of the 576-bit RLP frame.

RLP. With the TCH/F14.4 channel type, 576-bit RLP frames are handed over to the split/combine function with two 290-bit air interface frames, or exchanged with it. See Section 6.4.4 for details on the mapping of the RLP frame onto the 290-bit air interface frame.

S/C. The split/combine function distributes complete RLP frames (i.e., two 290-bit air interface frames) to the three subchannels. This function has already been described in detail in Section 6.2.1.5. This produces a synchronous rate of 290 bits/20 ms = 14.5 Kbps per subchannel.

FEC. Channel coding/decoding and subchannel junction to TDMA air interface time slots. The 3 + 1 configuration consists of:

- 1 × TCH/F14.4 = 1 main channel
- 2 × TCH/FD14.4 = 2 unidirectional, downlink only channels (BTS ⇒ MS)

The rule for the mapping of the subchannels to the TDMA time slots has been described with the split function in Section 6.2.1.1.

RF. High-frequency component of the mobile station, illustrated for completeness.

6.3.1.2 BTS

The BTS is limited to Layer 1 functions for data transfer during connection. In our example, three 16-Kbps channels form the Abis interface belong to the connection. E-TRAU frames are transmitted on them to the remote transcoder. The RA1′/RAA′ function (shown as RAA′/RA1′) performs the adaptations between the radio interface and the E-TRAU.

As a rule, there is a fixed mapping between the eight TDMA time slots and the 16-Kbps channels on the Abis interface. This is described in [1], Chapter 6. In Figure 6.12 the transcoder is remote to the BTS, and the RAA′/RA1′ rate adaptation is therefore also remote to the codec (FEC).

FEC. There is no difference between the channel coding and its counterpart in the mobile station. Here, the 3 + 1 configuration also consists of:

- 1 × TCH/F14.4 = 1 main channel;
- 2 × TCH/FD14.4 = 2 unidirectional, downlink only channels.

The unused uplink resources of the two TCH/FD channels must be filled with idle E-TRAU frames (C6 = 1) (see also Section 6.4.3).

RAA′/RA1′. Adaptation between the TRAU and the radio interface. More information can be found in Section 6.4.3 (TRAU rate adaptations).

RF. High-frequency component of the BTS, also illustrated for completeness here.

6.3.1.3 BSC

The base station controller is completely transparent for data connection. As a coupling element, it switches the 16-Kbps channels (intermediate rate) between the BTS (Abis interface) and the TRAU (Ater interface) and, of necessity, also the A interface (the interface between TRAU and MSC has fixed mapping). Resource allocation must take place in such a way that all the 16-Kbps channels of the Ater interface belong to one 64-Kbps channel on the A interface. So for successful establishment of connection, three of four 16-Kbps channels of a 64-Kbps A interface must be free.

6.3.1.4 TRAU

The RAA′ function adapts between the A-TRAU on the A interface and the E-TRAU on the Ater or Abis interface and multiplexes the three 16-Kbps channels into a 64-Kbps channel on the A interface (to ITU-T I.460).

6.3.1.5 MSC/IWF

The modem to be used for establishing the connection via the PSTN to the modem pool of the ISP is situated in the MSC.

Modem. Modem function in the MSC. With interworking with the PSTN, these analog modems are in a pool or SW modems on interface components.

Since there are many analog modem standards, the mobile station has a number of control possibilities for signaling during establishment of a connection. Here are some examples:

- Autobauding type 1, automatic speed selection, only possible with nontransparent connection;
- 300 bps (V.21);
- 1,200 bps (V.22);
- 2,400 bps (V.22bis);
- 2,400 bps (V.26ter);
- 4,800 bps (V.32);
- 9,600 bps (V.32);

- 9,600 bps (V.34);
- 14,400 bps (V.34);
- 19,200 bps (V.34);
- 28,800 bps (V.34);
- 33,600 bps (V.34).

In our example, the parameter "Autobauding type 1" is used. If there is a V.90 modem in the IWF, a downlink data rate of 43.2 Kbps with 3 × 14.4 Kbps is possible (+ data compression). If not, then the V.34 speed of 33.6 Kbps or 28.8 Kbps is probable. If only 28.8 Kbps is achieved, the network may automatically reduce channel bundling to 2 + 1, with 2 × 14.4 Kbps.

L2R. Counterpart to the Layer 2 relay in the mobile station. Together, both enable the transparent connection between the modem in the MSC and the PC at the MS.

RLP. Counterpart to the RLP in the MS. With the TCH/F14.4 channel type, 576-bit RLP frames with two 290-bit air interface frames each are handed over to the split/combine function or exchanged with it. See Section 6.4.4 for details on the mapping of the RLP frame on to the 290-bit air interface frames.

S/C. The split/combine function distributes complete RLP frames (two 290-bit air interface frames) to the three subchannels. This function has been described in detail in Section 6.2.1.5. This gives a synchronous rate of 290 bits/20 ms = 14.5 Kbps per subchannel.

RAA". Adapts between the A-TRAU in the 16 Kbps channel and the synchronous rate of the split/combine function.

MUX. Same function as in the TRAU. Multiplexes the three 16-Kbps channels into one 64-Kbps channel on the A interface (to ITU-T I.460).

6.3.2 Call Setup

Now that the signal chain has been introduced, we have to select the connection parameters. The first thing to clarify in advance is that the subscriber can use the general bearer service and the service has been approved by the operator (subscription checking).

In order that data conversations can also be received [mobile terminated call (MTC)], the subscriber receives an additional MS-ISDN number. This is the only way to distinguish a data call from the PSTN from a speech call so that the modem in the MSC can take on the task of setting the connection parameters (e.g., transmission speed and V.42bis compression) and then make them available for the rest of the setup and undertake the necessary resource reservations.

A basic principle of call setup is that the more parameters there are available (bearer capability element in call setup/call confirm message), the greater the probability is that a multislot connection is achieved successfully.

6.3.2.1 Connection Parameters

Table 6.4 lists the essential connection parameters of the bearer capability elements. All the values selected for our example are in block.

None of these parameters answer the following questions: How does the PLMN know the multislot capabilities of the mobile station, and how are the RX and TX time slots ordered in the TDMA frame?

The MS must signal the information about the device class before the call setup; this is the only way that resources in the PLMN and the compatibility check can be performed. The *early classmark sending* is used for this purpose. As soon as the Layer 2 connection has been made on the main signaling channel; the classmark information is transmitted. The 5-bit device class value is then to be found in the CLM3 element.

6.4 Selected Details

6.4.1 ISDN Data Frames

Since large parts of the GSM PLMN are based on ISDN technology, the function of the data channels based on the V.110 frame has been taken from ISDN (Figure 6.13). ISDN terminal adapters support data terminals with V interfaces using the V.110 frame. This includes, for example, asynchronous interfaces such as V.24 (also known as RS232).

In the ISDN network, V.110 frames are transmitted between the following network elements:

- ISDN-TE (ISDN-compatible terminal equipment);
- TA-V (terminal adapter V): supports non-ISDN-compatible terminals with V-interfaces (e.g., an analog modem with asynchronous V.24 interface);

Table 6.4
Essential Connection Parameters of the Bearer Capability Elements

ITC	Information transfer capability: Speech Unrestricted digital 3.1-kHz audio ex PLMN Facsimile group 3 Other ITC
Transfer mode	Circuit mode Packet mode
Connection element	On entry from the list: Transparent Nontransparent Both, transparent preferred Both, nontransparent preferred
Number of stop bits	Does not apply in the PLMN (see Section 6.4.5.1), but only between PC and MS, and between the IWF modem and the remote modem: 1 stop bit 2 stop bits
Number of data bits	Also does not apply in the PLMN but only between PC and MS and IWF modem and remote modem: 7 data bits 8 data bits
Parity information	Also does not apply in the PLMN but only between PC and MS and IWF modem and remote modem: Odd Even None Fix 0 Fix 1
Compression	V.42bis: Compression possible and permitted Compression not possible or not permitted/desired
Modem type	None V.21 V.22 V.22bis V.26ter V.32 Modem for undef. interface Autobauding type 1
Other modem type	Continuation of the modem type list. Had to be introduced because of lack of codes: No other modem type specified in this field V.34

Table 6.4 (continued)

FNUR	This is ignored when modem type autobauding type 1 is selected: 9.6 Kbps 14.4 Kbps 19.2 Kbps 28.8 Kbps 38.4 Kbps 48.0 Kbps 56.0 Kbps 64.0 Kbps
WAIUR	This controls asymmetrical connection (AIUR = 14.4 Kbps, WAIUR = 43.3 Kbps): Air interface user rate not applicable 9.6 Kbps 14.4 Kbps 19.2 Kbps 28.8 Kbps 38.4 Kbps 43.2 Kbps 57.6 Kbps Interpreted by the network as 38.4 Kbps
ACC	Acceptable channel codings: TCH/F4.8 TCH/F9.6 TCH/F14.4
mTCH:	Maximum number of TCH/F: 1 TCH 2 TCH 3 TCH 4 TCH 5 TCH 6 TCH
UIMI	User initiated modification indicator: 1. UIMI not required 2. Up to 1 TCH/F 3. Up to 2 TCH/F 4. Up to 3 TCH/F 5. Up to 4 TCH/F can be requested

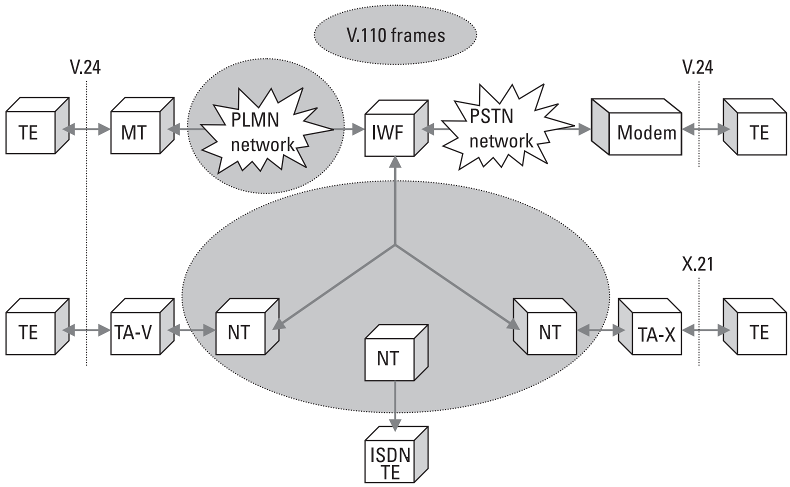


Figure 6.13 V.110 in ISDN and GSM PLMN.

- TA-X (terminal adapter X): supports non-ISDN-compatible terminals with X-interfaces (e.g., a synchronous terminal with X.21/X.21bis interface);
- IWF: Interworking function in the PLMN or ISDN network at the junction of non-ISDN networks.

In the GSM PLMN, the V.110 frames are transmitted between the terminal adapter of the mobile station (MT) and the IWF in the MSC.

6.4.1.1 Features of the V.110 Frame

The V.110 frame is of fixed length: 80 bits in its original form. Only 60% of this (i.e., 48 bits) are usable as payload; the rest are reserved for control and status information. This means that the maximum data rate in a 64-Kbps ISDN channel is limited to 38.4 Kbps. In order to increase the payload rate, two more frames with low overhead were defined: a 32-bit frame for up to 48 Kbps and two 64-bit frames for a maximum of 56 Kbps. The two 64-bit frames are active at different times. One is used for connection setup and termination and has a sync pattern. The other is active in the actual data transmission phase and does not have frame synchronization. Only the 80-bit frame supports asynchronous data connections with up to 38.4 Kbps. The higher data rates of 48 Kbps and 56 Kbps are only suitable for synchronous connections.

Table 6.5 gives an overview of the use of the frames and the necessary bandwidth that leads to submultiplexing when less than 64 Kbps is required

Table 6.5
Payload Options of the V.110 Frame

FNUR (Kbps)	Intermediate Rate (Kbps)	Asynchronous	Synchronous	V.110 Frame Type
0.6	8	✓	✓	80 bit
1.2	8	✓	✓	80 bit
2.4	8	✓	✓	80 bit
4.8	8	✓	✓	80 bit
7.2	16	✓	✓	80 bit
9.6	16	✓	✓	80 bit
12.0	32	✓	✓	80 bit
14.4	32	✓	✓	80 bit
19.2	32	✓	✓	80 bit
24.0	64	✓	✓	80 bit
28.8	64	✓	✓	80 bit
38.4	64	✓	✓	80 bit
48.0	64		✓	32 bit
56.0	64		✓	64 bit

for transmission (intermediate rates of 8, 16, and 32 Kbps, as defined in ITU-T I.460). We describe the payload rate as FNUR, in accordance with the GSM PLMN specifications.

Table 6.6 shows the intermediate rate frame duration of the 80-bit V.110 frame.

80-Bit V.110 Frame

The four parts of the 80-bit V.110 frame are as follows (Figure 6.14):

1. 48 bits payload;
2. 17-bit sync pattern for frame recognition;
3. 8 status bits (S and X), mainly for flow control of asynchronous terminals;
4. 7 control bits (E), mainly for coding the payload rate.

Bit Synchronization Pattern. Eight consecutive 0 bits characterize the synchronization element. The remaining nine 1 bits of the sync pattern ensure that this combination cannot occur anywhere else in the frame (i.e., in normal conditions, without bit errors).

8 Status Bits (S1, S3, S4, S6, S8, S9, 2 × X). The S and X bits transmit control and status signals between terminal equipment and terminal adapter from one

Table 6.6
Synchronization and Data Rates of the V.110 Frame

Intermediate Rate (Kbps)	V.110 80-bit Frame Period (ms)	Max. User Rate (Kbps)
8	10	4.8
16	5	9.6
32	2.5	19.2
64	1.25	38.4

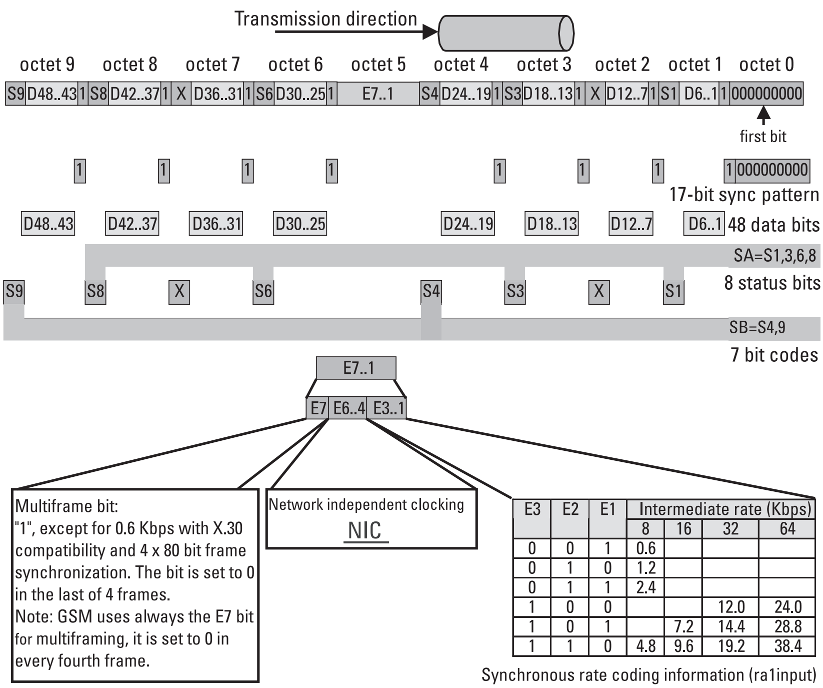


Figure 6.14 80-bit V.110 frame.

connection side to the same interface on the other side, and synchronization information between the two terminal adapters.

The following functions are performed with the S and X bits:

- Flow control;
- Synchronization of the terminal adapters in call setup and termination phase.

In HSCSD channel bundling, four of the eight S and X bits are used for numbering the subchannels and the 31-bit frame sync pattern. For details on this, see Section 6.2.1.

The S bits are classified in two groups:

1. SA-bits: S1, S3, S6, S8;
2. SB-bits: S4, S9.

Each S group transmits a scanned V interface HW signal from the input of the terminal adapter on one connection side to the output of the terminal adapter on the other side (i.e., the outband flow control of the V interface is transmitted to the inband flow control of the V.110 frames).

The X-bit is also used to transmit a scanned HW signal but also has the general task of flow control and can thus be used for controlling the internal buffers of both terminal adapters (Xon/Xoff). Therefore, an end-to-end flow control is realized (e.g., when the Vinterface of the terminal adapter/terminal equipment pair on one side works more slowly than that on the other side and is also slower than the connection channel in the network).

7 Code Bits (E1. . .7). The code bits are classified in three groups:

1. E1. . .3: Coding of the data rate as illustrated in Figure 6.14;
2. E4. . .6: Network independent clocking—for details see Section 6.4.2;
3. E7: Multiframe coding. Always set at 1 in the ISDN network except with 0.6 Kbps and X.30 compatibility, where four 80-bit frames form a multiframe. The last of the four frames is marked with E7 = 0. In the GSM PLMN, E7 is always used for multiframe identification, but the bit in the first of the four frames is 0 (see also Figure 6.15).

32-Bit V.110 Frame

The four parts of the 32-bit V.110 frame are as follows (Figure 6.16):

1. 24 bits payload;
2. 4-bit sync pattern for frame recognition;
3. 4 status bits;
4. No control bits.

As compared to the 80-bit V.110 frame, the payload proportion in the frame increases from 60% to 75%, and the maximum data rate in the 64-Kbps ISDN channel increases to 48 Kbps.

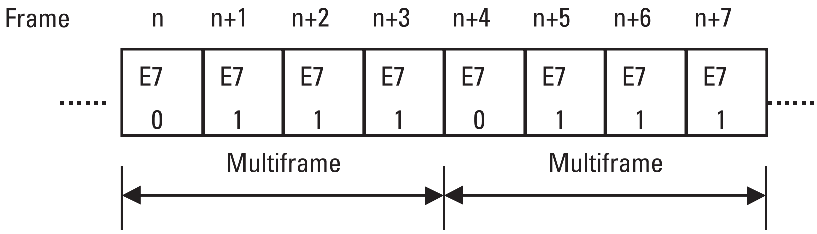


Figure 6.15 E7 multiframe bit.

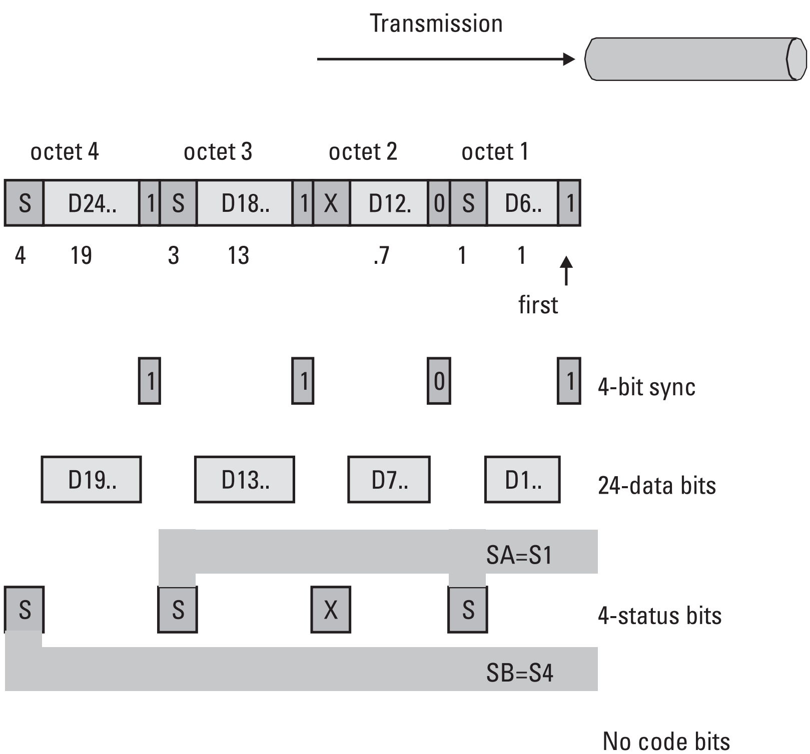


Figure 6.16 32-bit V.110 frame.

64-Bit V.110 Frame

The 64-bit frame has two formats, a basic frame for data transfer in the active part of the connection, and an alternative frame for synchronization, call setup, and termination (Figure 6.17).

The basic frame has the following four parts:

1. 56 bits payload;
2. 8 bits set at 1, to support ISDN networks restricted to 56 Kbps;
3. No status bits;
4. No control bits.

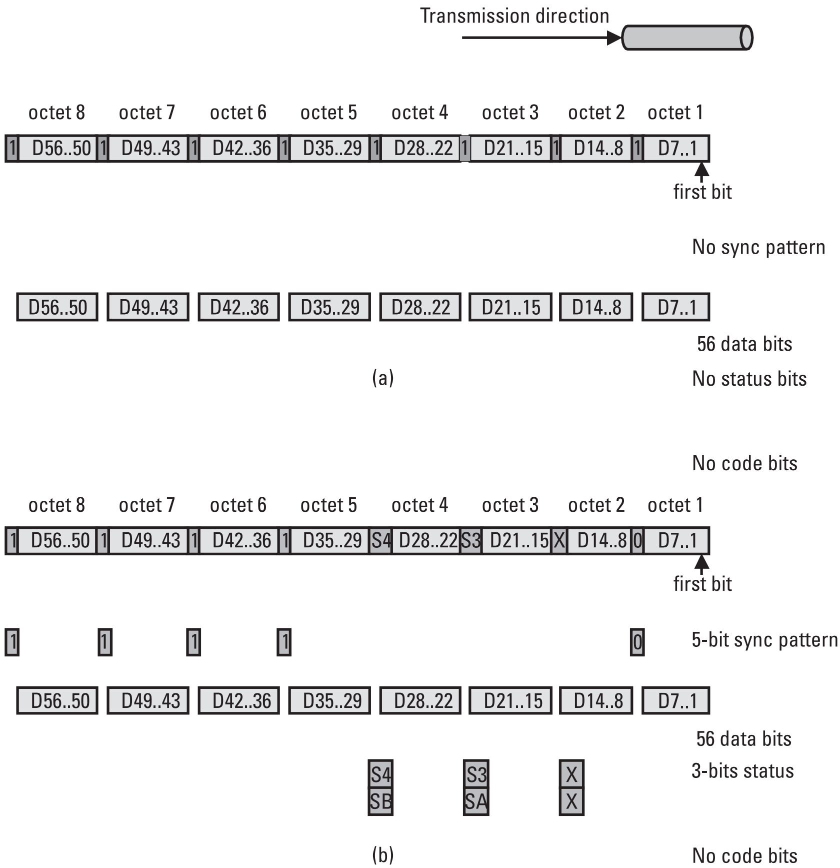


Figure 6.17 64-bit V.110 frame: (a) basic and (b) alternative.

The alternative frame has the following four parts:

1. 56 bits payload;
2. 5-bit sync pattern;
3. 3 bits of status bits;
4. No control bits.

As compared to the 80-bit V.110 frame, the payload proportion in the frame increases from 60% to 87.5%, and the maximum data rate in the 64-Kbps ISDN channel increases to 56 Kbps.

6.4.2 Network Independent Clocking

Network independent clocking (NIC) is the measuring and alignment of clocking differences between network elements that do not have synchronized clocking rates but support synchronous data connections (Figure 6.18). In the GSM PLMN, this affects synchronous data services with interworking to the PSTN or ISDN and 3.1-kHz audio option.

Figure 6.18 shows an example with interworking of the GSM PLMN to the PSTN. Two analog modems are connected via the PSTN, one at the PSTN access and one in the IWF of the PLMN. Both are synchronized via a carrier signal (e.g., V.34 modem). In the PLMN, a synchronous data service is chosen because, to give one example, the TE on the mobile station only supports synchronous services. The junction between two clocking domains, the PLMN with its 8-kHz PDH architecture, and the carrier-

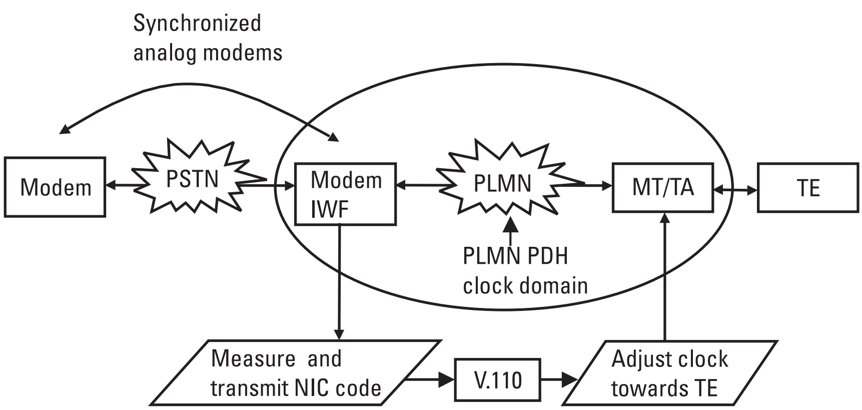


Figure 6.18 NIC.

synchronized analog modem, is situated at the modem in the IWF. Even if the same payload rate (e.g., 9.6 Kbps) has been configured in both domains, small clocking differences can add up in the course of the connection and reach a size of 1 bit (or baud). Depending on the sign of the deviation, this can lead to a loss of or surplus data. Such deviations (approximately 100 ppm) can be counterbalanced with the NIC function (approximately 100 ppm). Furthermore, information as to whether a 0 or 1 bit must be inserted, a bit is to be deleted, or no compensation is necessary is transmitted with a 5-bit code in the V.110 frame (C1. . .5). The code is selected so as to be resistant to individual bit errors in the V.110 frame (perfect linear block code). In ISDN, the NIC function is coded somewhat differently: 3 bits (E4. . .6) are used without block protection. The phase status of both clocking rates is also transmitted in steps of 20% with eight states. This is illustrated for both networks, ISDN and GSM, in Figure 6.19: in the upper part with a faster clocking rate and the addition of a bit, and in the lower part with a slower clocking rate and the deletion of a bit.

6.4.3 Rate Adaptations

The GSM rate adaptations will now be introduced briefly and will provide help in understanding the functions and layers of rate adaptations described in GSM 03.10, 04.21, and 08.20 more easily. The different rate adaptations result from the variety of channel types (TCH/F4.8, 9.6 and 14.4), connection types (asynchronous, synchronous, transparent, and nontransparent), interfaces (asynchronous, synchronous, terrestrial, air interface), and special functions (transcoder and data TRAU frame, split/combine function).

6.4.3.1 RA0, Asynchronous ⇔ Synchronous Adaptation

The RA0 function adapts asynchronous and synchronous interfaces up to the maximum data rate of 38.4 Kbps. Here, the asynchronous data rate is adapted to the next higher or same synchronous rate in multiples of 600 bps ($2^n * 600$ bps; $n = 0. . .6$). Rate differences and clocking tolerances are balanced out on the synchronous side by adding or deleting stop units.

The following shows the construction of the GSM serial data unit (Figure 6.20):

- 1 start and 1 stop bit;
- Data and parity:

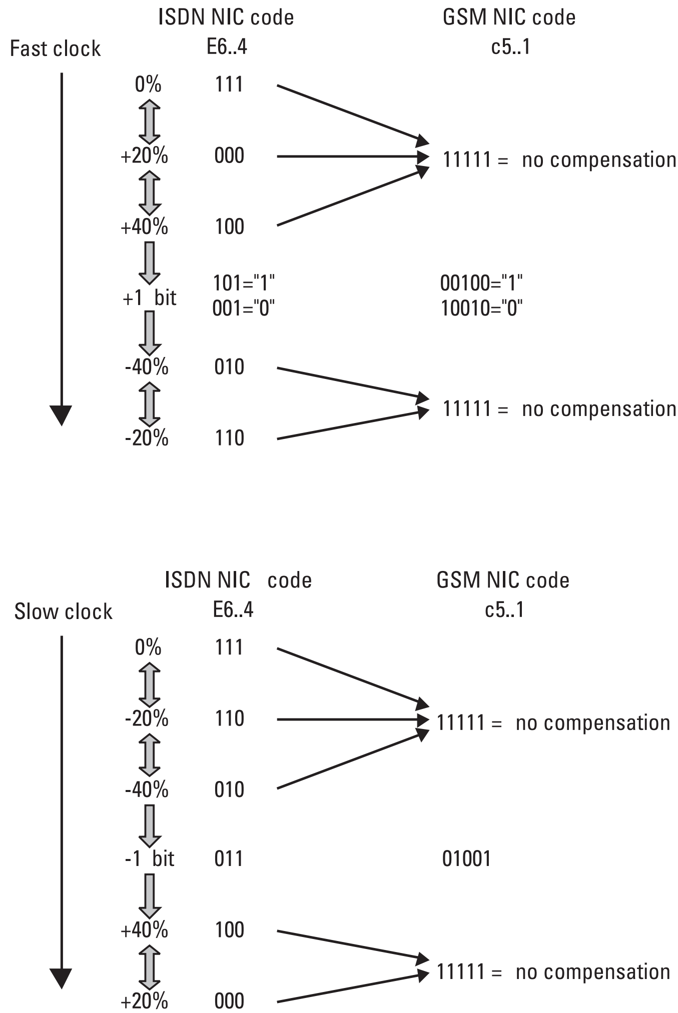


Figure 6.19 NIC coding.

1. 7 bits of data + 1 parity bit;
2. 8 data bits and no parity bits (see GSM 07.01);
3. Parity:
   - Even;
   - Odd;
   - No;
   - Fixed 1;
   - Fixed 0.

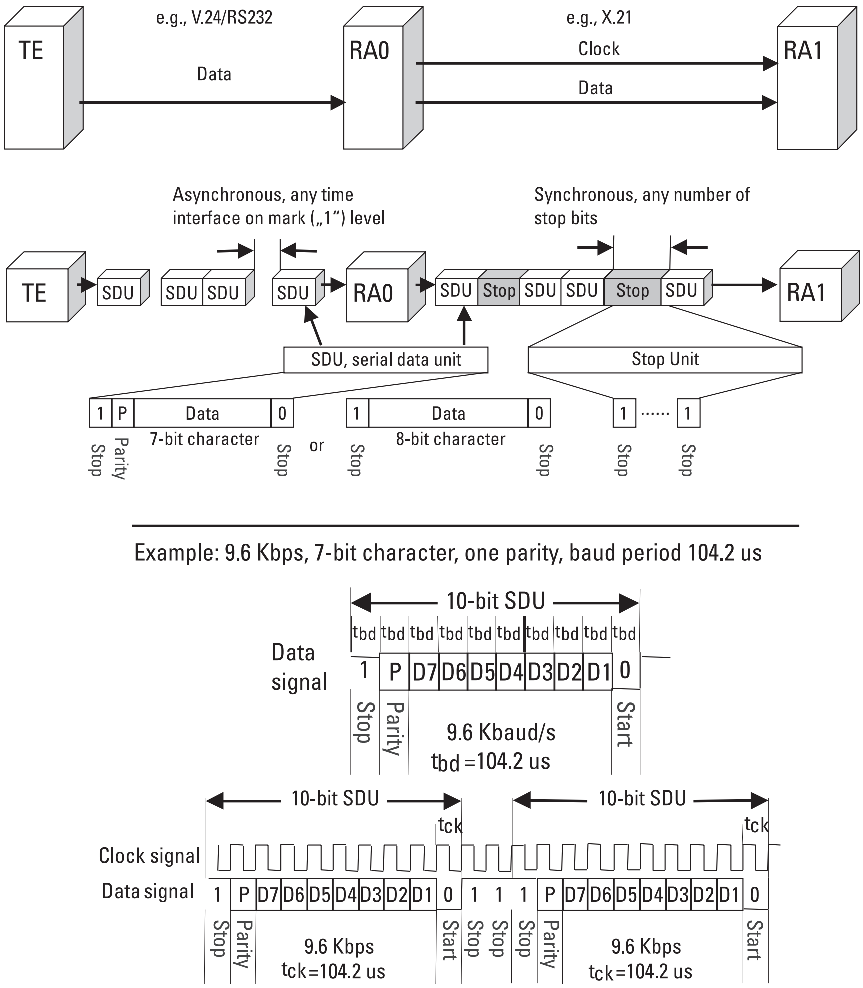


Figure 6.20 RA0 serial data units.

The RA0 function is also defined in ISDN, or rather it was taken over from ISDN and modified for GSM. The modifications include the following:

- No 5 and 6-bit data length;
- No 1.5 bit stop bit length;
- No 8-bit data + parity.

The RA0 function guarantees SDU integrity (i.e., on both sides of the function, the composition of the SDU is identical). However, the stop bit of the SDU may be deleted in order to balance the clocking rate if the data rate on the asynchronous side is greater than on the synchronous side; the SDU integrity must then be restored in the receiver by adding a stop bit. The deletion rate is limited to one of any eight consecutive SDUs.

Table 6.7 lists all the data rates of the RA0 function that are supported for ISDN and GSM.

6.4.3.2 RA1, Synchronous ⇔ Intermediate Rate of 8/16/32/64 Kbps

The RA1 function generates V.110 frames and takes over the adaptation of synchronous, serial data rates of up to 38.4 Kbps (e.g., from the output of the RA0 function) or a synchronous serial interface (e.g., X.21), to data rates derived from the octet structure of the 64-Kbps channel, known as intermediate rates. Depending on the data rate at the input to the RA1 function, there are four possible intermediate rates: 8, 16, 32, and 64 Kbps. In a further stage, the intermediate rates can be adapted to the standard ISDN data rate of 64 Kbps using the RA2 function; there is no difference between the intermediate rate of 64 Kbps and the 64 Kbps rate of the ISDN channel on the S interface. As can be seen in Table 6.8, not all the synchronous data rates defined by ISDN are supported in the GSM PLMN.

The functions of the RA1 adaptation are as follows:

Table 6.7
RA0 Data Rates

ISDN V.110		GSM 04.21	
Async. User Rate	**Sync. User Rate**	**Async. User Rate**	**Sync. User Rate**
≤0.6	0.6	≤0.6	0.6
1.2	1.2	1.2	1.2
2.4	2.4	2.4	2.4
3.6	4.8	—	—
4.8	4.8	4.8	4.8
7.2	9.6	—	—
9.6	9.6	9.6	9.6
12.0	19.2	—	—
14.4	19.2	14.4	14.4
19.2	19.2	19.2	19.2
24.0	38.4	—	—
28.8	28.8	28.8	28.8
38.4	38.4	38.4	38.4

Note: All rates are in kilobytes per second.

Table 6.8
RA1 Data Rates

Data Rate (bps)	ISDN RA1 Input Rate (bps)	GSM RA1 Input Rate (bps)	RA1 Output Rate (Kbps)	Bit Duplication Factor	Fill Bits Set to 1	E3..1 Coding (Binary)
50	600	600	8	8	0	001
75	600	600	8	8	0	001
110	600	600	8	8	0	001
150	600	600	8	8	0	001
200	600	600	8	8	0	001
300	600	600	8	8	0	001
600	600	600	8	8	0	001
1,200	1,200	1,200	8	4	0	010
2,400	2,400	2,400	8	2	0	011
3,600	4,800	—	8	1	0	—
4,800	4,800	4,800	8	1	0	110
7,200	9,600	—	16	1	0	—
9,600	9,600	9,600	16	1	0	110
12,000	19,200	—	32	1	16	100
14,400	19,200	14,400	32	1	12	101
19,200	19,200	19,200	32	1	0	110
24,000	38,400	—	64	1	16	100
28,800	38,400	28,800	64	1	12	101
38,400	38,400	38,400	64	1	0	110

- Bit repetition and fill bits (padding) to adapt the input and output data rates;
- Handling of the S and X bits of the V.110 frames, contains inband and outband flow control;
- V.110 frame synchronization;
- Coding of the data rate in the E1. . .3 bits;
- Support by NIC as described in Section 6.4.2;
- Multiframe synchronization with the E7 bit (see Section 6.4.1.1).

Bit Repetition and Filler Bits

Figure 6.21 illustrates the adaptation of the data rates by bit repetition and the insertion of filler bits.

Multiframing, E7 Bit

In the GSM PLMN, four V.110 frames form a multiframe with the help of the E7 bit (Figure 6.15). The first of the four frames is marked with E7 = 0.

6.4.3.3 RA1″, Synchronous 48/56 Kbps ⇔ Intermediate Rate 64 Kbps

The RA1″ function adapts between the two synchronous data rates of 48 and 56 Kbps and the intermediate rate of 64 Kbps. The standard 80-bit V.110 frame is not suitable for either data rate due to the frame overhead. As described in Section 6.4.1.1, one 32-bit frame and two 64-bit frames with a reduced overhead are required. These are generated and synchronized by the RA1″ function.

6.4.3.4 RA2, Intermediate Rate 8/16/32 Kbps ⇔ 64 Kbps (S Interface)

With the RA2 function, the intermediate rates of 8, 16, or 32 Kbps (e.g., at the output of the RA1 function) are adapted to the 64-Kbps ISDN rate on the S interface. The function does not change the V.110 frame—it only switches the highest-value unused bit positions in the octet-structured 64-Kbps channel to 1, as illustrated in Figure 6.22.

6.4.3.5 RA1/RA1′, Intermediate Rate ⇔ Radio Interface

The RA1/RA1′ adaptation is a relay function and adapts the intermediate rates of 8, 16, 32, and 64 Kbps to the radio interface rate of 6 Kbps (TCH/F4.8), 12 Kbps (TCH/F9.6), and 14.5 Kbps (TCH/F14.4); see Section 6.2.2.2 for further information on the radio interface rate. Logically speaking, the RA1/RA1′ function consists of two parts, the conversion of the intermedi-

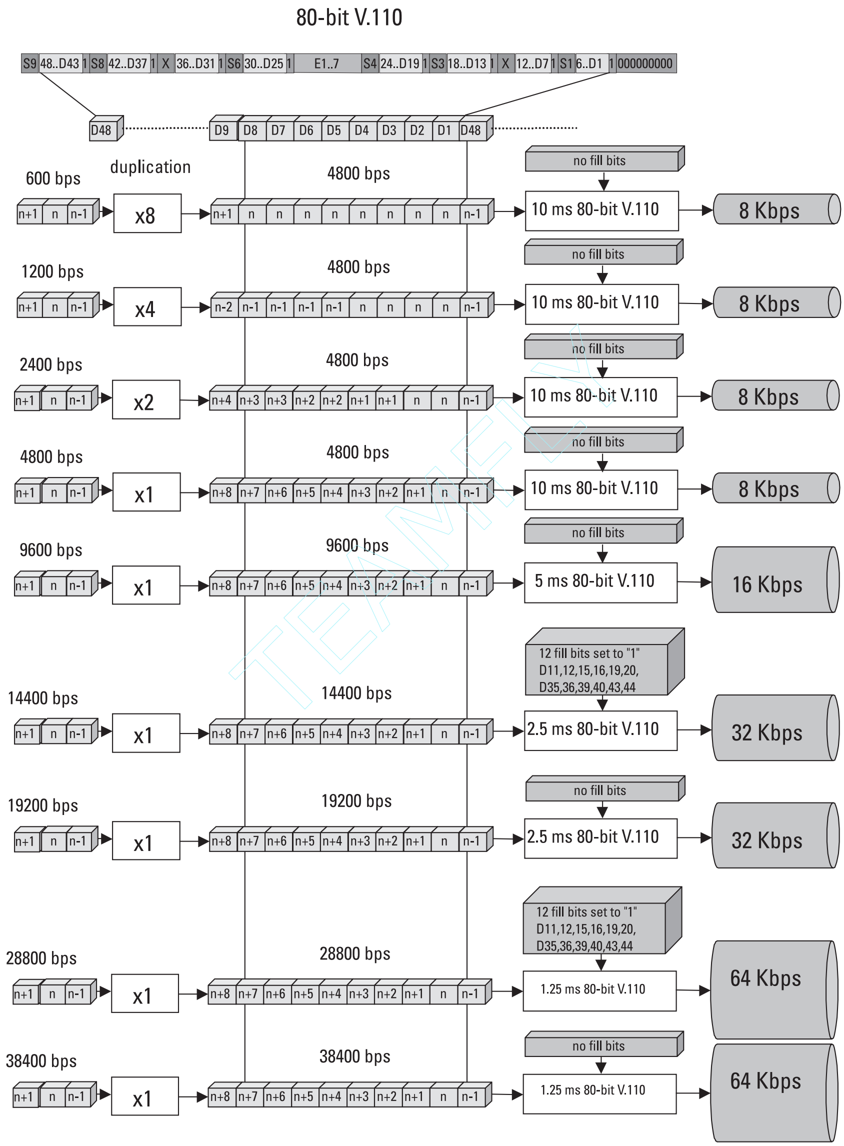


Figure 6.21 RA1, bit repetition and filler bits.

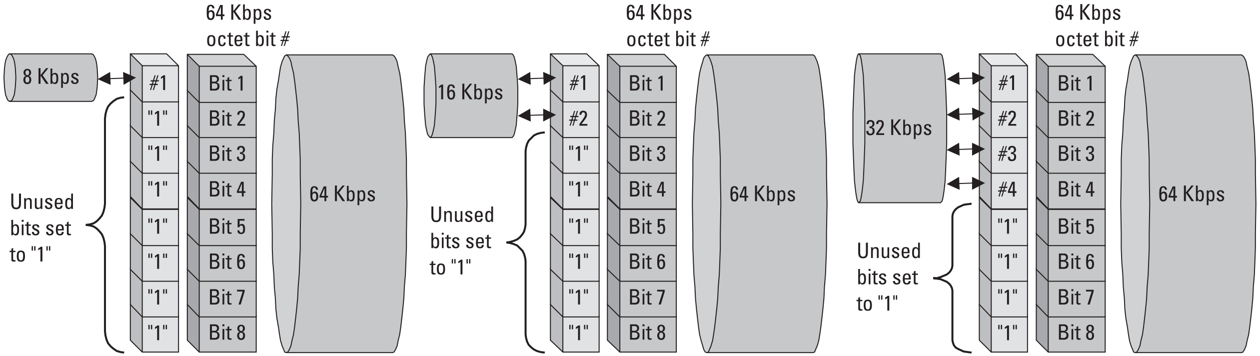


Figure 6.22 RA2, octet filling, intermediate rate ⇔ 64 Kbps.

ate rate to a synchronous rate, and the adaptation of the synchronous rate to the radio interface rate.

Individual cases have to be differentiated (e.g., whether, for 14.4 Kbps, a 290-bit frame with 16 Kbps or an 80-bit V.110 frame with 32 Kbps is to be used). In the former case, the network side is affected: the 290-bit frame is only used here for saving bandwidth, and the RA1/RAA′ adaptation takes on the adaptation to the radio interface. The latter case is valid for the mobile station. Here, the bandwidth is not as significant and the RA1/RA1′ adaptation is used.

The following sections result from and explain the peculiarities of the RA1/RA1′ function.

TCH/F4.8 and TCH/F9.6 Data Rates ≤ 38.4 Kbps

The basis for the adaptation is the 80-bit V.110 frame, which thus sets the limit at 38.4 Kbps. The RA2 function of the RA1/RA1′ adaptation is done without the remote transcoder at the BTS (i.e., without transmission of the V.110 frames to data TRAU frames), or at the mobile station using the S interface. It takes over the adaptation between the 64-Kbps interface and the intermediate rate. With a multislot configuration, the split/combine function can be present in between, and can, for example, distribute or join 38.4 Kbps in four TCH/F9.6 channels.

With the transcoder being remote to the BTS, RAA adaptation also has to be taken into account. Figure 6.23 illustrates all the cases mentioned above.

As an adaptation to the radio interface rate, 20 bits of the V.110 frame are added or removed—namely the 17-bit synchronization pattern and the 3 E bits, E1 to E3. The reduced frame then is synchronized to the frame timing of the air interface. For the nontransparent service, an unambiguous synchronization on the air interface is essential, because the E2 and E3 bits are used to count the four V.110 frames that compose an RLP frame. The synchronization mechanism is based on the definition of fixed start positions in the 26 TDMA framing period. All start positions are spaced by four TDMA frames, which is obviously the length of the RLP frame on the air interface (456 coded bits with interleaving not considered). The first start position is in frame 0, the second in frame 4, and so on. At frame 12 the first jump occurs, because the SACCH is fixed mapped there. All together, six start positions exist: 0, 4, 8, 13, 17, and 21. For simplicity they are identified as blocks B0 to B5. Each block spans 22 TCH frames, which is equal to the time spread of 456 coded bits after interleaving. A doubled distance of start positions applies to the data traffic channel TCH/F4.8. The

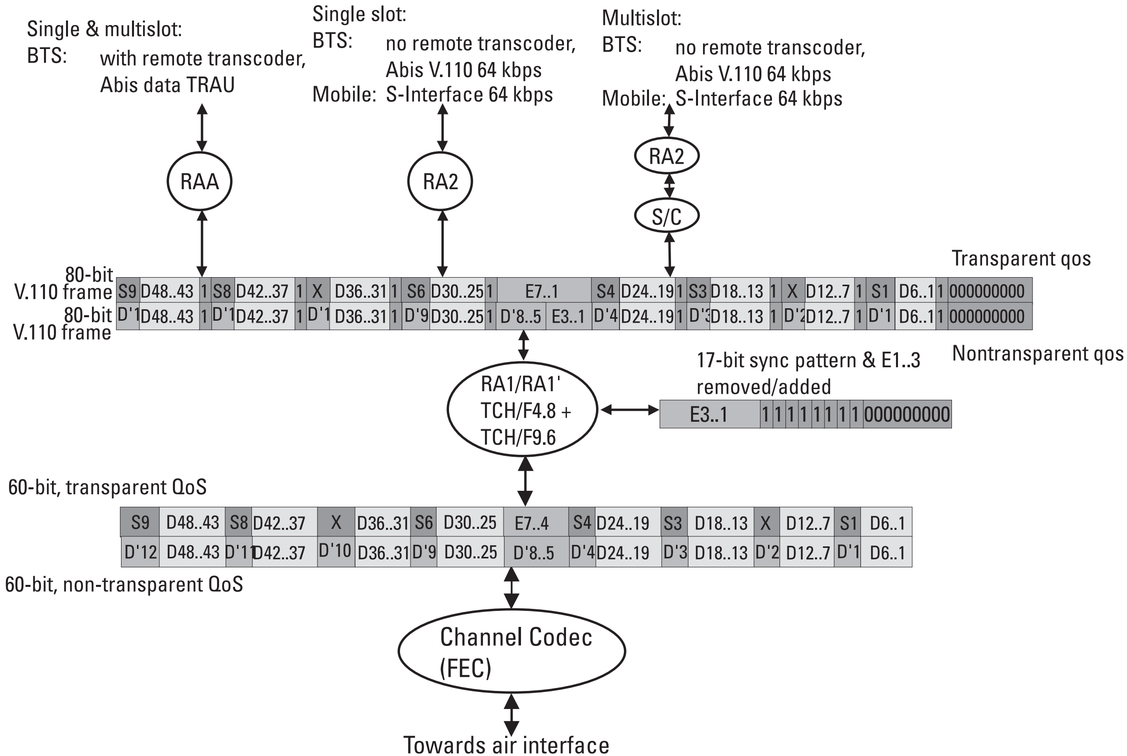


Figure 6.23 RA1/RA1′, TCH/F4.8, and TCH/F9.6, ≤ 38.4 Kbps.

1/3 channel coding is responsible for the larger period of eight TDMA frames (see also Section 6.2.2). Only the even-numbered blocks are used in this case, B0, 2, and 4 (Figure 6.24).

Exception. 14.4 Kbps and 28.8 Kbps. For the data rates of 14.4 Kbps and 28.8 Kbps, 12 of 48 payload bits in the 80-bit V.110 frame are padded (see Section 6.4.3). If the data rates are achieved by channel combining of either TCH/F9.6 or TCH/F4.8, a conflict appears with the definition of the padding from the split/combine function, because only the highest numbered substream is allowed to be padded.

Example. 14.4 Kbps is either realized with 2 × 7.2 Kbps or 1 × 9.6 Kbps plus 1 × 4.8 Kbps and the highest substream padded at a rate of 4.8 Kbps. Now, on both sides of the connection, different rate adaptations can be active, RA1/RA1′ on the mobile side (80-bit V.110 frame at 32 Kbps) and RA1′/RAA′ on the network side with 290-bit frames at 16 Kbps. Obviously, only one padding rule is acceptable, the padding of the split/combine function.

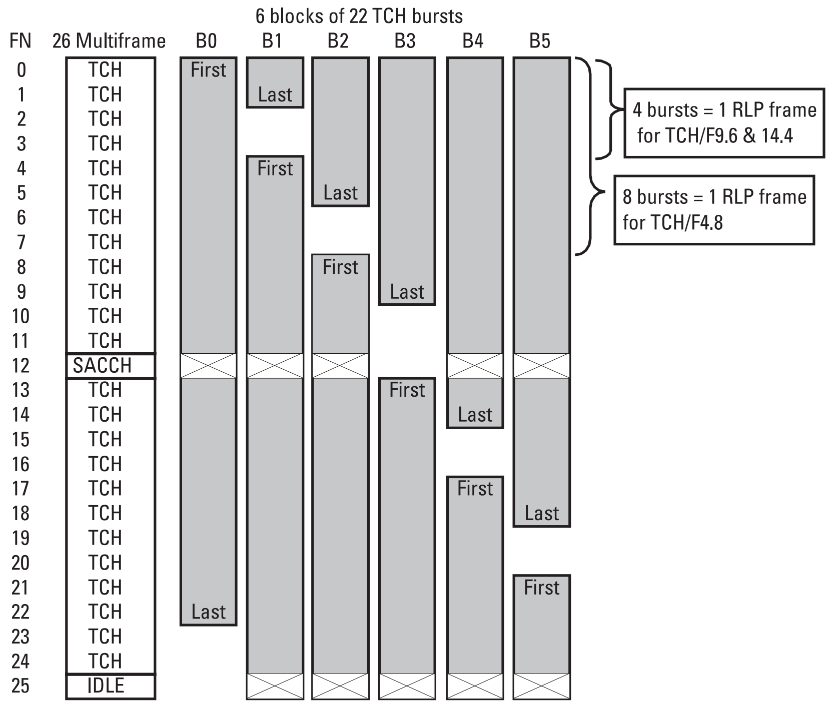


Figure 6.24 Synchronization of V.110 multiframe (RLP-Frame) and TDMA multiframe.

To avoid the padding conflict, four 80-bit V.110 frames with each 36-bit payload are combined to three 80-bit frames with each 48-bit payload. This is illustrated in Figure 6.25.

TCH/F14.4, Data Rates < 38.4 Kbps

The RA1/RA1′ function performs the adaptation between eight 80-bit V.110 frames with $8 \times 36 = 288$ bit payload and the 290-bit frame used on the air interface. At the network side, the RA1/RA1′ function is not used; here only the RA1′/RAA′ function together with the split/combine function and E-TRAU is applicable (Figure 6.26).

Out of the 80-bit V.110 frame not all status bits are mapped to the 290-bit frame of the air interface—the S bits are lost. The 5-bit GSM NIC code is directly mapped from the appropriate V.110 frame. Its coding can be derived from Section 6.4.2.

5 × TCH/F9.6 and 48-Kbps Data Rate, 64-Kbps Intermediate Rate

Two 32-bit V.110 frames (each of 500-μs duration) are mapped to one 60-bit frame on the air interface (5-ms duration). The split/combine function serves five substreams (Figure 6.27).

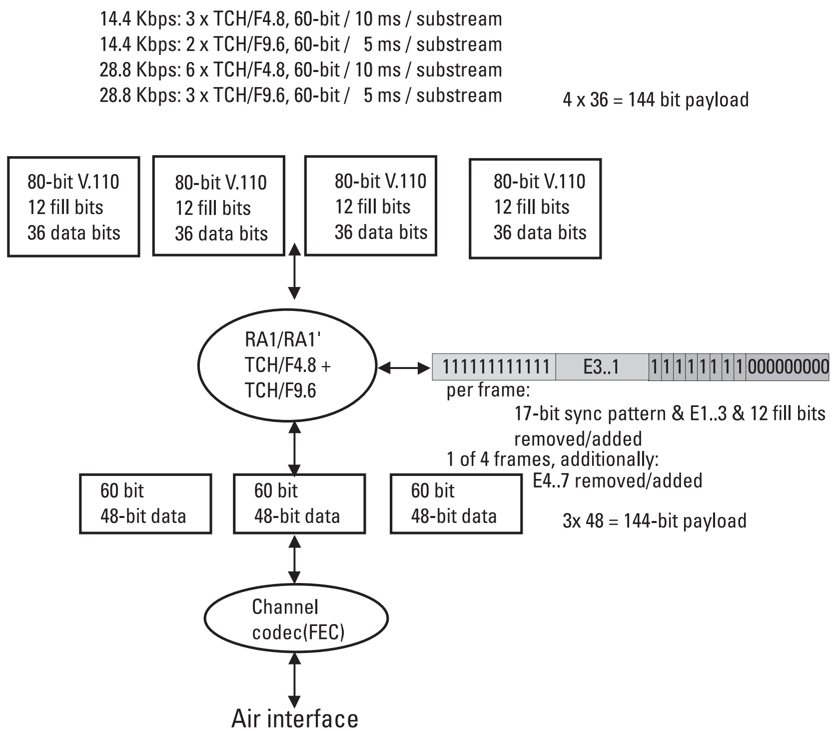


Figure 6.25 RA1/RA1′ exception for 14.4 Kbps and 28.8 Kbps.

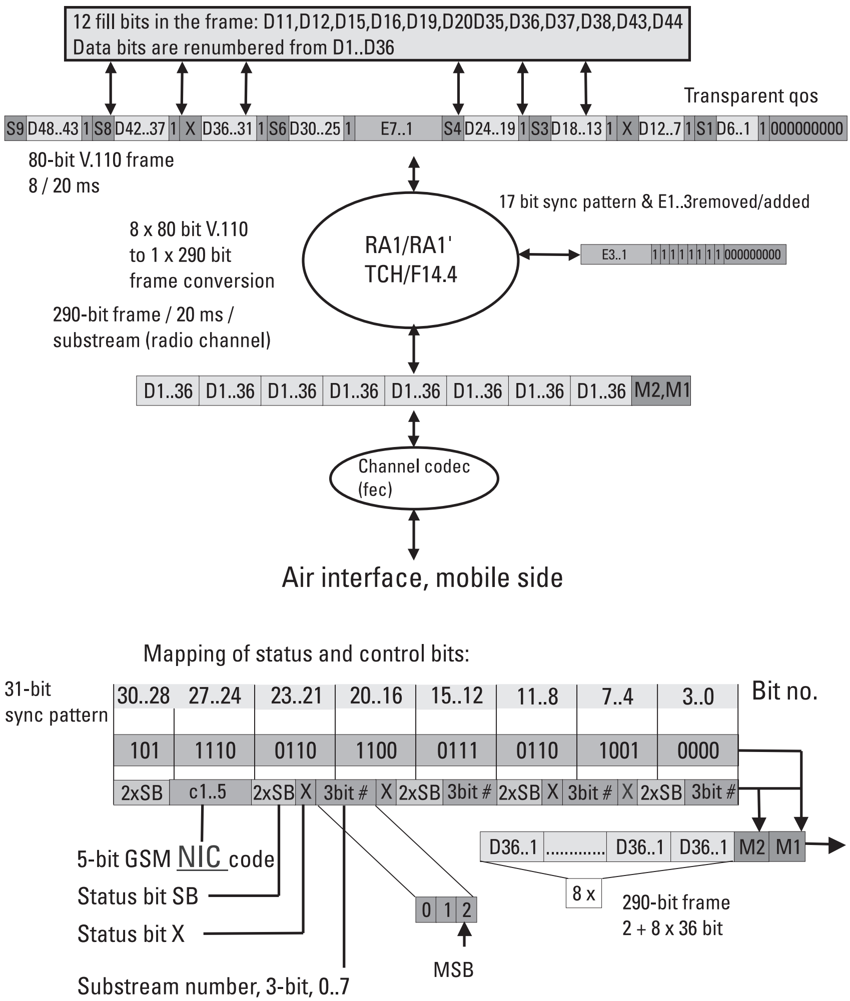


Figure 6.26 RA1/RA1′, TCH14.4, data rate < 38.4 Kbps.

4 × TCH/F14.4 and 48-Kbps Data Rate, 640-Kbps Intermediate Rate

In this case 12 32-bit V.110 frames are adapted to one 290-bit frame on the air interface and transmitted into the substreams 0 to 2. The highest substream (number 3) is padded and carries the payload of eight 32-bit V.110 frames (see Section 6.2.1) (Figure 6.28).

This kind of adaptation is described here to provide a complete documentation. In practice, it is of very limited relevance, as only synchronous

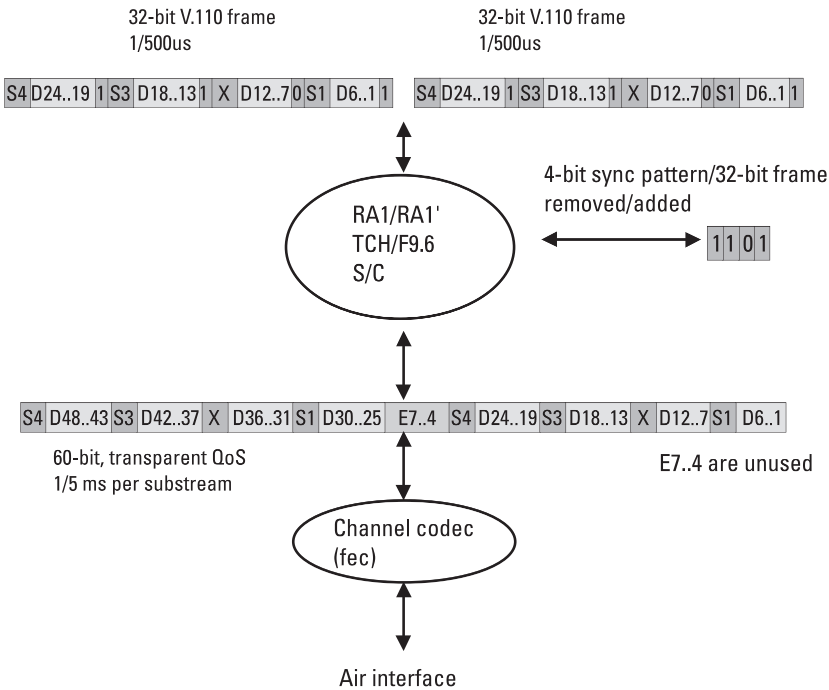


Figure 6.27 RA1/RA1′, TCH/F9.6, 48 Kbps, synchronous.

connections are supported. If used at all, it will exist only on the mobile side.

5 × TCH/F9.6 and 56-Kbps Data Rate, 64-Kbps Intermediate Rate

Again this is a rate adaptation that has no real relevance in deployed systems. It supports only transparent synchronous connections, and because the A interface is limited to 4 × 16 Kbps, the split/combine function must be located in the BSS to handle five substreams over the air interface.

Furthermore, the 60-bit frame is modified to increase the payload from 48 to 56 bits. This is feasible for synchronous connections, as the S, X, and parts of the E bits are not used. Eight S and X bits, as well as four of 7 E bits are reduced to 4 T bits in this modified 60-bit frame (Figure 6.29).

4 × TCH/F14.4 and 56-Kbps Data Rate, 64-Kbps Intermediate Rate

The 56 data bits of the 64-bit V.110 frame are byte oriented mapped to the 290-bit frame of the air interface and transmitted into the substreams 0 to 2. The highest substream (number 3) is padded and carries only 256 payload bits of which 32 bits are padded (see Section 6.2.1) (Figure 6.30).

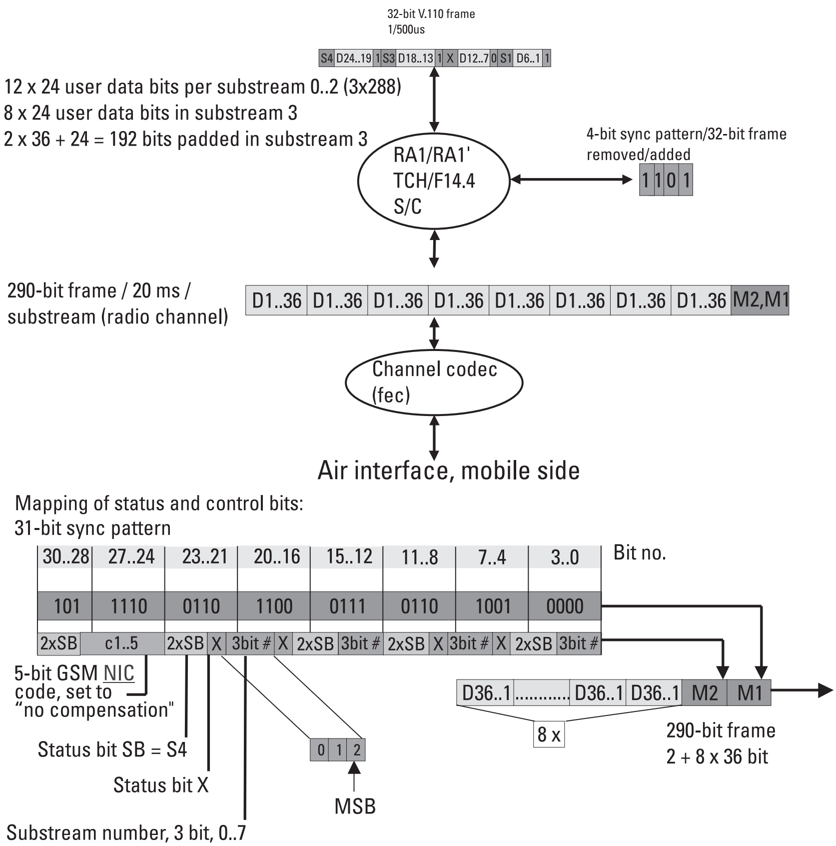


Figure 6.28 RA1/RA1′, TCH/F14.4, 48 Kbps, synchronous.

Again, this kind of adaptation is described here to provide a complete documentation. In practice it is of no relevance, as only synchronous connections are supported. If used at all, it will exist only on the mobile side.

64 Kbps

For the mobile customer, the 64 Kbps could be a desired data rate. With HSCSD, it will remain a dream. Besides the required high number of bundled channels (6 × 9.6 or 5 × 14.4), the split/combine function on the network side must reside within the BTS. With 64-Kbps user data rate, there is no space for any data framing overhead with substream numbering and synchronization on the ISDN-based network links of 64 Kbps up to the MSC. In the same manner, there is no space for the RLP protocol overhead

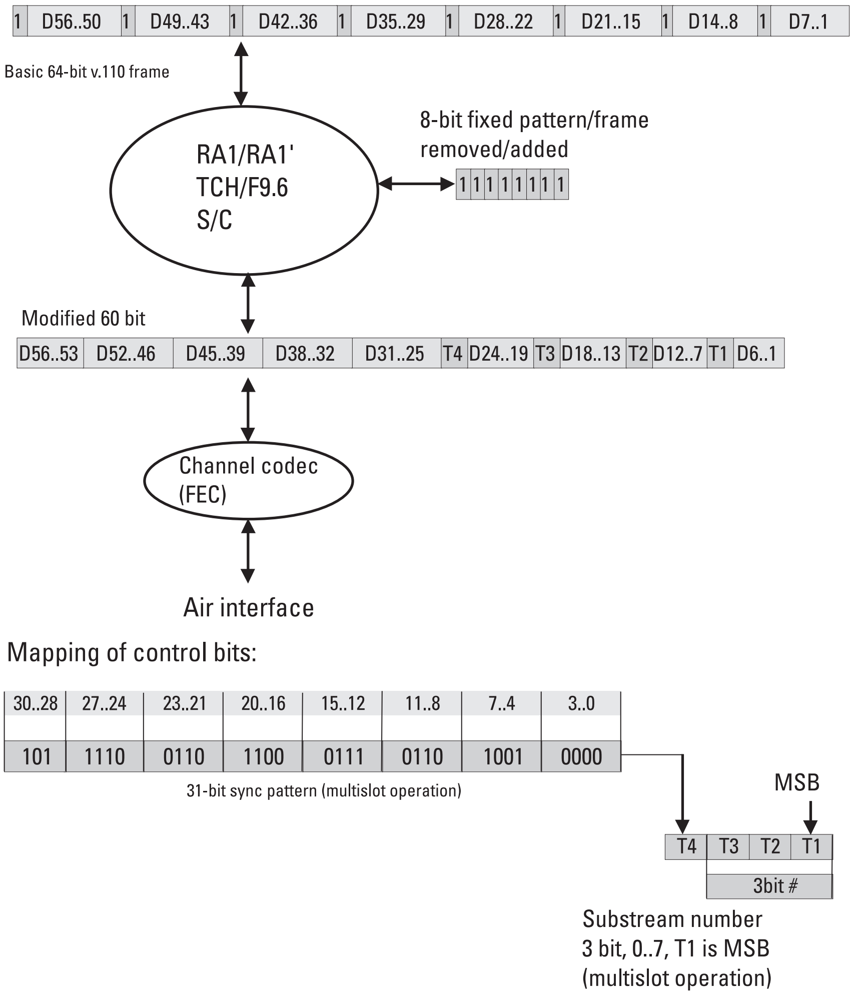


Figure 6.29 RA1/RA1′, TCH/F9.6, 56 Kbps, synchronous.

and thus, only transparent connections are feasible. The two adaptation schemes are given in Figures 6.31 and 6.32.

6.4.3.6 RA1′, Synchronous ⇔ Radio Interface

The RA1′ function performs direct adaptation between any synchronous interface data rate based on the V.110 frame (for example, as output of the RA0 function) and the air interface frame and its 60 or 290-bit frames. The functional description can be derived from the appropriate figures of

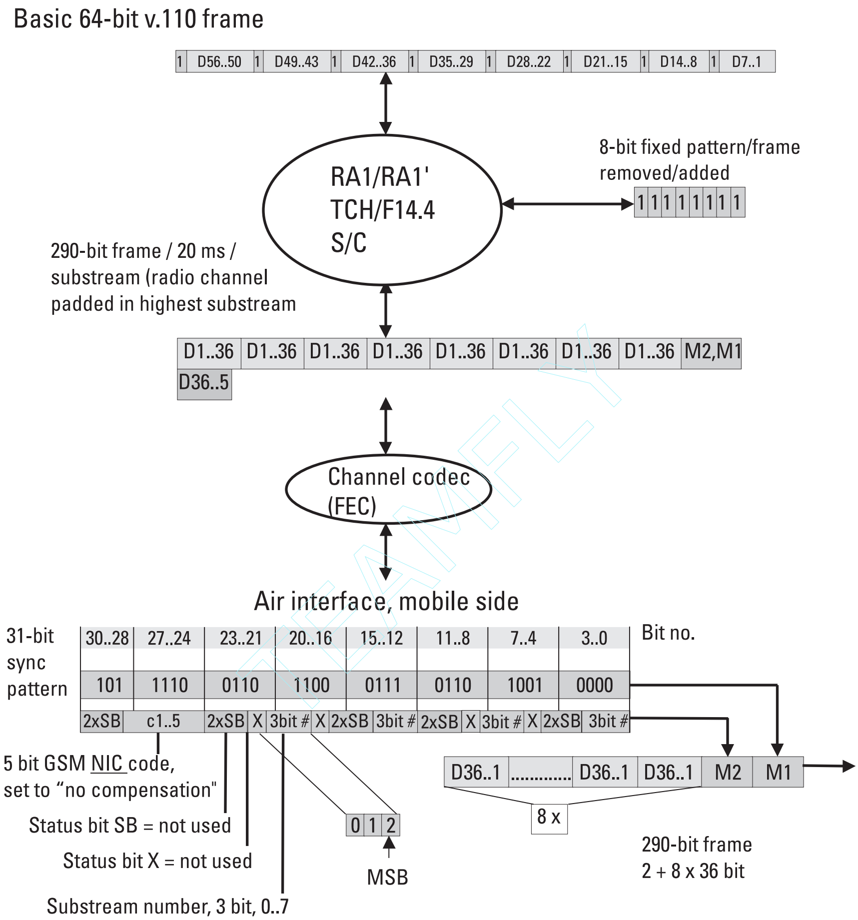


Figure 6.30 TCH/F14.4, 56 Kbps, synchronous.

the last section about RA1/RA1′, 6.4.3.5, from the part above the channel codec.

6.4.3.7 TRAU Rate Adaptations

Data TRAU Frame

RA1′/RAA′, TCH/F14.4, Radio Interface ⇔ E-TRAU. For the 14.4-Kbps channel coding, conversion between the E-TRAU frame and the 290-bit air interface frame is done. The E-TRAU frame is the 290-bit frame of the air interface plus a 17-bit synchronization pattern and a 13-bit control field, added as header (Figure 6.33).

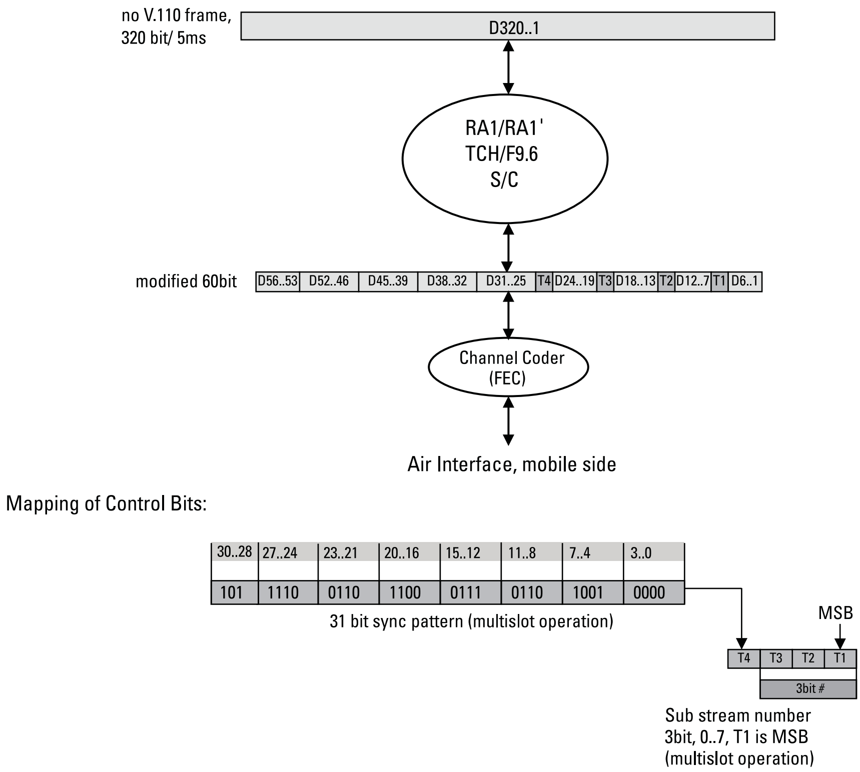


Figure 6.31 RA1/RA1′, 64 Kbps, 6 × TCH/F9.6, synchronous, transparent only.

RAA′, TCH/F14.4, E-TRAU ⇔ A-TRAU. The RAA′ function applies to the TCH/F14.4 channel and performs an adaptation between the A-TRAU and E-TRAU frame. Both frames are detailed in the next paragraphs.

RAA″, TCH/F14.4, A-TRAU ⇔ Synchronous Rate. At the MSC side of the A interface, the RAA″ function is used to adapt between synchronous data rates (e.g., as output of the RA0 function) and the A-TRAU frame, which is detailed below.

A-TRAU. First of all, the 17-bit synchronization pattern of the A-TRAU is completely different from the 17-bit synchronization pattern of the V.110 frame:

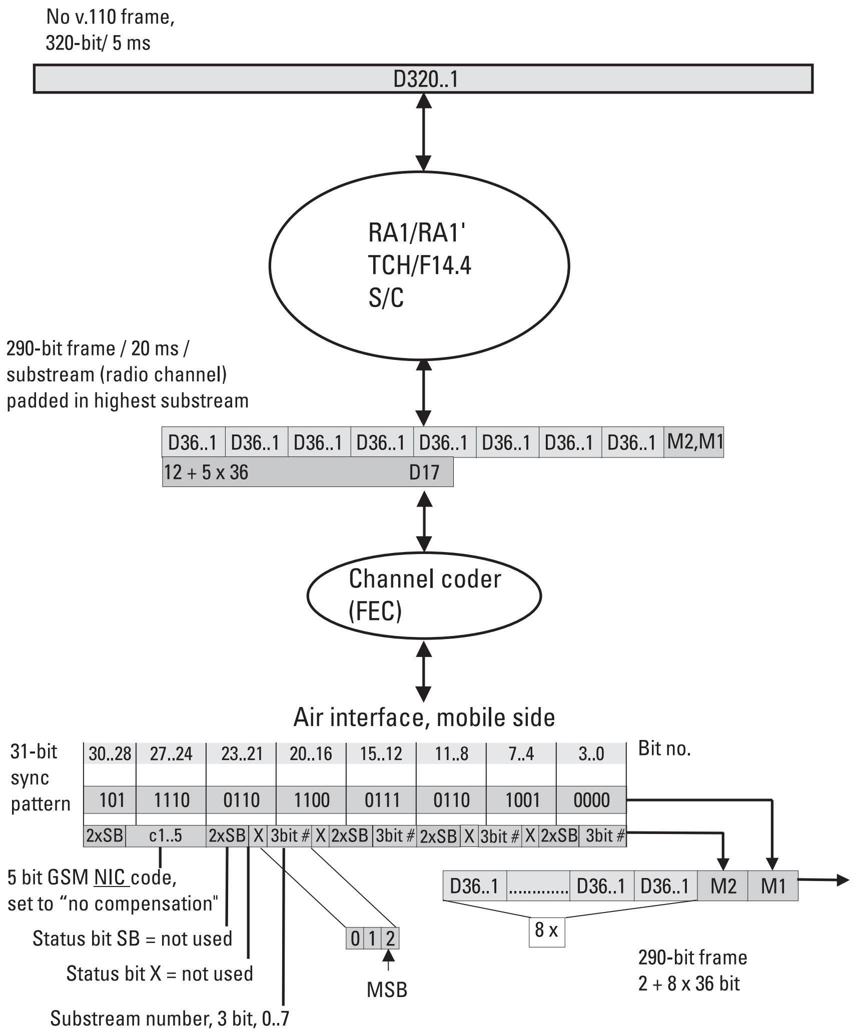


Figure 6.32 RA1/RA1′, 64 Kbps, 5 × TCH/F14.4, synchronous, transparent only.

- V.110 frame: 1 octet zero (00000000) + 1. Bit in each octet 1;
- A-TRAU frame: 2 octets zero (0000000000000000) + 1. Bit of the next octet 1.

Figure 6.34 shows this difference in in graphical form.

The remaining part of the A-TRAU is composed of eight blocks of each 36-bit payload. Within the payload part, the 17-bit A-TRAU synchronization

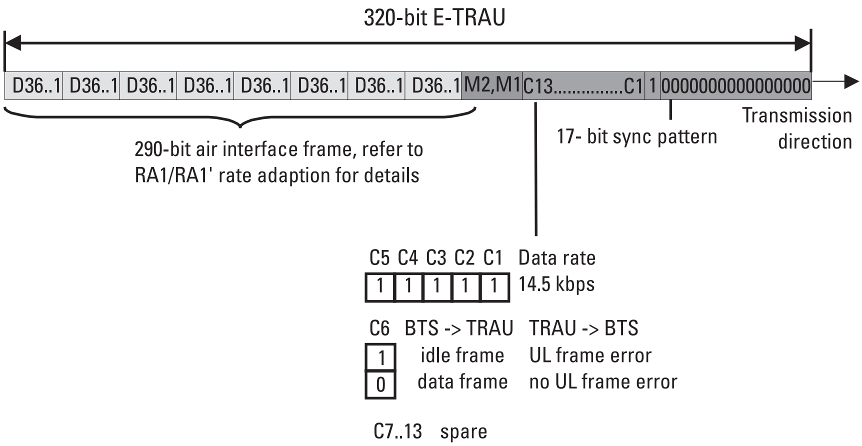


Figure 6.33 320-bit E-TRAU.

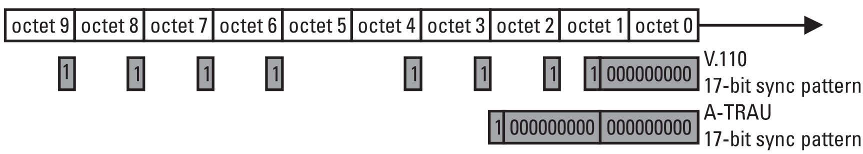


Figure 6.34 Comparing A-TRAU and V.110 synchronization pattern.

pattern could occur occasionally. To avoid this, a substitution method is used that replaces each instance of eight consecutive zeros by a zero sequence position (ZSP) field.

The ZSP field consists of the following:

- 5-bit address field: points to the next ZSP field in the A-TRAU. The size of the 32-bit offset is sufficient because the last part of the frame from bit D32 to D36 is less than 8 bits large;
- 1 continuation bit: indicates if the ZSP field is the last one in the block or not;
- 2 fix 1 bits to protect the ZSP to become a 0 byte.

Each payload block starts with a ZSP field. It indicates if a ZSP field exists in the block or not. If a ZSP field is present, then the first one follows immediately after the Z bit and replaces the data bits D1 to D8. The full substitution rule is given in Figure 6.35.

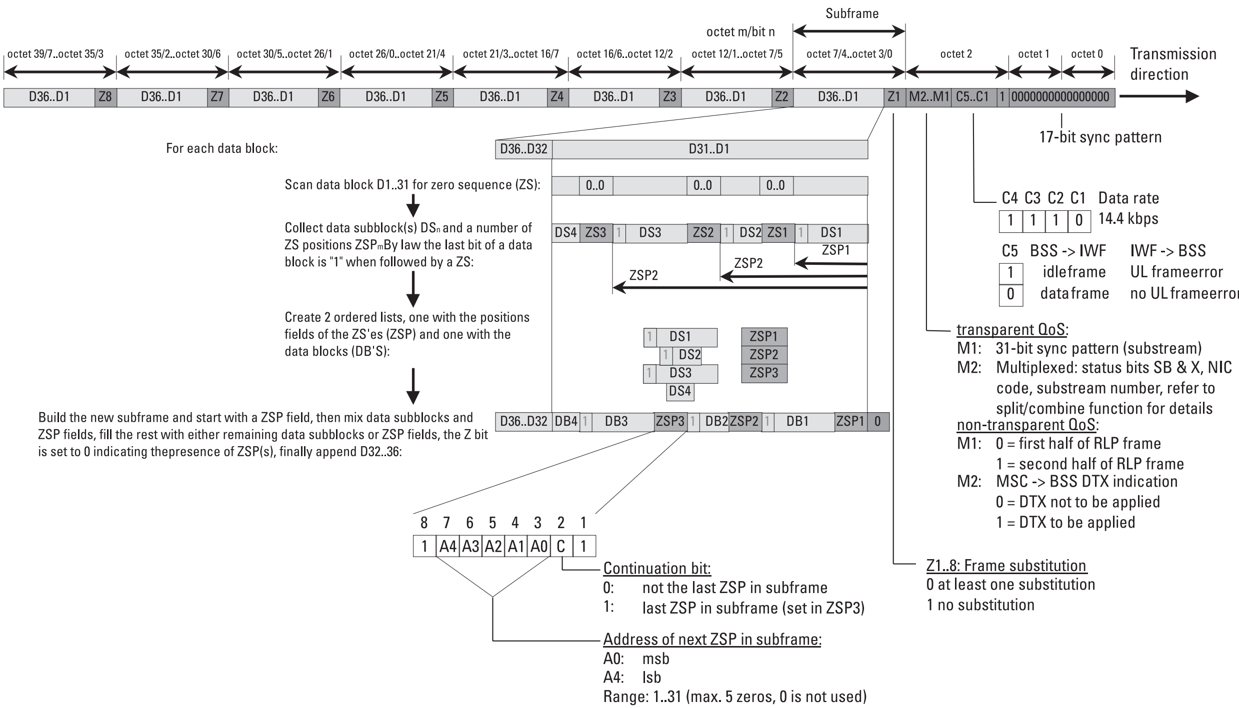


Figure 6.35 Null octet substitution with ZSP fields.

E-TRAU. The E-TRAU, or extended TRAU, is equal to the 290-bit air interface frame (see also Sections 6.2.1 and 6.4.3), plus a 17-bit synchronization pattern (see Figure 6.33).

In contrast to the A-TRAU frame, no zero sequence substitution is used and no protection exists to avoid the presence of the sync pattern within the payload part. The GSM recommendations propose to use first the standard TRAU frame for synchronization and then to switch to the E-TRAU. This approach is too weak in practice, and thus, vendor specific implementations exists to solve the problem by means of substitution mechanisms as done within the A-TRAU.

Standard Data-TRAU, 320 Bit. The standard 320-bit data-TRAU frame is constructed from four 80-bit V.110 frames by removing the leading zero octet of the sync pattern from each V.110 frame, then combining the remaining part and finally adding a 15-bit control word and a 17-bit sync pattern (Figure 6.36).

The 320-bit data-TRAU consists of the following:

- 17-bit sync pattern;
- 15 control bits, C1..15;
- 4×63 bit payload from the V.110 frame;
- A fix "1" in each payload octet to inhibit the occurrence of the sync pattern in the payload part.

6.4.4 RLP

The RLP is the core of all nontransparent data connections and provides Layer 2 functions such as error recognition and frame repetition. The protocol belongs to the high level data link control (HDLC) family and is very similar to the X.25 LAPB. One essential difference is the shorter and fixed frame length of the RLP: 240 bits with 192 payload bits (24 bytes) with TCH/F4.8 and 9.6 channels and 576 bits with 528 payload bits (66 bytes) with TCH/F14.4 channels. By comparison, LAPB has a configurable frame length of 16, 32, 64, 128, 256, 512, 1,024, 2,048, or 4,096 bytes.

The Layer 2 connection with the RLP secures the transmission path in the GSM PLMN from the interworking function of the G-MSC to the mobile station as a terminal adapter.

The numbering of the HSCSD subchannels is supported with version 2 of the protocol. Up to four subchannels can be used (4×16 Kbps in a 64-Kbps A interface channel).

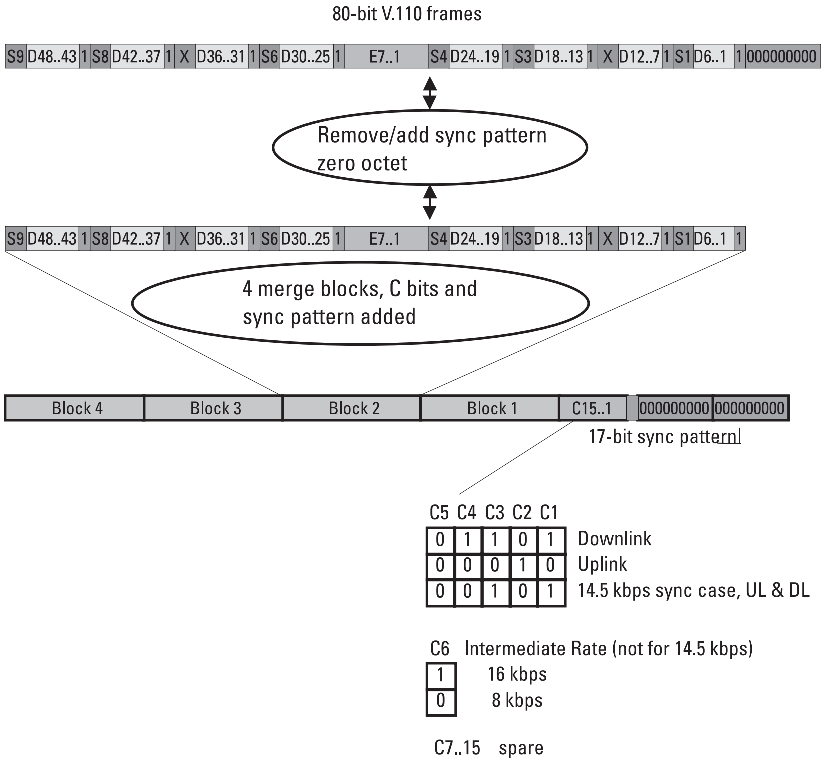


Figure 6.36 320-bit data TRAU.

Each RLP frame consists of a header, a data part, and a frame check sequence (FCS), which is formed as a checksum with the header and the data. The FCS received is compared in the receiver with the calculated FCS; if they are not identical, there are two possibilities for repeating:

1. Selective repetition of an individual frame (selective reject ARQ, automatic repeat request);
2. Repetition of all frames from a given sequence number (go-back ARQ).

For this purpose, each frame carries two 9-bit sequence numbers, a transmission sequence number and a reception sequence number. The transmitter does not have to wait for the confirmation of each individual frame; it can transmit further frames without confirmation up to a preset window size. Without HSCSD (or to be more precise, channel bundling), the maximum window size is 61 frames. With HSCSD and version 2, the value is increased and is then dependent on the number of subchannels. The value proposed by GSM for four subchannels is 240 frames; the maximum set value is 493. In all, this corresponds to an amount of pending frames on the transmission path of nearly 10 seconds (493 * 20 ms).

6.4.4.1 Connection States

The RLP has two connection states:

1. *Asynchronous disconnect mode:* The state with an inactive Layer 2 connection; only unsecured frame exchange is possible (UI frames). ADM is the basic state after initializing and the final state after an error. Each connection side can always send U-frames autonomously. The XID command is used to set connection parameters (e.g., window size and timer).
2. *Asynchronous balance mode:* By transmitting an SABM frame and receiving a UA response, the Layer 2 connection is established and the ABM mode activated. Asynchronous means that each connection side can set up the connection. However, the mobile station should preferably setup the Layer 2 connection in the GSM PLMN. Numbered and unnumbered frames can be exchanged.

6.4.4.2 Frame Groups

There are three different groups of frames: U-frame, S-frame, and I+S-frame.

U-Frame. Unnumbered frames—frames without sequence counters:

- SABM (set asynchronous balance mode): initiates the Layer 2 connection (ADM ⇒ ABM) or triggers a reset of an existing connection (ABM ⇒ ABM)—for example, after N2 repeat request failures (N2 default value is 6); all puffers are deleted in this case;
- UA (unnumbered acknowledge): positive response to SABM or DISC;
- DISC (disconnected): ends the Layer 2 connection (ABM ⇒ ADM);
- DM (disconnected mode): status sent to the called station that the ADM mode is active, or an indication that a change to the ABM mode is not possible (negative response to SABM);
- NULL (null information): transmitted when there is no payload;
- UI (unnumbered information): unsecured data exchange in the ADM and ABM modes;

- XID (exchange identification): exchange and fixing of protocol parameters (e.g., window size, timer, and V.42bis parameter);
- TEST (connection test): the called station must reply to the transmission of a test frame with another test frame;
- REMAP: enables switching between the 240-bit and 576-bit frames. This is necessary when the channel configuration changes (e.g., during a handover with a change from 3 × TCH/F9.6 to 2 × TCH/F14.4).

S-Frame. Supervisory frame—frames with sequence counters but an empty data part:

- RR (receive ready): frames up to the counter status N(R) − 1 have been received correctly; the called station can transmit the frame N(R) and all further frames up to the window size;
- REJ (reject): frames have been received with gaps or errors; all the frames from the counter status N(R) must be repeated. Simultaneously signals correct reception of all frames up to N(R) − 1 inclusive;
- RNR (receive not ready): frames cannot currently be received; simultaneously, signals correct reception of all frames up to N(R) − 1 inclusive.
- SREJ (selective reject): a single frame [N(R)] is requested for repetition out of a block of frames, which has been received but not confirmed. Any confirmation of receipt of the frames N(R) − 1 is not linked with this.

I+S-Frame. Information and supervisory frame: frames with supervisory type, sequence counter and payload, only used in ABM mode. The supervisory types, RR, REJ, RNR, and SREJ, are identical to those described above.

6.4.4.3 Frame Setup, 240 Bits and 576 Bits

A 240-bit RLP frame is transmitted with four 80-bit V.110 frames. The E2 and E3 bits are used for numbering and lose their original significance. The two E bits are not transmitted on the air interface. However, the RLP frame integrity is guaranteed by the synchronization of the RLP frames with the TDMA frame structure.

Similarly, the 576-bit RLP frame is transmitted with two 320-bit A-TRAU frames. In this case, the M1 bit is used for numbering. Note that the M1 bit does not carry the 31-bit frame sync pattern in this case.

This frame setup is shown in Figure 6.37.

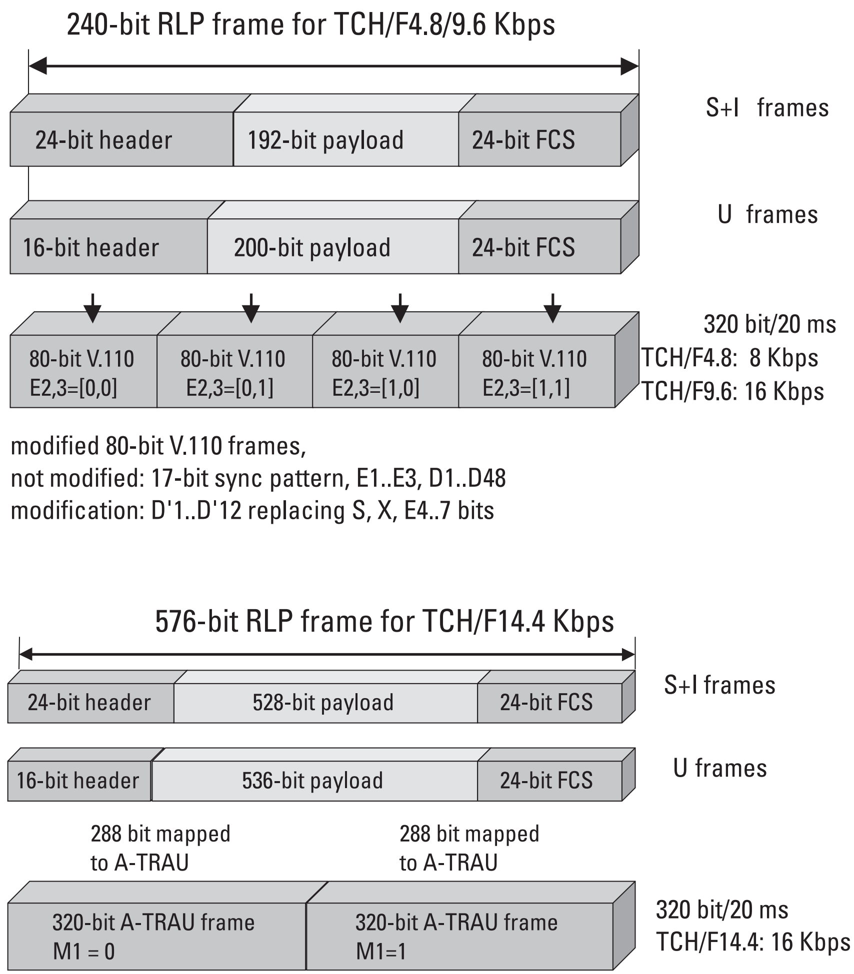


Figure 6.37 RLP frame setup and V.110/A-TRAU mapping.

6.4.4.4 Frame Types and Frame Header

Figures 6.38 and 6.39 show header coding of the frame types U and I+S, respectively.

6.4.5 Layer 2 Relay

L2R is the direct user of the RLP protocol and offers the following range of functions in two variations:

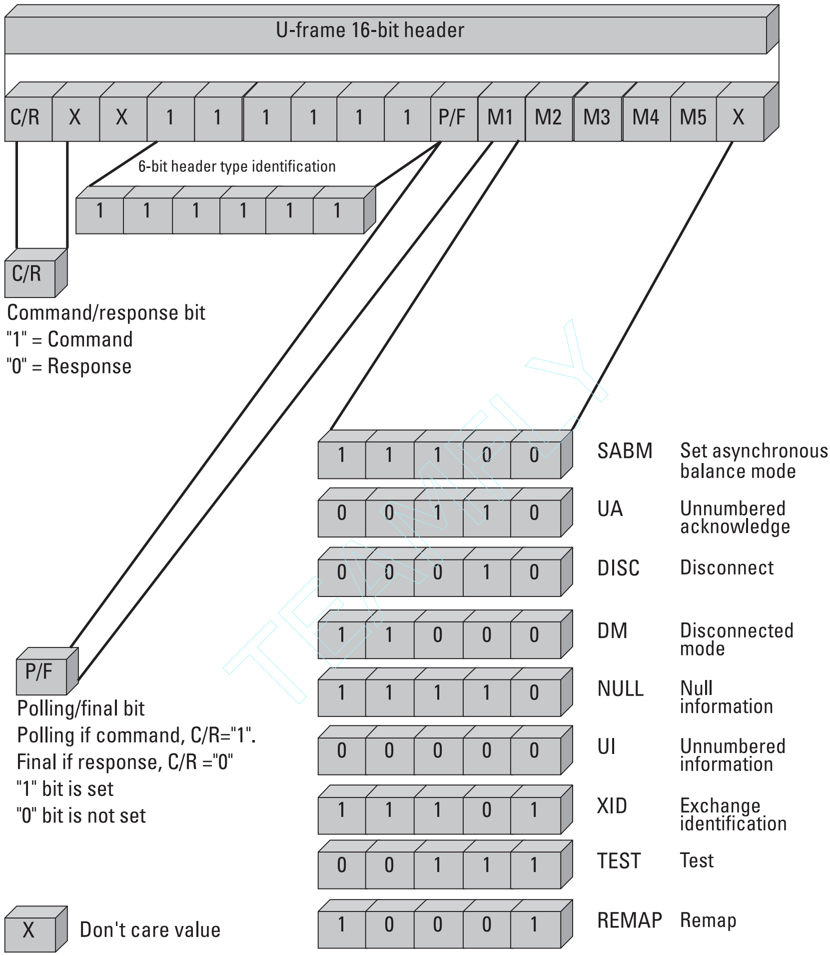


Figure 6.38 U-frame header.

- *Character-Oriented Protocol:* The L2R-COP connects the modem or terminal interfaces with the RLP protocol and thus realizes transparent transmission features with a lower bit error rate than the transparent data service based on the V.110 frame. However, longer delays and inconstant data rates remain unchanged as disadvantages. The L2R-COP V.42bis offers data compression as an option. The above-mentioned disadvantages then become even worse.
- *Bit-Oriented Protocol:* The L2R-BOP takes over the relay function between the X.25 and FAX LAPB protocol and the RLP protocol.

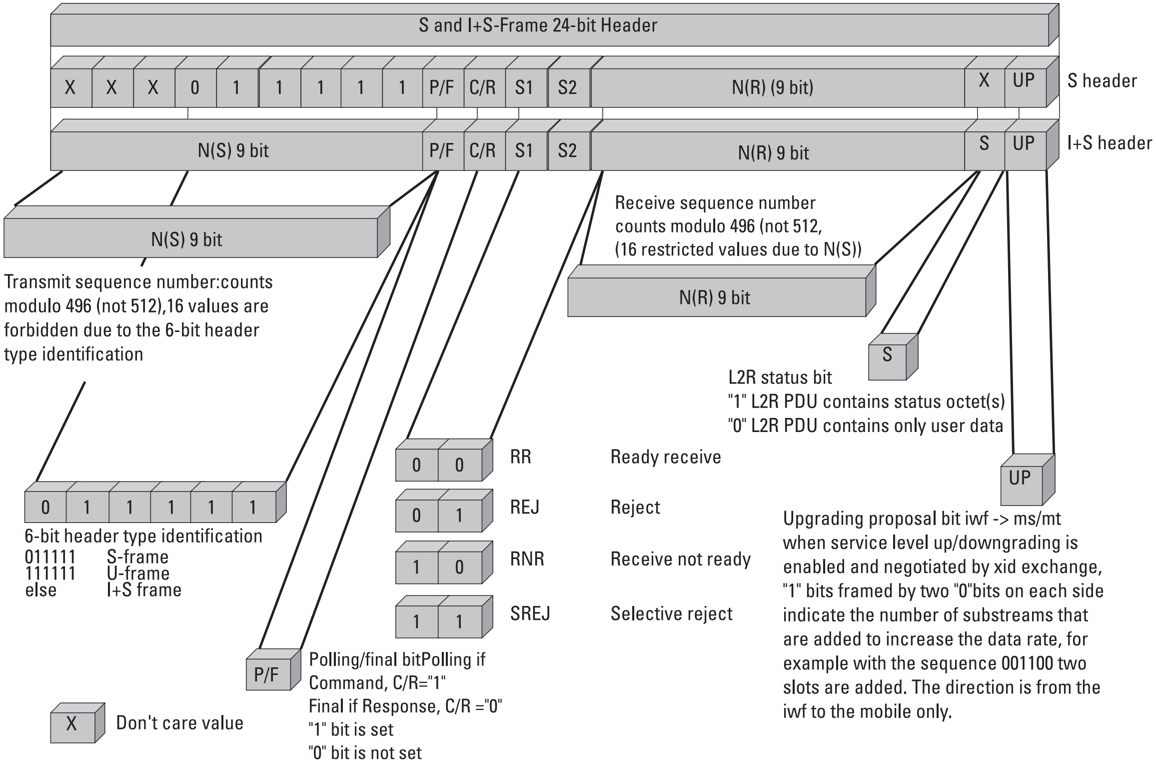


Figure 6.39 S, I + S-frame header.

Its main task is the adaptation of the variable LAPB frame length to the fixed frame size of the RLP protocol.

6.4.5.1 L2R-COP

The L2R packs data status information from the modem interface into L2R-COP PDUs (Figure 6.40). Status information can, for instance, be the outband HW (hardware) signals from a V.24 interface (compare S and X bits of the V.110 frame) or software flow control signals such as Xon/Xoff or break as a "special event." Note that the status information is not transmitted with the S and X bits of the V.110 frame, but as a status octet in the L2R-PDU. The S and X bits of the V.110 frame are not available because they are used by the RLP protocol as 12 additional data bits (D′1. . .12) (see Section 6.4.4.3).

L2R-COP PDU

Depending on the RLP frame type, the L2R PDUs contain 24 or 66 octets (or bytes), which are made up either entirely of data or as a mixture of data and status octets (Figure 6.41). The greater part of the status octet consists of a pointer, which counts the data octets up to the next status octet or indicates that there are no further status octets in the PDU. In the RLP frame header (I & S+I frames), a bit is used to code whether there are any status octets in the PDU at all (see also Section 6.4.4.4). If there are, the first octet of the PDU must be a status octet.

A status octet can also terminate a PDU that is only partly filled with data and declare the rest to be empty. This function is important when

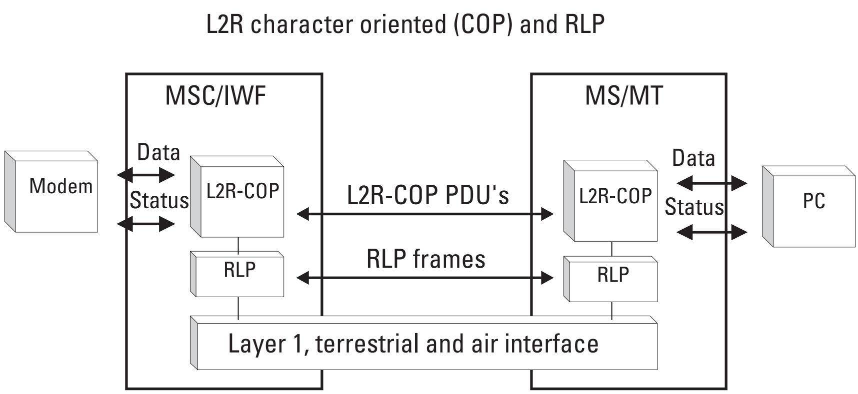


Figure 6.40 L2R-COP.

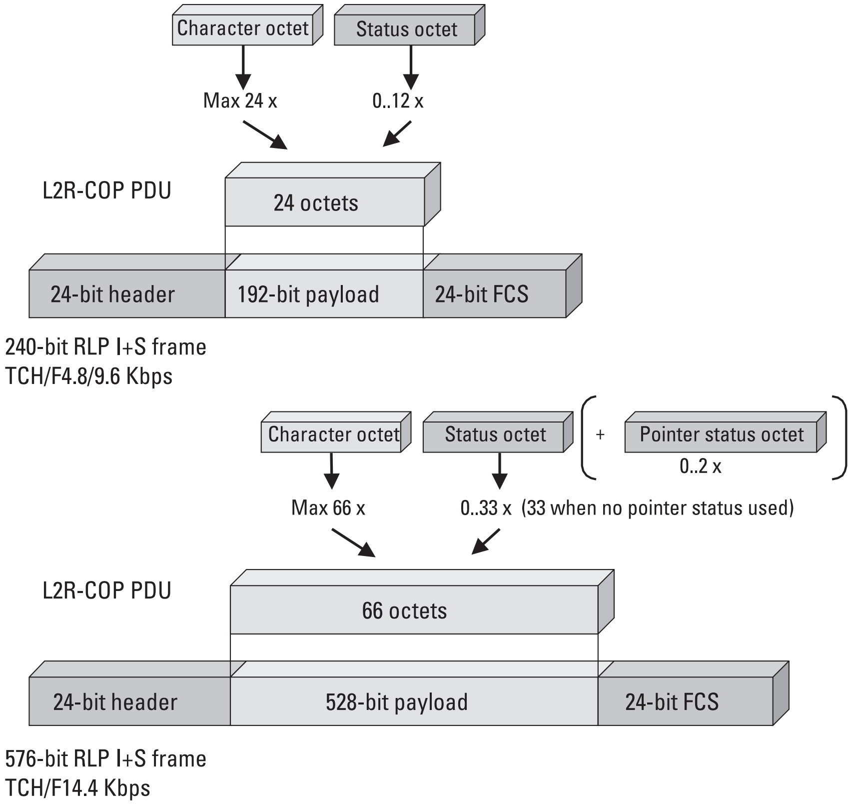


Figure 6.41 L2R-COP PDU.

no further data is available and the RLP can receive further PDUs (delay minimizing).

Data Byte or Data Character Octet

The composition is as follows (Figure 6.42):

- 7-bit data without parity bit, bit #8 is set at 0;
- 7-bit data with parity bit;
- 8-bit data;
- V.42bis compressed 8-bit coded SDUs.

The start and stop bits of the modem SDUs are removed (see Section 6.4.3.1).

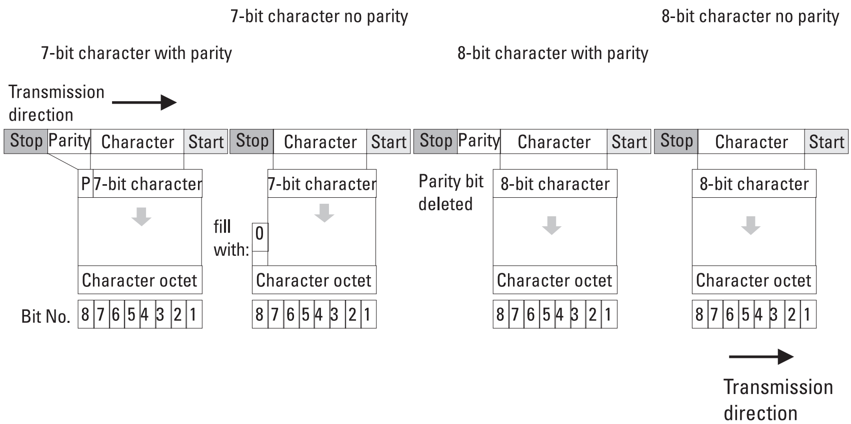


Figure 6.42 L2R-COP character octet.

Status Octet and Pointer Status Octet

With the introduction of the TCH/F14.4 channels, 66-byte PDUs became necessary. Consequently, the 5-bit pointer in the status octet became too small. For this reason, an additional pointer status octet was introduced. It directly follows a status octet if more than 23 data octets follow the pointer status octet to the next status octet. Figure 6.43 illustrates the coding of the two status octets.

L2R-COP Examples of PDUs

Here are four PDU examples (Figure 6.44):

1. 66-octet PDU with pointer status octet and data octet after the last status octet;
2. As 1, but frame terminated after last status octet;
3. 24-octet PDU, one status octet at the beginning and end, but no pointer status octet necessary.
4. 24-octet PDU without any status octet.

6.4.5.2 L2R-BOP

As mentioned at the start, the L2R-BOP is a bridging function between the LAPB protocol and the RLP protocol. Its basic task is the segmenting and reassembly of the mostly longer LAPB frames to the RLP frames of fixed length.

Table 6.9 shows a comparison of frame lengths.

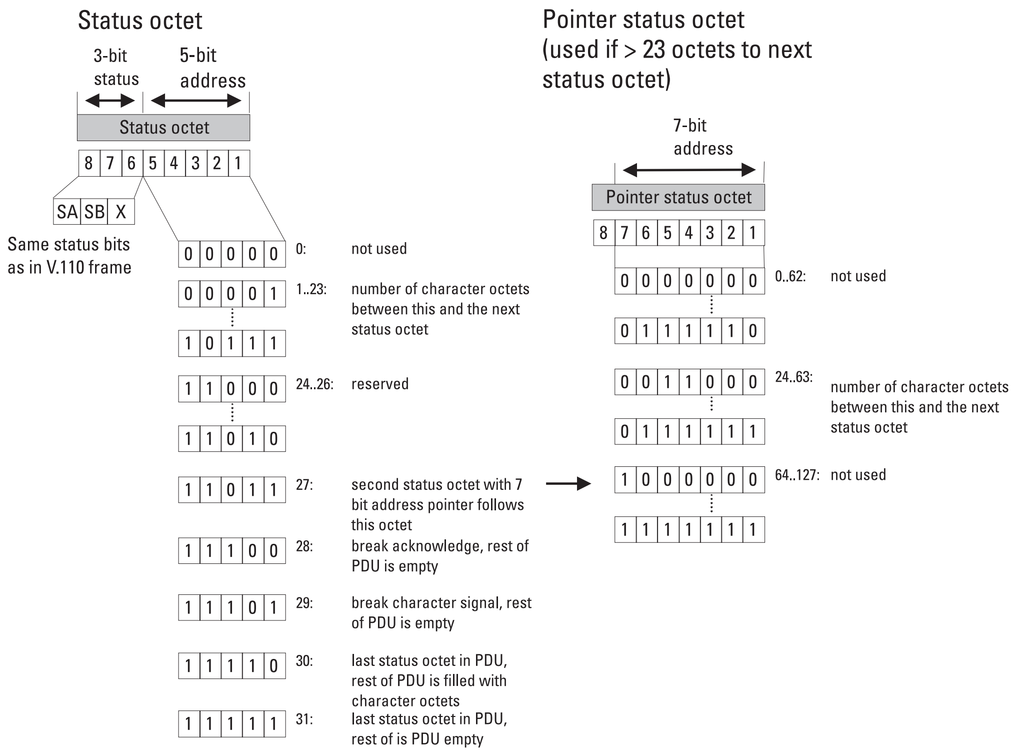


Figure 6.43 L2R-COP PDU: status octet and pointer status octet.

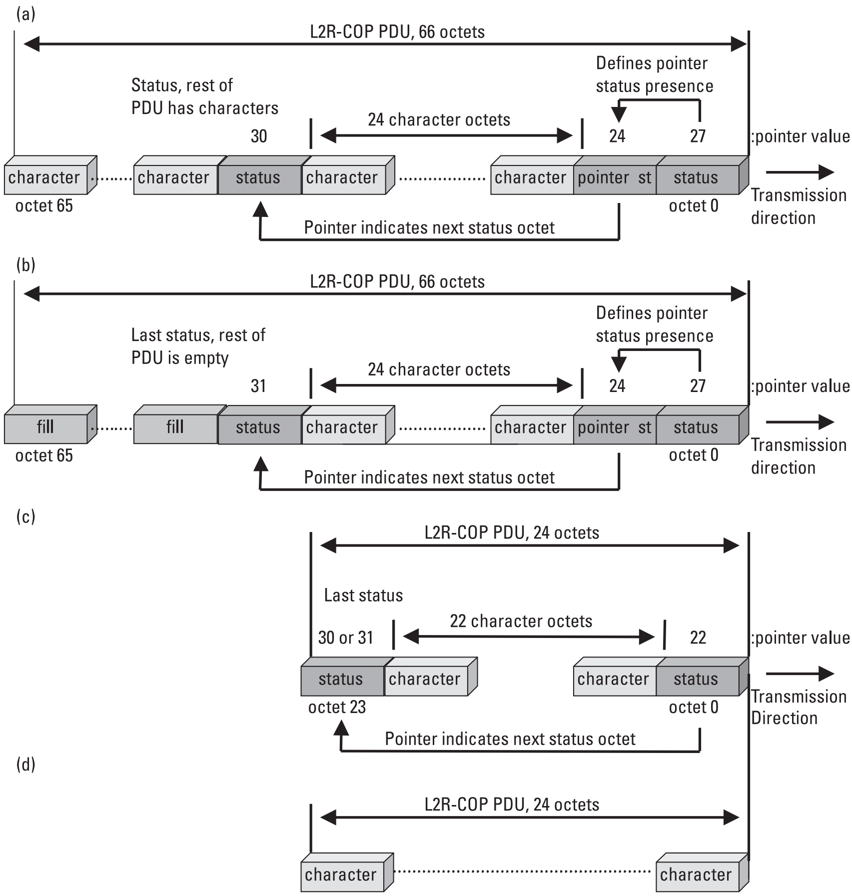


Figure 6.44 L2R-COP PDU examples.

L2R-BOP PDU

The composition of the L2R-BOP PDU is identical to that of the L2R-COP, apart from the renaming of the character octet as "user info octet."

Here, too, the PDU can either be completely filled with data or be a mixture of data and status octets. The status octet also has a 5-bit pointer, which counts the number of data octets to the next status octet or indicates that there are no further status octets in the PDU. LAPB-specific codings, which are presented in the status octet section below, are also defined.

As with the L2R-COP, there is also a code in the RLP frame header (I & S+I frames) that indicates whether there are any status octets in the PCU at all (see also Section 6.4.4.4). If so, the first octet in the PCU must be a status octet.

Table 6.9
Frame Length Comparison

LAPB 9 Data Field Sizes [Bytes]	Ratio LAPB/RLP RLP 24 Bytes	Ratio LAPB/RLP RLP 66 Bytes
16	0.67	0.24
32	1.33	0.48
64	2.67	0.97
128*	5.33	1.94
256	10.67	3.88
512	21.33	7.76
1,024	42.67	15.52
2,048	85.33	31.03
4,096	170.67	62.06

*LAPB default value

A status octet can also terminate a PDU that is only partly filled with data and declare the rest to be empty.

What is new in comparison with the L2R-COP is the possibility of the status octet indicating the number of valid data bits (1. . .8) in the previous user info octet. This enables a restructured orientation of the LAPB frames to the RLP frames (i.e., the latter do not have to be fixed in terms of byte limits), even if both protocol data fields are always integer multiples of bytes.

The L2R-BOP PDU is shown in Figure 6.45.

L2R-BOP Status Octet and Pointer Status Octet

The differences between the L2R-COP and the L2R-BOP status octet are as follows (Figure 6.46):

- The S and X status bits can transmit signals from either V interfaces (compare V.110 SA, SB, X-bits) or X interfaces (synchronous interfaces).
- Address value 29 (bits 1. . .5) terminates an LAPB frame and, with the 3 status bits (bits 6. . .8), codes the number of valid LAPB data bits in the last user info octet. The values 1. . .7 indicate the bit positions 1. . .7; the value 0 indicates bit position 8. If the status octet is not the last octet in the PDU, the next one follows in order to define the contents of the remaining PDU, either as empty (address value 31) or as completely filled with user info octets (address value 30).

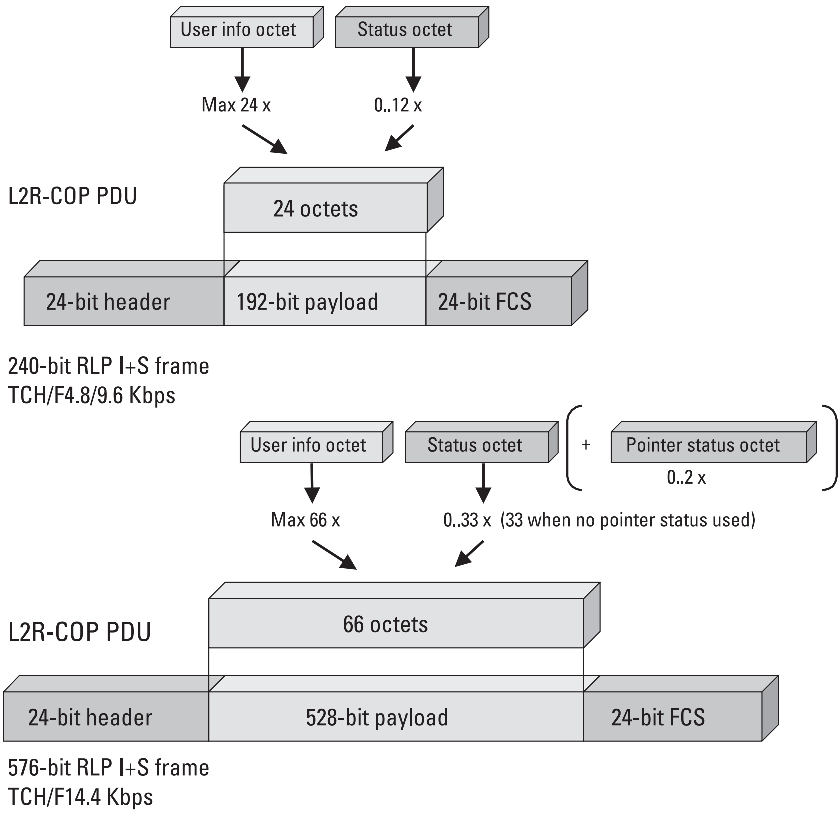


Figure 6.45 L2R-BOP PDU.

- Address value 28 cancels the LAPB frame. If the status octet is not the last octet in the PDU, then, as before, the next one follows in order to define the contents of the remaining PDU, either as empty (address value 31) or as completely filled with user info octets (address value 30).
- LAPB control frames or XID frames for parameter exchange are characterized with a status octet positioned first in the PDU and the address value 0. The subsequent two octets then have the following significance:
  - First octet: LAPB connection number, always 0;
  - Second octet: LAPB connection control information: 1 = Connect, 2 = Reset, 3 = Disconnect, 4 = Loss of LAPB interframe fill; or XID frame: 5 = XID Request, 6 = XID Acknowledge; value range 7. . .255 is unused;
  - The remaining octets are unused and declared to be reserved.

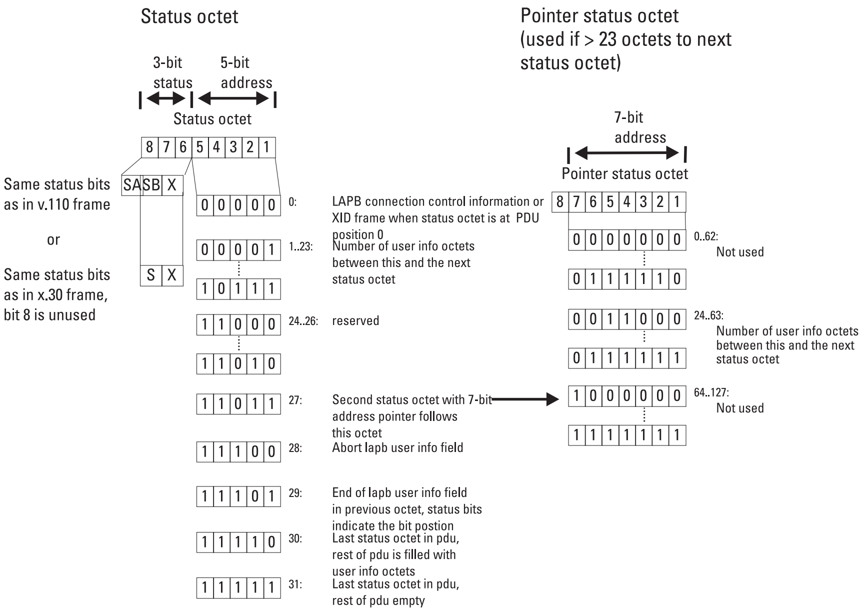


Figure 6.46 L2R-BOP PDU: status octet and pointer status octet.

Reference

[1] Heine, G., *GSM-Signalisierung verstehen und praktisch anwenden,* FRANZIS Verlag.

List of Acronyms

3GPP	Third Generation Partnership Project
8-PSK	8 symbol phase shift keying
AA	anonymous access
A-Bit	acknowledgment request bit (LLC)
ABM	asynchronous balanced mode
ACCH	associated control channel
ADM	asynchronous disconnected mode
AGCH	access grant channel
AM	amplitude modulation
APN	access point name (reference to a GGSN)
ARFCN	absolute radio frequency channel number
ARQ	automatic repeat request
AT-Command	attention-command
AuC	authentication center
BCCH	broadcast control channel
BG	border gateway
BIB	backward indicator bit
BS_CV_MAX	maximum countdown value to be used by the mobile station (countdown procedure)
BSC	base station controller
BSIC	base station identity code
BSN	block sequence number (RLC)/backward sequence number (SS7)
BSS	base station subsystem

BSSAP	base station subsystem application part
BSSGP	base station system GPRS protocol
BSSMAP	base station subsystem mobile application part
BTS	base transceiver station
BVCI	BSSGP virtual connection identifier
C/R-bit	command/response bit
CBCH	cell broadcast channel
CC	call control
CCCH	common control channel
CDMA	code division multiple access
CDR	Charging/Call Data Record
CEPT	Conférence Européenne des Postes et Télécommunications
CG	Charging Gateway
CGF	charging gateway function
CHAP	challenge handshake authentication protocol (PPP)
CS-X	coding scheme (1-4)
CV	countdown value
DCS	digital communication system
DHCP	dynamic host configuration protocol
DL	downlink
DLR	destination local reference
DNS	domain name system
DPC	destination point code
DRX	discontinuous reception
DTAP	direct transfer application part
DTX	discontinuous transmission
ECSD	enhanced circuit switched data (HSCSD + EDGE)
EDGE	Enhanced Data Rates for Global Evolution
EGPRS	Enhanced General Packet Radio Service
E-GSM	Extended GSM (GSM 900 in the Extended Band)
EIR	equipment identity register
ERAN	EDGE Radio Access Network
ESN	electronic serial number (North American Market)
ETSI	European Telecommunications Standard Institute
FACCH	fast associated control channel
FBI	final block indicator
FCCH	frequency correction channel
FCS	frame check sequence (CRC-check)

FDD	frequency division duplex
FDMA	frequency division multiple access
FIB	forward indicator bit
FISU	fill in signal unit
FMC	fixed mobile convergence
FN	frame number
FR	full-rate or frame relay
FRMR	frame reject
FSN	forward sequence number
GEA	GPRS encryption algorithm
GGSN	gateway GPRS support node
GMM	GPRS mobility management
G-MSC	gateway MSC
GMSK	Gaussian minimum shift keying
G-PDU	T-PDU + GTP-header
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
GTP	GPRS tunneling protocol
HDLC	high level data link control
HLR	home location register
H-PLMN	home PLMN
HR	half-rate
HSCSD	high-speed circuit switched data
HTTP	hypertext transfer protocol
I+S	information + supervisory
IAM	initial address message (ISUP-ISDN User Part)
IETF	Internet Engineering Task Force (www.ietf.org)
IHOSS	Internet Hosted Octet Stream Service
IMEI	international mobile equipment identity
IMSI	international mobile subscriber identity
IMT-2000	International Mobile Telecommunications for the year 2000
IP	Internet Protocol
IPCP	Internet Protocol Control Protocol
ISDN	Integrated Services Device Network
ISP	Internet service provider
ITU-T	International Telecommunication Union-Telecommunication Sector
L2TP	Layer 2 Tunneling Protocol

LA	location area
LAC	location area code
LAI	location area identification
LAPB	link access procedure balanced
LAPD	link access protocol for the ISDN D-channel
LCP	link control protocol (PPP)
LLC	logical link control
LPD	link protocol discriminator
LSB	least significant bit
LSSU	link status signal unit
MAC	medium access control
MCC	mobile country code
MCS-X	modulation coding scheme (1-9)
MIME	multipurpose Internet mail extensions
MIN	mobile identity number (North American market)
MM	mobility management
MNC	mobile network code
MOC	mobile originating call
MRU	maximum receive unit (PPP)
MS	mobile station
MSB	most significant bit
MSC	mobile services switching center
MSU	message signal unit
MT	mobile terminal or mobile terminating
MTC	mobile terminating call
MTP	message transfer part
NCC	network color code
NCP	network control protocol (PPP)
NI	network indicator
N-PDU	network-protocol data unit (IP-Packet, X.25-Frame)
NS	network service
NSAPI	network service access point identifier
NSS	Network Switching Subsystem
OMC	operation and maintenance center
OPC	originating point code
OSI	open system interconnection
OSP	octet stream protocol
P/F-Bit	polling/final-bit
PACCH	packet associated control channel

PAD	packet assembly disassembly
PAGCH	packet access grant channel
PAP	password authentication protocol (PPP)
PBCCH	packet broadcast control channel
PCCCH	packet common control channel
PCM	pulse code modulation
PCN	Personal Communication Network
PCS	Personal Communication System
PCU	packet control unit
PD	protocol discriminator
PDCH	packet data channel
PDP	packet data protocol
PDTCH	packet data traffic channel
PDU	protocol data unit or packet data unit
PLMN	Public Land Mobile Network
PNCH	packet notification channel
POP	post office protocol
PPCH	packet paging channel
PPP	point-to-point protocol
PRACH	packet random access channel
PSTN	Public Switched Telephone Network
PT	protocol type (GTP or GTP′)
PTCCH	packet timing advance control channel
PTCCH/D	packet timing advance control channel/downlink direction
PTCCH/U	packet timing advance control channel/uplink direction
PTM	point to multipoint
P-TMSI	packet TMSI
PTP	point to point
QoS	quality of service
RA	routing area
RAC	routing area code
RACH	random access channel
RAI	routing area identification
RAND	random number
REJ	reject
RFC	request for comment (Internet standards)
R-GSM	railways-GSM

RLC	radio link control
RNC	radio network controller
RNR	receive not ready
RNS	radio network subsystem
RR	radio resource management
RR (LAPD)	receive ready
RRBP	relative reserved block period
SABM(E)	set asynchronous balanced mode (extended)
SACCH	slow associated control channel
SACCH/MD	SACCH multislot downlink (related control channel of TCH/FD)
SAPI	service access point identifier
SCH	synchronization channel
SDCCH	stand alone dedicated control channel
SDMA	space division multiple access
SDU	service data unit
SGSN	serving GPRS support node
SI	service indicator
SIF	signaling information field
SIM	subscriber identity module
SIO	service information octet
SLC	signaling link code
SLR	source local reference
SLS	signaling link selection
SLTA	signaling link test acknowledge
SLTM	signaling link test message
SM	session management
SMS	Short Message Service
SMSCB	Short Message Services Cell Broadcast
SMS-G-MSC	SMS gateway MSC (for short messages destined to mobile station)
SMS-IW-MSC	SMS interworking MSC (for short messages coming from mobile station)
SMTP	simple mail transfer protocol
SNDCP	subnetwork dependent convergence protocol
SNMP	simple network management protocol
SNN	SNDCP N-PDU number flag
SN-PDU	segmented N-PDU (SN-PDU is the payload of SNDCP)

SPC	signaling point code
SRES	signed response
SSN	send sequence number
SUERM	signal unit error rate monitor
TA	timing advance
TAI	timing advance index
TBF	temporary block flow
TCAP	transaction capabilities application protocol
TCH	traffic channel
TCH/FD	traffic channel/full-rate downlink
TCP	transmission control protocol
TDD	time division duplex
TDMA	time division multiple access
TE	terminal equipment
TFI	temporary flow identity
TI	transaction identifier
TID	tunnel identifier
TLLI	temporary logical link identifier
TLV	tag/length/value notation
TMSI	temporary mobile subscriber identity
TQI	temporary queuing identifier
TRAU	transcoding rate and adaption unit
TS	time slot
TSC	training sequence code
UA	unnumbered acknowledgment
UDP	user datagram protocol
UI	unnumbered information (LAPD)/unconfirmed information (LLC)
UL	uplink
UMTS	Universal Mobile Telecommunication System
USF	uplink state flag
UTRAN	UMTS Terrestrial Radio Access Network
UWC	Universal Wireless Convergence (Merge IS-136 with GSM)
VLR	visitor location register
V-PLMN	visited PLMN
VPN	virtual private network
XID	exchange identification
XOR	exclusive-or logical combination

About the Authors

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