NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

GSM Call Demonstrator

Graduation Project
EE 400

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. And above, I thank God for giving me stamina and courage to achieve my objectives.

ABSTRACT

It's easy to make successful call with good quality no interference and many services are offered in the global system for mobile communication, anyone who starts thinking about the features of this system or the process he will be sure that to create system such GSM system need many of principles, equipment and hard work to come with such system with all features to be accepted in the worldwide.

In this project we discussed the global system for mobile communication structure with details to each components it contain then we discussed the call process which include two main process which are transmission process and reception process with full details of the conception and the parameters of each process such as speech coding, channel coding, interleaving and modulation in the transmission process and the reverse of this processes in the reception process.

Interface in the global system for mobile communication (GSM) is also covered in this project; we discussed two type of interface in GSM system

We start with the radio interface which is the interface between the mobile stations and the fixed infrastructure; it is one of the most important interfaces of the GSM system. with explanation to conception, component and work principles then we discussed the second type of interface in GSM which is the Air-interface which is the central interface of every mobile system, with explanation to the structure of Air-interface which include the physical characteristics of the system, the conception and the principle of work for Air-interface also covered in this project.

INTRODUCTION

In the beginning of the 1980s several different systems for mobile communications were developed in Europe. The need for a common system that allowed roaming between countries was early recognized. In 1982 a number of European countries created a new standardization organisation called ``Groupe Speciale Mobile'' (GSM). The mandate of this group was to develop a standard to be common for the countries that created it. In 1988 the GSM was included in the European Telecommunication Standards Institute (ETSI), and the standards developed by GSM thus became standards for all telecommunication administrations in Europe.

The main work with the GSM took place from 1988 - 1990 and resulted in 12 series of specifications that in great detail specified the inner workings of GSM. In 1990, when phase 1 of the specifications was finished, there were three dominating automatic systems for mobile communications in the world :

- American AMPS from 1984, with networks in the US.
- British TACS from 1985, with network in Britain.
- Nordic NMT from 1981, with networks in the nordic countries.

Unlike these systems, the GSM is a fully digital system, allowing both speech and data services and allowing roaming across networks and countries. These features made GSM a very popular system, not only in european countries but also elsewhere. The term GSM has been chosen as a trademark for the system, meaning ``Global System for Mobile communications'', whereas the group within ETSI working with the standards has been renamed SMG (Special Mobile Group). Today GSM is the largest system for mobile communications in the world, and exist on all continents. From 1995, the specifications of GSM has moved into phase 2.

Like all modern mobile networks, GSM utilizes a cellular structure The basic idea of a cellular network is to partition the available frequency range, to assign only parts of that frequency spectrum to any base transceiver station, and to reduce the range of a base station in order to reuse the scarce frequencies as often as possible, one of the major goals of network planning is to reduce interference between different base stations.

GSM transmitter and receivers Process will start with the primary functions of a MS is to convert the analog speech information into digital form for transmission using digital signal, its include main three steps which are sampling, quantization, and coding and then the procedure will continue to the receiver side, the transmission process and the reception process are discussed with details in the second and the third chapters.

Last chapter divided to two sections the first section discussed the Radio interface, the radio interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces of the GSM system, this section include also Frequency allocation, GSM Channel Structure, Structure of TDMA Slot with a Frame, Frequency Correction Burst and more

The second section discussed the Air-interface, in this section the Structure of the Air-Interface in GSM, Physical Versus Logical Channels, Logical-Channel Configuration, Signaling on the Air-interface, with details and more.

1. SYSTEM ARCHITECTUTE of GSM

1.1 Overview

Like all modern mobile networks, GSM utilizes a cellular structure as illustrated in Figure 1.1. The basic idea of a cellular network is to partition the available frequency range, to assign only parts of that frequency spectrum to any base transceiver station, and to reduce the range of a base station in order to reuse the scarce frequencies as often as possible. One of the major goals of network planning is to reduce interference between different base stations.

Anyone who starts thinking about possible alternatives should be reminded that current mobile networks operate in frequency ranges where attenuation is substantial. In particular, for mobile stations with low power emission, only small distances (less than 5 km) to a base station are feasible.

Besides the advantage of reusing frequencies, a cellular network also: comes with the following disadvantages:

- An increasing number of base stations increase the cost of infrastructure and access lines.
- All cellular networks require that, as the mobile station moves, an active call is handed over from one cell to another, a process known as handover.
- The network has to be kept informed of the approximate location of the mobile station, even without a call in progress, to be able to deliver an incoming call to that mobile station.
- The second and third items require extensive communication between the mobile station and the network, as well as between the various network elements. That communication is referred to as signaling and goes far beyond the extent of signaling that fixed networks use. The extension of communications requires a cellular network to be of modular or hierarchical structure. A single central computer could not process the amount of information involved.

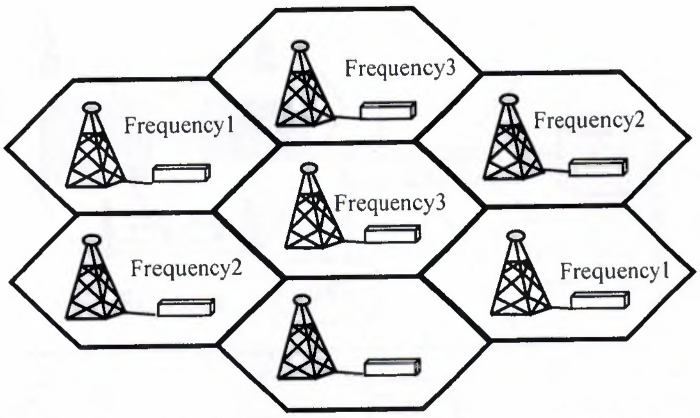


Figure 1.1 the Radio Coverage of an Area by Single Cells

1.2 An Overview on the GSM Subsystems

A GSM network comprises several elements: the mobile station (MS), the subscriber identity module (SIM), the base transceiver station (BTS), the base station controller (BSC), the transcending rate and adaptation unit (TRAU), the mobile services switching center (MSC), the home location register (HLR), the visitor location register (VLR), and the equipment identity register (EIR). Together, they form a public land mobile network (PLMN). Figure 1.2 provides an overview of the GSM subsystems.

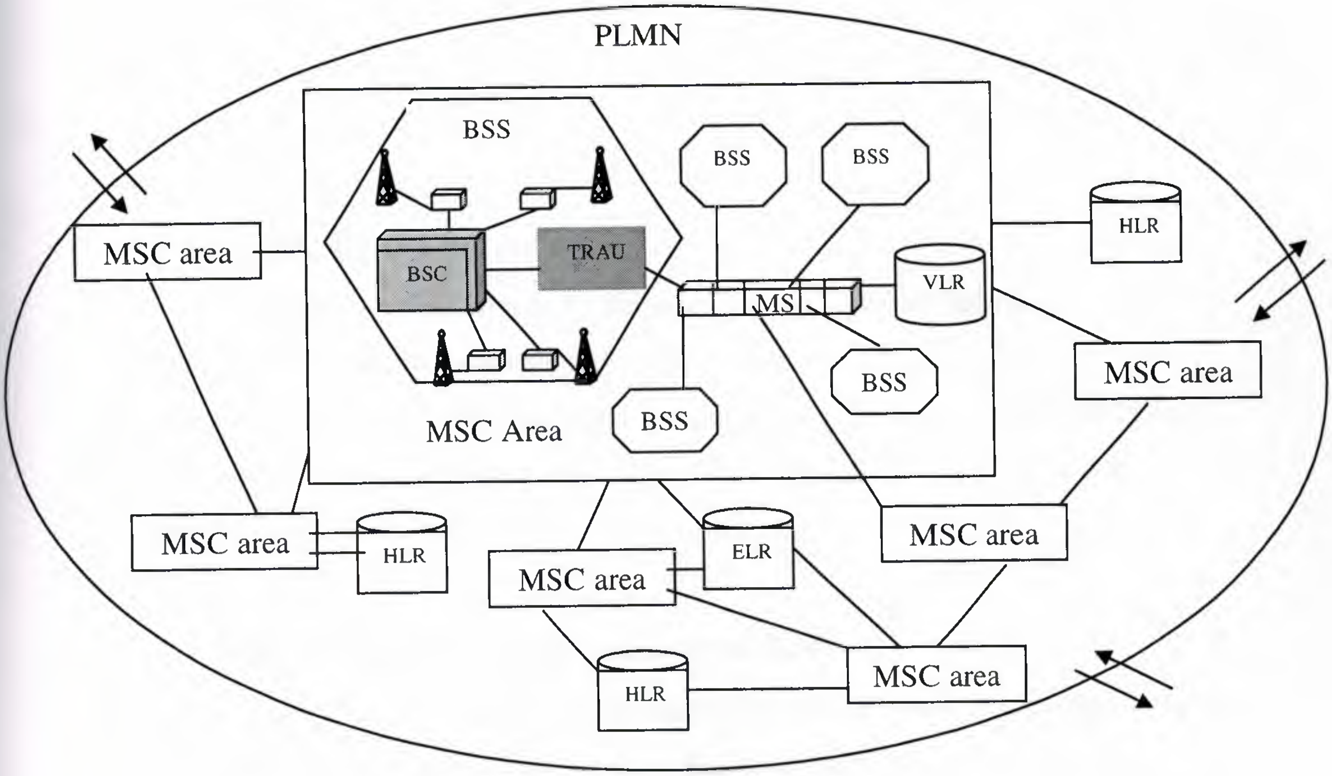


Figure 1.2 the Architecture of a PLMN

1.3 Mobile Station

GSM-PLMN contains as many MSs as possible, available in various styles and power classes. In particular, the handheld and portable stations need to be distinguished.

1.4 Subscriber Identity Module

GSM distinguishes between the identity of the subscriber and that of the mobile equipment. The SIM determines the directory number and the calls billed to a subscriber. The SIM is a database on the user side. Physically, it consists of a chip, which the user must insert into the GSM telephone before it can be used. To make its handling easier, the SIM has the format of a credit card or is inserted as a plug-in SIM. The SIM communicates directly with the VLR and indirectly with the HLR.

1.5 Base Transceiver Station

A large number of BTSs take care of the radio-related tasks and provide the connectivity between the network and the mobile station via the Air-interface.

1.6 Base Station Controller

The BTSs of an area (e.g., the size of a medium-size town) are connected to the BSC via an interface called the Abis-interface.

The BSC takes care of all the central functions and the control of the subsystem, referred to as the base station subsystem (BSS). The BSS comprises the BSC itself and the connected BTSs.

1.7 Transcoding Rate and Adaptation Unit

One of the most important aspects of a mobile network is the effectiveness with which it uses the available frequency resources. Effectiveness addresses how many calls can be made by using certain bandwidth, which in turn translates into the necessity to compress data, at least over the Air-interface. In a GSM system, data compression performed in both the MS and the TRAU. From the architecture perspective, the TRAU is part of the BSS. An appropriate graphical representation of the TRAU is a black box or, more symbolically, a clamp.

1.8 Mobile Services Switching Center

A large number of BSCs are connected to the MSC via the A-interface. The MSC is very similar to a regular digital telephone Exchange and is accessed by external networks exactly the same way. The major tasks of an MSC are the routing of incoming and outgoing calls and the assignment of user channels on the A—interface.

1.9 Home Location Register

The MSC is only one sub center of a GSM network. Another sub center is the HLR, a repository that stores the data of a large number of subscribers. An HLR can be regarded as a large database that administers the data of literally hundreds of thousands of subscribers. Every PLMN requires at least one HLR.

1.10 Visitor Location Register

The VLR was devised so that the HLR would not be overloaded with inquiries on data about its subscribers. Like the H LR, a VLR contains subscriber data, but only part of the data in the HLR and only while the particular subscriber roams in the area for which the VLR is responsible. When the subscriber moves out of the VLR area, the HLR requests removal of the data related to a subscriber from the VLR. The geographic area of the VLR consists of the total area covered by those BTSs that are related to the MSCs for which the VLR provides its services.

1.11 Equipment Identity Register

The theft of GSM mobile telephones seems attractive, since the identities of subscribers and their mobile equipment are separate. Stolen equipment can be reused simply by using any valid SIM.Barring of a subscriber by the operator does not bar the mobile equipment .To prevent that kind of misuse, every GSM terminal equipment contains a unique identifier, the international mobile equipment identity (IMEI). It lies within the realm of responsibilities of a network operation to equip the PLNM with an additional database, the EIR, in which stolen equipment is registered and so can he used to bar fraudulent calls and even, theoretically, to track down a thief (by analyzing the related SIM data).

1.12 GSM Base Station Measurements and its Methods

In recent years there has been a proliferation of base station towers designed to meet increased demands placed on mobile telephone networks by the growing number of mobile phone users .In parallel with the construction of these base station towers there has been an increase in community concern about possible health effects from the radio frequency (RF) radiation emissions from the towers. The Australian Government Committee on Electromagnetic Energy (EME) Public Health Issues (CEMEPHI), as part of the public information component of its RF EME program, considers it important that the general public be informed about the RF EME levels to which they may be exposed. Accordingly, the CEMEPHI requested the Australian Radiation Protection and Nuclear Safety Agency (ARPANSA) to carry out a survey of the RF EME levels in the vicinity of mobile telephone base stations. This report provides information on the levels of RF radiation from RF transmitter towers (base stations) to which members of the public may be exposed. Reviews on the potential health risks of RF radiation are available elsewhere (e.g., UNEP/WHO/IRPA, 1993; Barnett, 1994; McKinley etal, 1996; ICNIRP, 1998; Repacholi, 1998; Byrus et al., 1999).

A survey on RF EME in and around five Vancouver schools by Thansandote et al. (1999),

Both at indoor and outdoor sites, yielded power density measurements well within Canada's safety code limits (Safety Code 6, 1990). Signal sources investigated in the Thansandote et al survey included base station frequency bands for analog cellular phones and personal communication services (PCS – the new generation of digital cellular phone), as well as AM radio, FM radio and TV broadcasts. A US study by Petersen and Testagrossa (1992) characterized RF EME fields in the vicinity of several frequencies modulated (FM) cellular radio antennae towers, at heights varying from 46 to 82 meters. They reported maximum power densities considered representative of public exposure levels to be less than 0.0001 W/m 2 per transmitter. Hence, in a worst-case scenario of 96 transmitters operating at an

Effective radiated power (ERP) of 100 watts per transmitter; the aggregate maximum power density was estimated by Petersen and Testagrossa to be below 0.01 W/m^2. In Poland, where the maximum permissible power density value is 0.1 W/m^2 at relevant base station.

Frequencies, measurements of electromagnetic fields (EMF) in the surrounds of 20 GSM base stations showed that 'admissible EMF intensities at the level of people's presence, in existing buildings, in surroundings of base stations and inside buildings with antennas, were not exceeded' (Aniolczyk, 1999, p.57).

The purpose of the work reported here is to provide data on RF EME levels at independently nominated sites, over the range of the digital Global System for Mobile communication (GSM) mobile telephone base stations frequency band (935 – 960 MHz), and to make comparisons with the limit for non-occupational exposure specified in the relevant Australian exposure standard. The Radio communications (Electromagnetic Radiation Human Exposure) Standard 1999 adopted by the Australian Communications Authority (ACA) requires mobile phones and mobile phone base stations to comply with the exposure limits in the interim Australian and New Zealand Standard 2772.1(Int): 1998 which has now been withdrawn by

Standards Australia. The ACA standard is subsequently abbreviated as ACAS in this publication. The non-occupational exposure limit specified in the ACAS, expressed in terms of power flux density, is 2 W/m^2 (equivalent to 200 $\mu W/cm^2$) for frequencies between 10 MHz and 300 GHz, averaged over a 6 minute period. It should be noted that the exposure limits in the ACAS were 'developed on the basis of there being a threshold of 4 W/kg whole body specific absorption rate (SAR) before any adverse health consequences are likely to appear' (ibid, p.13). However, because the SAR (units W/kg) is difficult and often impractical to measure, the ACAS provides derived levels of electric (E) and magnetic (H) field strengths, as well as the equivalent plane wave power flux densities (S), which are more readily measured.

Although the primary focus of the ARPANSA study was to measure the RF EME emission levels from GSM base stations, fixed site environmental measurements from other RF EME sources were also recorded, including the analog mobile phone system (AMPS), VHF TV UHF TV, AM radio, FM radio and Paging.

1.12.1 Method of Measurement Locations

Measurements were performed at fourteen different locations throughout Australia. Two localities were chosen from each state, and the Northern Territory. In most instances the sites were chosen by local governments, who were asked to nominate two mobile telephone base stations sites in major population centers that were of concern to local communities. Security of monitoring equipment for the 24-hour data-logging component was taken into account in the final selection of the measurement sites. Following the nature and type of the measurements required.

- Fixed Site Environmental Measurements

Broadcast communication sources such as television, and both AM radio and FM radio, are usually transmitted at high powers from a single base facility. Such sources have very extensive areas of effective reception frequently extending to many hundreds of kilometers from a single station transmitter. Furthermore, for such sources and considering their necessary broadcast design requirements, we do not expect to encounter significant or strong variations in signal strength in relatively open areas surrounding a mobile telephone base station. Given the nature and emphasis of our study we therefore adopted a protocol of making a single set of static environmental measurements for all broadcast sources other than mobile telephone base stations.

Buildings or other likely objects may significantly attenuate or scatter the RF signal. Hence, where possible, measurements were made in locations that maintained direct line-of-sight with known RF sources, at a height of 1.7 meters above ground, in open areas in the near vicinity of the GSM base station of interest. Measurement antenna were oriented to obtain a maximum signal strength for the particular frequency band

Being measured. The environmental RF EME signals were measured at a location within 500 meters of the base station.

Measurement of such fixed site environmental RF EME levels involved investigating a number of different RF EME sources. These included GSM, AMPS, VHF TV, UHF TV, AM radio, FM radio and paging. All signals with power densities greater than 1% of the observed maximum for each frequency band were recorded individually. Other signals, such as emergency services (police, ambulance, etc.) and taxis, were rarely detected and are not included in this summary report. To measure the environmental RF EME levels the average RF EME levels over a six minute scanning period during the day was determined. The time taken to record all the relevant sources of environmental RF EME at each site were approximately one hour. A spectrum analyzer was used and some transient signal sources,

Such as paging services, may have gone undetected if by chance the relevant frequency band was not swept by the spectrum analyzer when the signal was transmitted.

- GSM Base Station Activity Measurements

The primary aim of this study was to determine the RF EME level resulting from all signal frequencies produced by the particular GSM base stations under survey. Mobile telephone communication signals are both transient and partly random in their occurrence and distribution. In this context, we were interested in determining the RF EME levels at many locations and more particularly, we wanted to estimate both maximum and minimum levels and also the long term average value for each location and to map such levels in the area surrounding the base station. Because telephone communications are based on human activity,

A diurnal signal pattern is generally observed. Site-specific GSM mobile telephone exposure levels were therefore monitored over a 24-hour period. Relevant spectrum analyzer data were recorded automatically under PC control and subsequently analyzed to determine both the temporal and daily average activity. Measurements were performed within a single sector, at a fixed location close to the base station, by continuously scanning the frequency bands and logging the signal level for the GSM mobile phone systems. The recorded data were used to determine the temporal activity for the GSM systems over the 24-hour period.

The activity level of the data samples was determined by counting the number of simultaneous active time slots for a single carrier base station. For the majority of GSM base stations there is a possible minimum of eight and a possible maximum of thirty-two time slots for any given sector.

Hence, eight time slots will amount to 25% of the total activity possible from the transmitting antenna of a single carrier GSM base station.

The digital GSM base stations produce carrier frequencies between 935 to 960 MHz (analog AMPS system operates at 870 to 890 MHz). The GSM system transmits data in bursts of 0.6μsec with a repetition rate of 217 Hz. The temporal RF EME levels of the transmitting antennae at GSM base stations were analyzed to identify control frequencies or additional carrier frequencies. For GSM the frequency range investigated was divided up into three sub-bands, with the sampling order of each sub-band and frequency randomized to avoid bias. The system was optimized to gather as much data as possible by sampling more often when fewer frequencies were detected. Post logging data analysis was performed to determine the average activity over a six-minute scanning period, yielding an activity value for every six minutes of the day. The analysis software included only the signals identified as belonging to the base station in question. Where more than one carrier (Telstra, Optus or Vodafone) shared the same tower, the combined activity from all carriers was determined. A diurnal correction factor was derived from analysis of the 24-hour activity measurements for use in mobile measurements.

- Mobile GSM Base Station Area Measurements

A fixed antenna was roof mounted on a car and automated mobile measurements were made whilst driving around the streets near the GSM base station under survey. Both signal data and position information [using Global Positioning System (GPS)] were recorded. For technical reasons, we were not able to make simultaneous measurements of all frequencies at each particular mobile measurement sample location. However, for each base station sector there is always a single "control frequency" present and this frequency is produced at a constant transmitter power. The control frequency is broadcast from the same antennae as additional transient carrier frequencies. In addition, the control frequency will have similar propagation characteristics to those of any additional frequencies. Hence, to determine the RF EME area levels, only the control frequency (surrogate for all frequencies) was measured. Application of the diurnal correction factor obtained by previous activity data analysis yielded an estimate of the average RF EME over 24 hours at each measured point in the mapping area.

Maps of each survey area displaying the distribution of the 24-hour average RF EME levels at each measured point are presented in the individual reports for each survey site.

2. GSM TRANSMISSION PROCESS

2.1 Overview

In this chapter I'm going to talk about the incoming and the outgoing calls and I'll explain some main parameters and operations that will be important to the call before the call being transmitted or delivered to the other side, such as signaling and sampling and using quantization to get the suitable and correct form of the sent samples, after that I'll talk about speech coding so we can make the data suitable to the channel and the bandwidth so it can be sent easily, after that modulation will be talked about.

2.2 Introduction

The basic components associated with a GSM transmitter and receivers are represented in Figure 2.1. These blocks are implemented in order to transmit and receive data with the least possible number of errors.

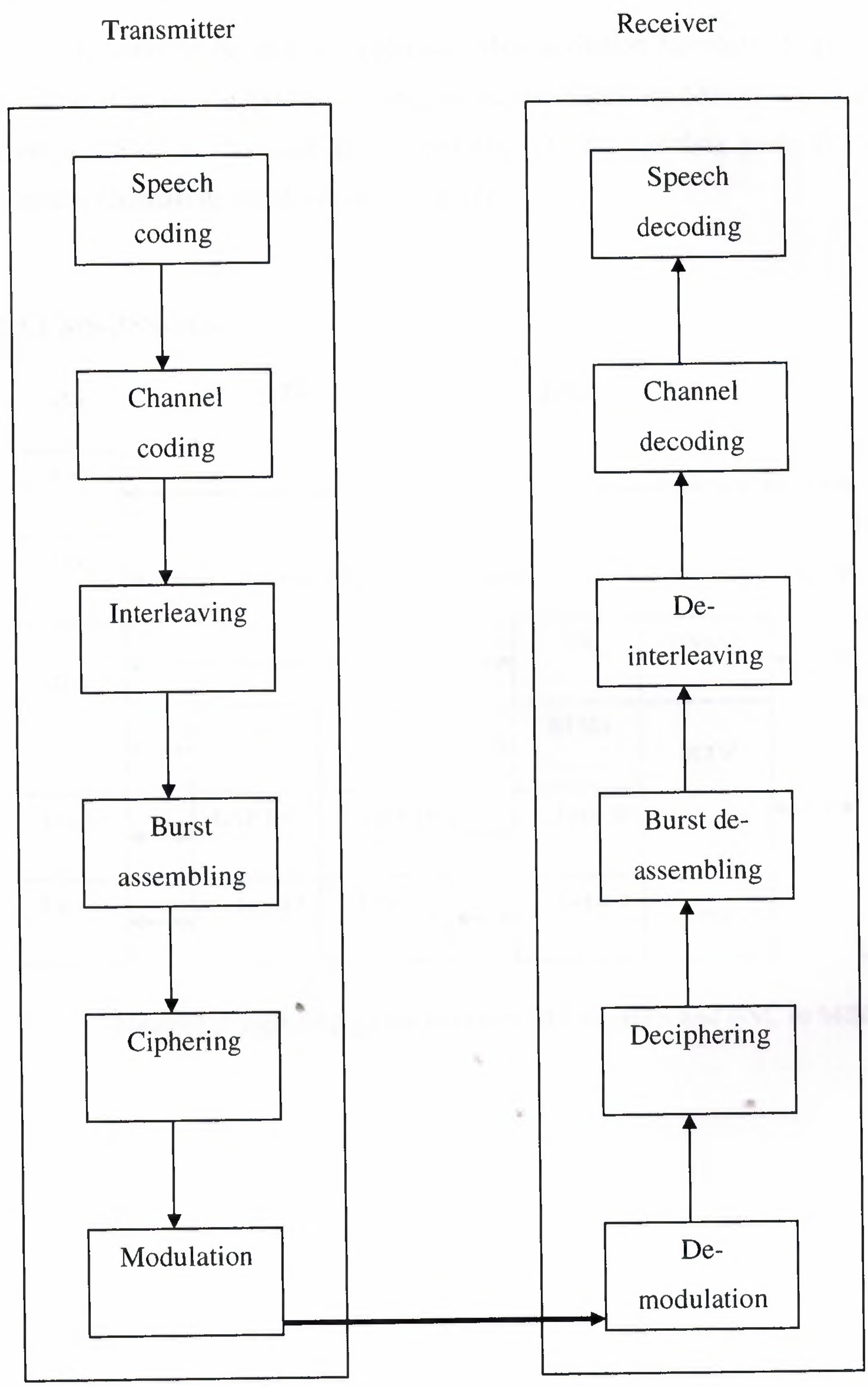


Figure 2.1: GSM Transmission Process.

2.3 Signaling

In order to be able to implement Mobile Station Location (MSL) in a GSM network, it is very important to understand the Signaling protocols and procedures used in GSM. In this section, an overview of the Signaling protocols and some important Signaling sequences will be given.

2.3.1 MS-BSS-MSC

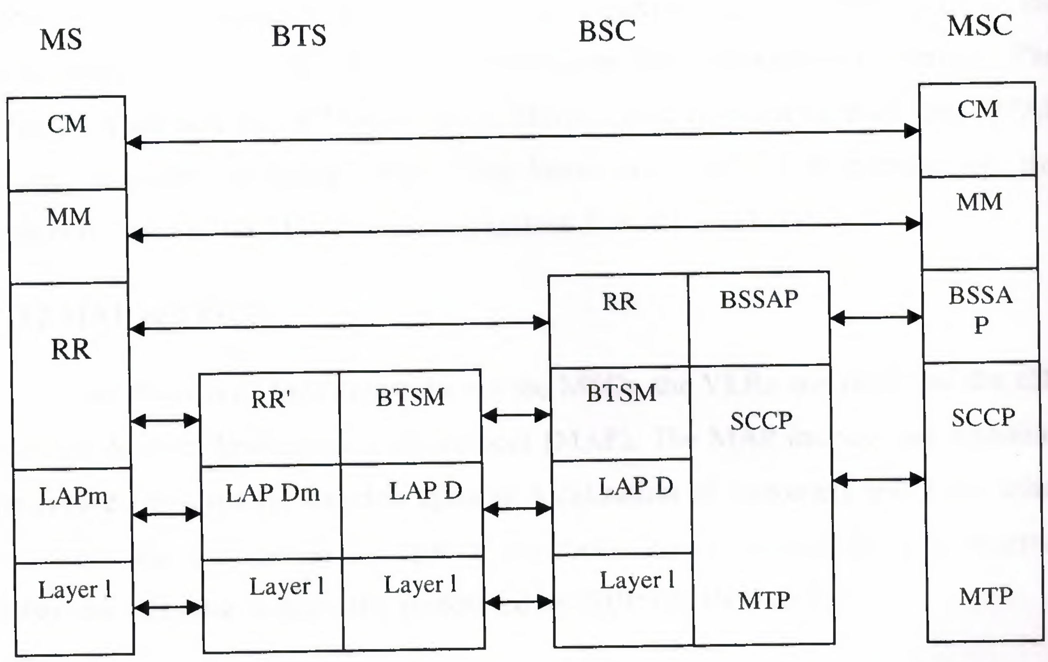


Figure2.2 Signaling protocols from MS via BTS and BSC to MSC

Figure 2.2 shows an overview of the Signaling protocols in the GSM network between the entities MS and MSC. Above the lower layers in the BSS, is the Radio Resources Protocol (RR). This protocol deals with the allocation, deallocation and parameters of the radio-channel and is crucial in the setup of all communication with the MS. Above this layer is the Mobility Management (MM) and Circuit Mode Connection Call Protocol (CM or CC). The MM deals with administration of localization and handover. The CM administrates the setup and termination of calls. There also exist protocols between the different entities in the network intended for network internal messages. These are BTS Management protocol (BTSM) across the Abis interface and the BSSAP (BSS Application Part) across the A interface. The BSSAP is divided into BSSMAP (BSS Management Application Part) and DTAP (Direct Transfer Application Part). The lower layers of the A interface are the transport layers of the ITU-T Signaling system 7, SCCP and MTP.

2.3.2 MAP and ISUP

All functional Signaling between the MSCs, the VLRs, the HLR and the EIR uses the Mobile Application Part protocol (MAP). The MAP includes all Signaling procedures required for location updates, localization of customers and many other functions that are special for mobile networks. To be compatible with external networks, call setup is normally performed by ISUP (ISDN User Part).

2.3.3 Call setup

To get an idea of the complexity of the signaling procedures and show some of the signals that later will be used; the complete signal-sequence for a mobile-terminated call will be shown here. Figure 2.3 shows the signaling sequence between the ISDN network and the GSM network.

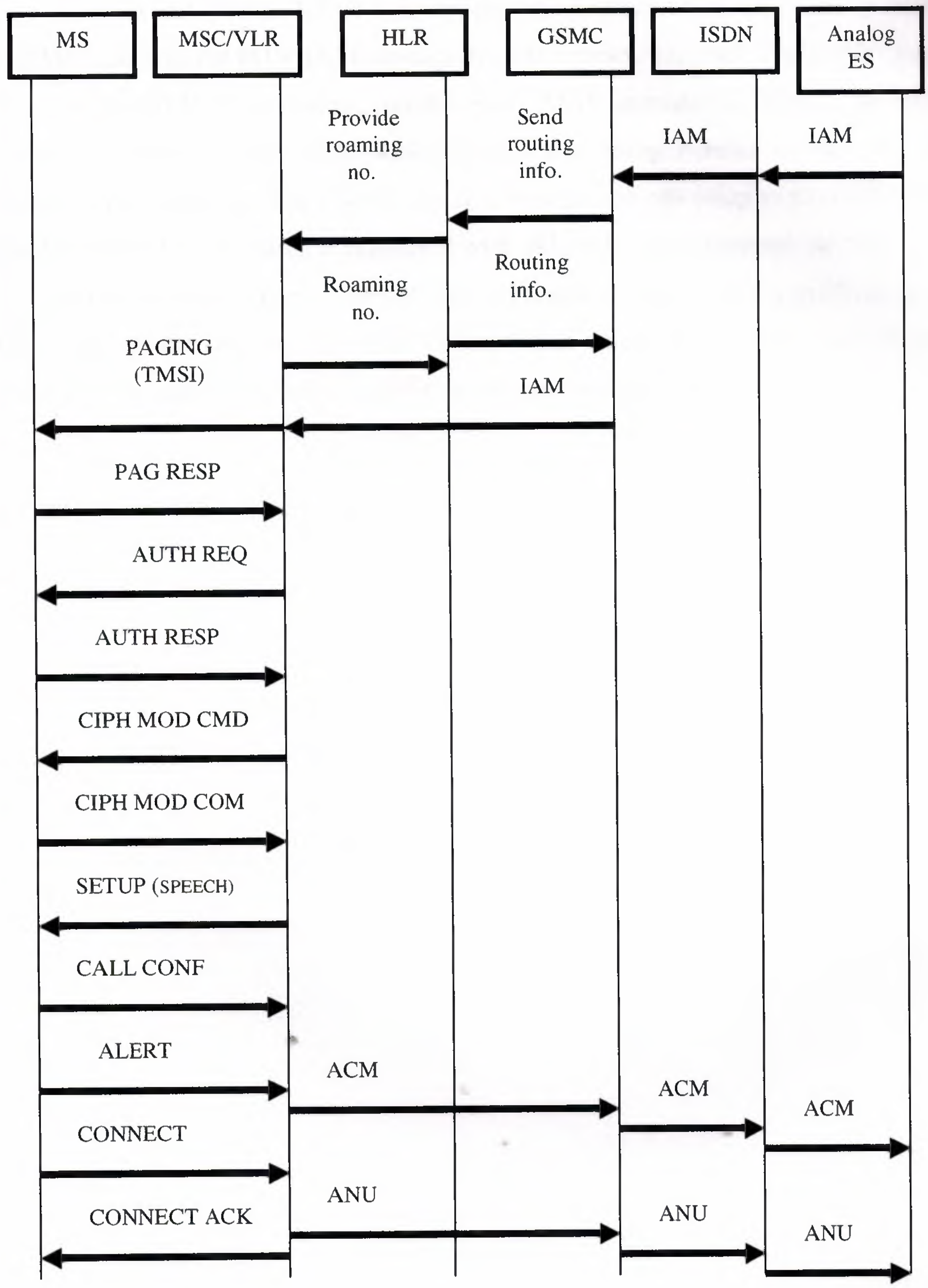


Figure2.3 Signaling between ISDN and GSM at a mobile terminated call setup.

As we can see on figure 2.3, the procedure starts when the Gateway MSC (GSMC) receives the ISUP IAM message from the remote network. The GMSC must then ask the HLR for a roaming number using MAP procedures. Further, the HLR sends this request to the VLR, which assigns a roaming number to the IMSI in question, and returns it. The GMSC can now forward the call setup request (IAM) to the MSC the MS in question is registered with. When the setup between the MSC and the MS is finished, the user is alerted (the cell phone is ringing) and a notification of this is sent to the caller via the ISUP ACM. When the receiver accepts the call, the ISUP ANU is sent to the caller, and the connection is established.

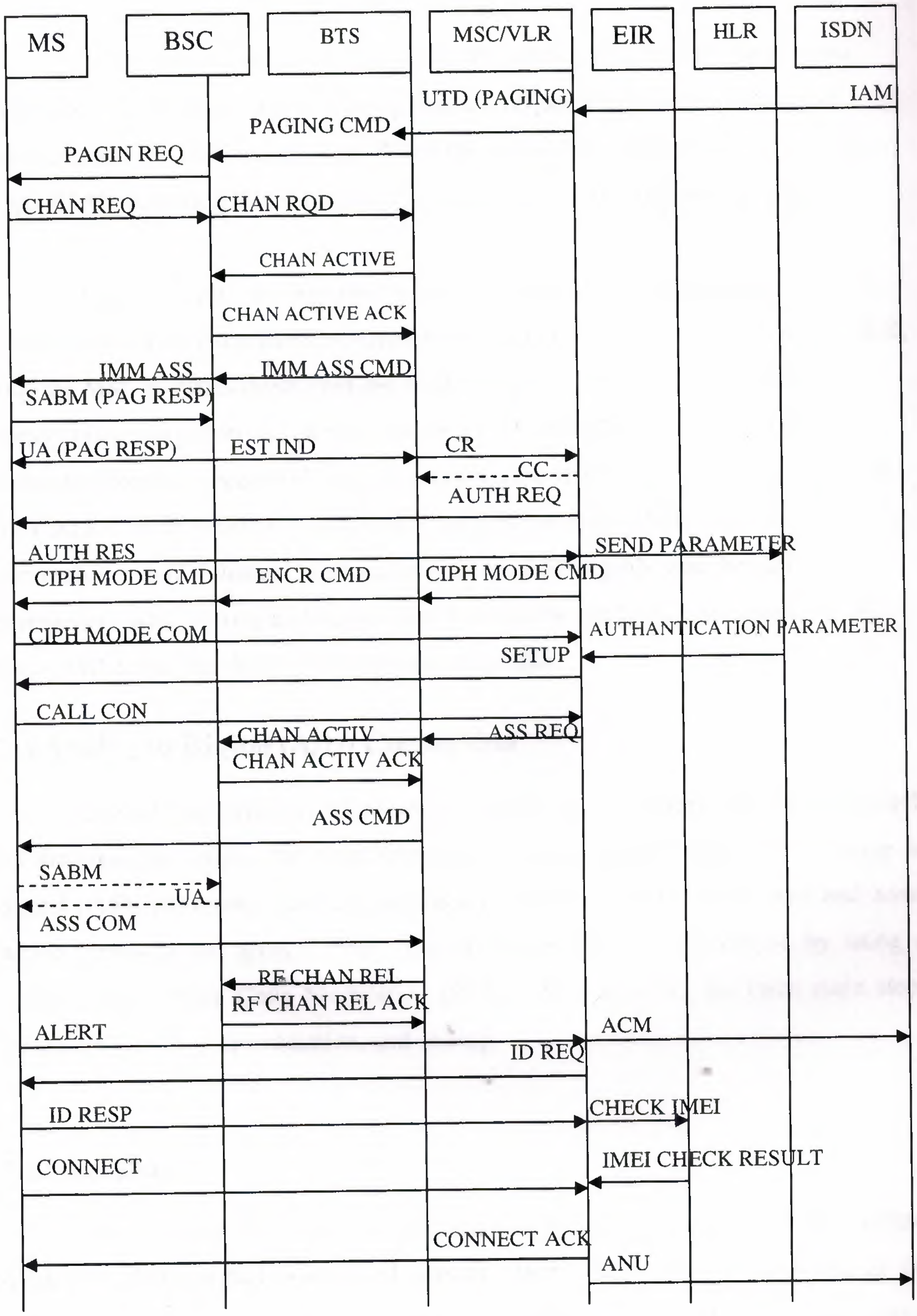


Figure 2.3 shows in detail what happens between the MSC and the MS.

The paging request is sent out on all the base stations in the location area. When the MS discovers that it is being paged it requests a channel on the radio interface, and the BSC assigns one. When the channel is active, the MS sends the PAG RESP indicating that it has been paged, and is ready to answer the paging.

When the MSC receives this, it commences with authentication of the MS. The authentication parameters received from the MS must be checked with the HLR, thus the MSC requests these from the HLR with the ``Send Parameters" request. Meanwhile, encryption can be initiated with the CIPH MODE signals. If the authentication was successful, the call setup is sent to the MS, which responds with the CALL CONF, where it's indicated if the MS can respond this call type. If this is successful, a traffic channel is allocated with the ASS signals, and the call commences with alerting and connection. Optionally, the MSC can request the MS for its IMEI, and check if it is blacklisted in the EIR.

2.4 Analog to Digital (A/D) Conversion

One of the primary functions of a MS is to convert the analog speech information into digital form for transmission using digital signal. The analog to digital (A/D) conversion process outputs a collection of bits: binary ones and zeros which represent the speech input. The A/D conversion is performed by using a process called Pulse Code Modulation (PCM). PCM involves the three main steps which are sampling, quantization, and coding.

2.4.1 Sampling

The accuracy of describing the analog signal in digital terms depends on how often the analog signal is sampled, among other things. This is expressed as the sampling frequency. The sampling theory states that: "To reproduce an analog signal without distortion, the signal must be sampled with at least twice the frequency of the highest frequency component in the analog signal".

Normal speech mainly contains frequency components lower than 3400 Hz. applying the sampling theory to analog speech signals; the sampling frequency should be at least 2 * 3.4 kHz = 6.8 kHz. Telecommunication systems use a sampling frequency of 8 kHz, which is acceptable based on the sampling theory.

2.4.2 Quantization

The next step is to give each sample a value. For this reason, the amplitude of the signal at the time of sampling is measured and approximated to one of a finite set of values. A slight error is introduced in this process when the signal is quantized or approximated. The degree of accuracy depends on the number of quantization levels used. In GSM 8192 levels are used.

2.4.3 Coding

Coding involves converting the quantized values into binary. Every value is represented by a binary code of 13 bits ($2^{13} = 8192$).

2.5 Speech Coding

In mobile communication systems, the design and subjective tests of speech coders have been extremely difficult. Without low data rate speech coding, digital modulations schemes offer little in the way of spectral efficiency for voice traffic. To make speech coding practical, implementations must consume little power and provides tolerable, if not excellent, speech quality.

The goal of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity. This has to be accomplished while maintaining certain required levels of complexity of implementation and communication delay. In general, there is a positive correlation between coder bit-rate efficiency and the algorithmic complexity required to achieve it. The more complex an algorithm is, the more its processing delay and cost of implementation. A balance needs to be struck between these conflicting factors. And it is the aim of all speech processing developments to shift the point at which this balance is made toward even low bit rates .

The GSM speech coder takes advantage of the fact that in a normal conversation each person speaks on average for less than 40% of the time. By incorporating a voice activity detector (VAD) in the speech coder, GSM systems operate in a discontinuous transmission mode (DTX) which provides a longer subscriber battery life and reduces instantaneous radio interference since the GSM transmitter is not active during silent periods. A comfort noise subsystem (CNS) at the receiving end introduces a background acoustic noise to compensate for the annoying switched muting which occurs due to DTX.

2.5.1 The Dimensions of Performance in Speech Compression

Speech coders attempt to minimize the bit rate for transmission or storage of the signal while maintaining required levels of speech quality, communication delay, and complexity of implementation (power consumption). The following criteria represent the performance of speech compression.

2.5.1.1 Speech Quality

Speech quality is usually evaluated on a five-point scale, known as the mean-opinion score (MOS) scale, in speech quality testing---an average over a large number of speech data, speakers, and listeners. The five points of quality are: bad, poor, fair, good, and excellent. Quality scores of 3.5 or higher generally imply high levels of intelligibility, speaker recognition and naturalness.

2.5.1.2 Bit Rate

The coding efficiency is expressed in bits per second (bps).

2.5.1.3 Communication Delay

Speech coders often process speech in blocks and such processing introduces communication delay. Depending on the application, the permissible total delay could be as low as 1 m, as in network telephony, or as high as 500 ms, as in video telephony. Communication delay is irrelevant for one-way communication, such as in voice mail.

2.5.1.4 Complexity

The complexity of a coding algorithm is the processing effort required to implement the algorithm, and it is typically measured in terms of arithmetic capability and memory requirement, or equivalently in terms of cost. A large complexity can result in high power consumption in the hardware.

2.5.2 Speech Coders Categories

Speech coders differ widely in their approaches to achieve signal compression, based on the means by which they achieve compression, speech coders are broadly classified into two categories as shown in Figure 3.2: waveform coders and vocoders.

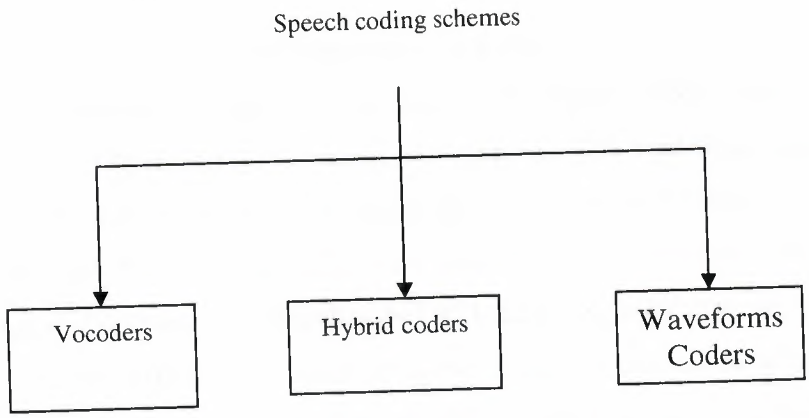


Figure 2.5: Speech Coders Categories.

2.5.2.1 Waveform Coders

Waveform coders essentially strive to reproduce the time waveform of the speech signal as closely as possible. They are designed to be source independent and hence code equally well a variety of signals. They have the advantage of being robust for a wide range of speech characteristics and for noisy environment .The waveform coders are able to produce high-quality speech at high enough bit rates .

2.5.2.2 Vocoders

Vocoders achieve very high economy in transmission bit rate, and in general it's more complex. They are based on using a priori knowledge about the signal to be coded, and for this reason, they are, in general, signal specific .

Vocoders produce intelligible speech at much lower bit rates and the applications of vocoders so far, have been limited to low-bit-rate digital communication channels.

2.5.3 Speech Coding in GSM

GSM is a digital communications standard, but the voice is analog, and therefore it must be converted to a digital bit stream. GSM uses Pulse Coded Modulation (64kbps) to digitize voice, and then uses the Full-Rate speech codec to remove the redundancy in the signal and achieve a bit rate of 13 kbps.

In order to send our voice across a radio network, we have to turn our voice into a digital signal. GSM uses a method called RPE-LPC (Regular Pulse Excited - Linear Predictive Coder with a Long Term Predictor Loop) to turn our analog voice into a compressed digital equivalent with lower bit rate. Once we have a digital signal we have to add some sort of redundancy so that we can recover from errors when we tram our digital voice over the radio channel.

The LPC encoder fits a given speech signal against a set of vocal characteristics. The best-fit parameters are transmitted and used by the decoder to generate synthetic speech that is similar to the original. Information from previous samples is used to predict the current sample. The coefficients of the linear combination of the previous samples, plus an encoded form of the residual, the difference between the predicted and actual sample, represent the signal. Speech is divided into 20 millisecond samples, each of which is encoded as 260 bits, giving a total bit rate of 13 kbps. See Figure 3.3 for a representation of RPE-LPC .

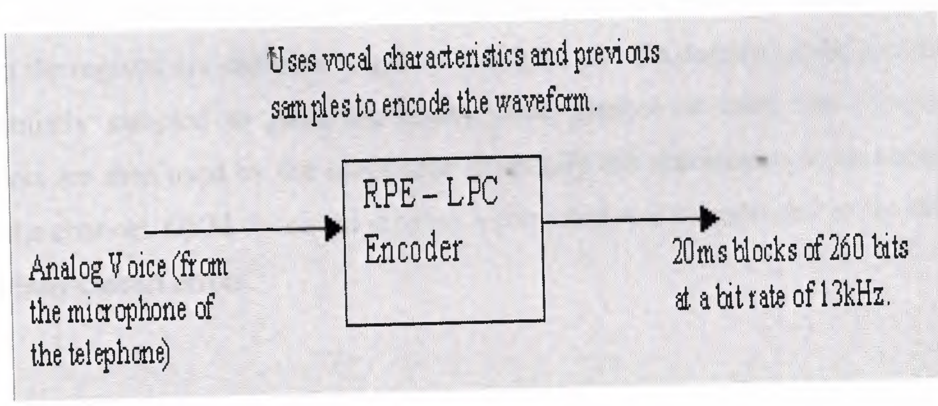


Figure 2.6: RPE-LPC Encoder.

2.6 Channel Coding

Channel coding refers to the class of signal information's designed to improve communications performance by enabling the transmitted signal to better withstand the effects of various channel impairment, such as noise, interference, and fading. It introduces redundancy into the data flow in order to allow the detection or even the correction of bit errors introduced during the transmission. In order to create a reliable connection we need to protect the data as much as possible at that bit error rate (BER) we cannot rely on error detection and retransmission. So in order to get over this you have to use channel coding .

2.6.1 Convolutional Encoder

A convolutional code which is used in GSM is described by three integers, n, k, and K, where the ratio k/n has the same code rate significance (information by coded bit) that it has for block codes; however, n does not define a block or codeword length as it does for block codes. The integer K is a parameter known as the constraint length; it represents the number of k-tuple stages in the encoding shift register, the n-tuple emitted by the convolutional encoding procedure is not only a function of an input k-tuple, but is also a function of the previous K-1 input k-tuples

At each unit of time, k bits are shifted into the first k stages of the register; all bits in the register are shifted k stages to the right, and the outputs of the n adders are sequentially sampled to yield the binary code symbol or code bits. These code symbols are then used by the modulator to specify the waveforms to be transmitted over the channel. GSM choose to employ a convolutional encoder due to its ability to efficiently correct errors.

2.6.2 Channel Coding Process in GSM

The speech coding produces a 260 bit block for every 20 ms speech sample. From subjective testing, it was found that some bits of this block were more important for perceived speech quality than others. The bits are thus divided into three classes (See Figure 2.7):

- Class Ia 50 bits - most sensitive to bit errors
- Class Ib 132 bits - moderately sensitive to bit errors
- Class II 78 bits - least sensitive to bit errors

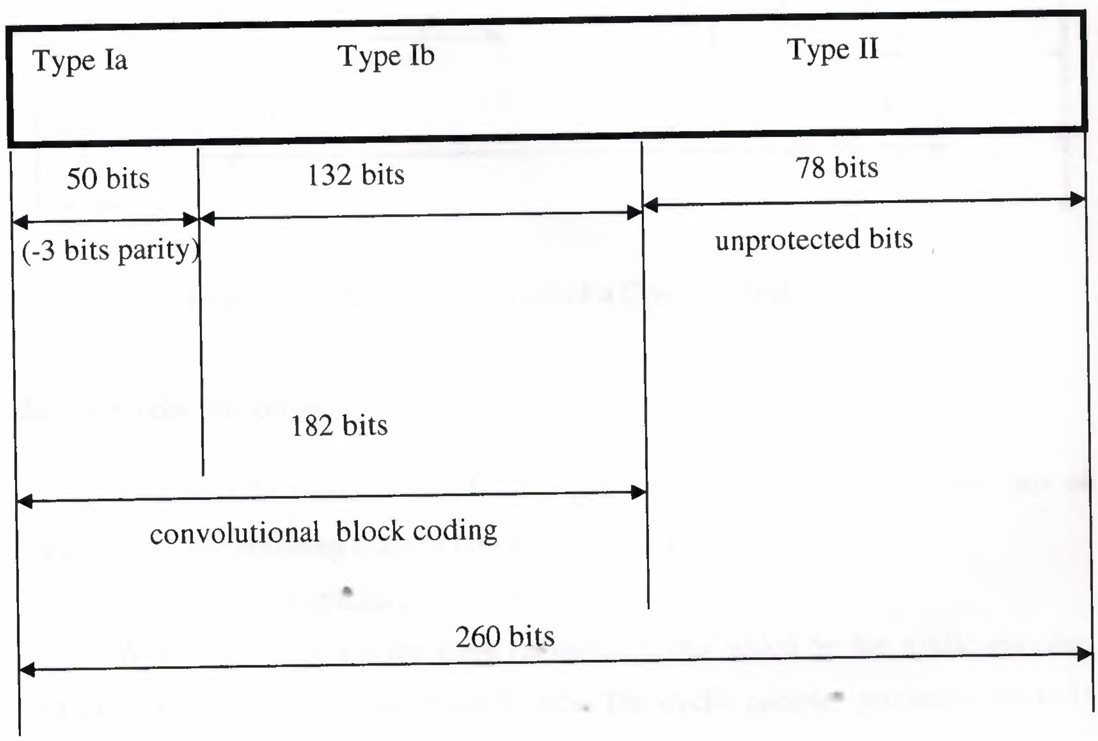


Figure 2.7: Channel Coding In GSM.

As a result of some bits being more important than others, GSM adds redundancy bits to each of the three classes differently. The class Ia bits are encoded in a cyclic encoder. The class Ib bits (together with the encoded class Ia bits) are encoded using convolutional encoding. Finally, the class II bits are merely added to the result of the convolutional encoder. The Figure below is the operation of each encoder as related to each class of bits.

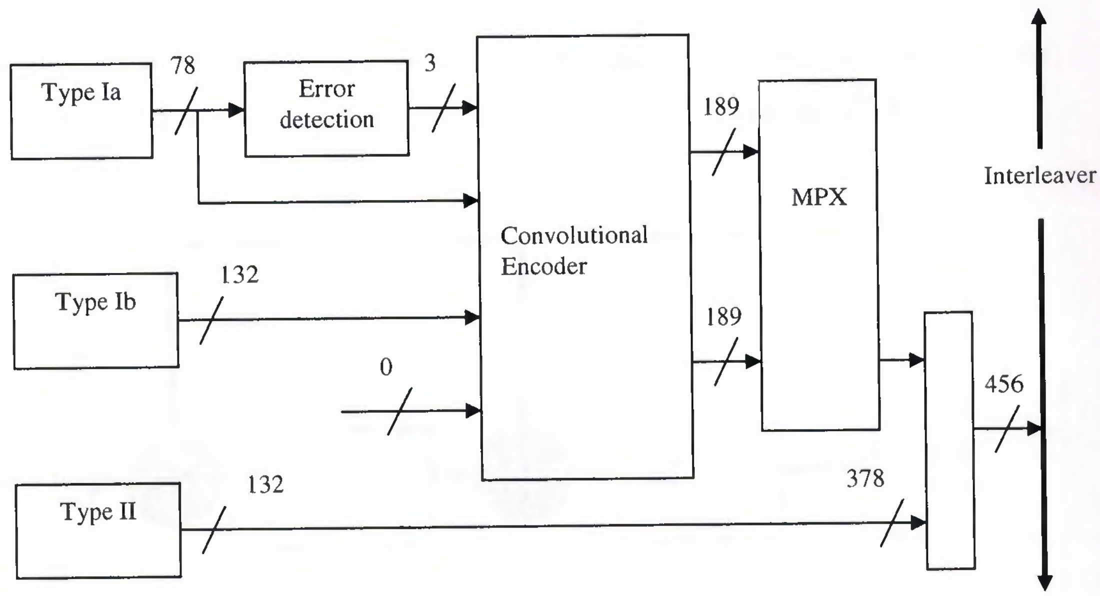


Figure 2.8: A Block Diagram of a Convolutional Encoder.

2.6.2.1 Cyclic Encoding

The class Ia bits are encoded using a cyclic encoder to add three bits of redundancy. The resulting class Ia bits are of the form:

$$(b_0, b_1, b_2, m_0, m_1, \ldots\ldots\ldots m_{49})$$

Where b_0, b_1, b_2 are the three redundancy bits added by the cyclic encoder, and $m_0 \ldots m_{49}$ are the original Class Ia bits. The cyclic encoder produces 50+3=53 bits.

Cyclic codes are linear codes (the sum of any two codes is also a codeword). In addition to being linear, a cyclic shift, or rotate, of a codeword produces another codeword, since the code used in GSM is a (53, 50) code, the generator polynomial used in the encoding is of degree 53-50 = 3. The following block diagram can produce the codeword. In Figure 2.9 once the data has been completely shifted through the system, the contents of Reg0 through Reg2 will contain the three additional bits.

GSM chooses to use cyclic encoding due to the ability to quickly determine if errors are present. The three redundancy bits produced by the cyclic encoder enable the receiver to quickly determine if an error was produced. If an error was produced the current 53 bit frame is discarded and replaced by the last known "good" frame.

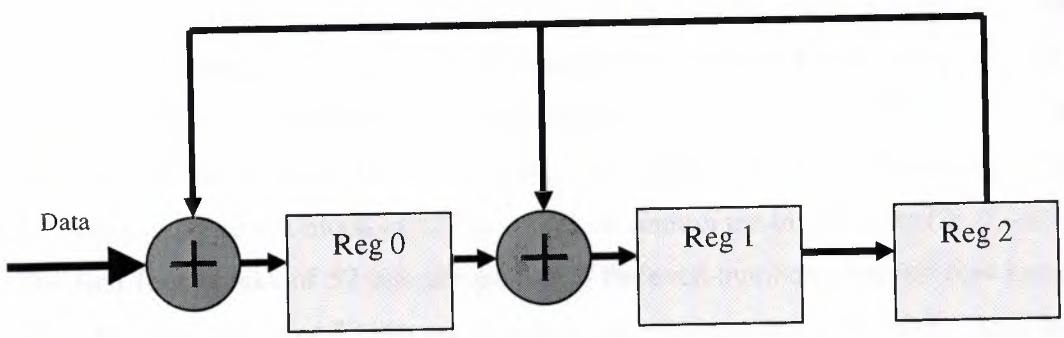


Figure 2.9: Cyclic Encoding.

2.7 Interleaving

Interleaving is meant to de-correlate the relative positions of the bits respectively in the code words and in the modulated radio bursts. The aim of the interleaving algorithm is to avoid the risk of loosing consecutive data bits. GSM blocks of full rate speech are interleaved on 8 bursts: the 456 bits of one block are split into 8 bursts in sub-blocks of 57 bits each. A sub-block is defined as either the odd- or the even-numbered bits of the coded data within one burst. Each sub-blocks of 57 bits is carried by a different burst and in a different TDMA frame. So, a burst contains the contribution of two successive speech blocks A and B. In order to destroy the proximity relations between successive bits, bits of block A use the even positions inside the burst and bits of block B, the odd positions. De-interleaving consists in performing the reverse operation. The major drawback of interleaving is the corresponding delay: transmission time from the first burst to the last one in a block is equal to 8 TDMA frames (i.e. about 37 ms).

Rearrange a group of bits in a particular way. It is used in combination with FEC codes in order to improve the performance of the error correction mechanisms. The interleaving decreases the possibility of losing whole bursts during the transmission, by dispersing the errors. Being the errors less concentrated, it is then easier to correct them.

A burst in GSM transmits two blocks of 57 data bits each. Therefore the 456 bits corresponding to the output of the channel coder fit into four bursts (4*114 = 456). The 456 bits are divided into eight blocks of 57 bits. The first block of 57 bits contains the bit numbers (0, 8, 16...448), the second one the bit numbers (1, 9, 17...449), etc. The last block of 57 bits will then contain the bit numbers (7, 15...455). The first four blocks of 57 bits are placed in the even-numbered bits of four bursts. The other four blocks of 57 bits are placed in the odd-numbered bits of the same four bursts. Therefore the interleaving depth of the GSM interleaving for control channels is four and a new data block starts every four bursts. The interleaver for control channels is called a block rectangular interleaver.

2.7.1 Interleaving for the GSM speech channels

The block of 456 bits, obtained after the channel coding, is then divided in eight blocks of 57 bits in the same way as it is explained in the previous paragraph. But these eight blocks of 57 bits are distributed differently. The first four blocks of 57 bits are placed in the even-numbered bits of four consecutive bursts. The other four blocks of 57 bits are placed in the odd-numbered bits of the next four bursts. The interleaving depth of the GSM interleaving for speech channels is then eight. A new data block also starts every four bursts. The interleaver for speech channels is called a block diagonal interleaver.

2.7.2 Interleaving for the GSM data TCH channels

A particular interleaving scheme, with an interleaving depth equal to 22, is applied to the block of 456 bits obtained after the channel coding. The block is divided into 16 blocks of 24 bits each, 2 blocks of 18 bits each, 2 blocks of 12 bits each and 2 blocks of 6 bits each. It is spread over 22 bursts in the following way :

1-the first and the twenty-second bursts carry one block of 6 bits each
2-the second and the twenty-first bursts carry one block of 12 bits each
3-the third and the twentieth bursts carry one block of 18 bits each
4-from the fourth to the nineteenth burst, a block of 24 bits is placed in each burst
A burst will then carry information from five or six consecutive data blocks. The data blocks are said to be interleaved diagonally. A new data block starts every four bursts.

2.8 Burst Format

GSM uses both FDMA and TDMA for multiple accesses. GSM900 uses the 890 MHz to 960 MHz frequency band for communications (Uplink 890-915 MHz, Downlink 935-960 MHz). GSM900 has two frequency bands 45 MHz. The 890 MHz to 915 MHz frequency band is used for uplink from a MS to a BS and 935 MHz to 960 MHz frequency band is used for the downlink from a BS to a MS. Each of the two bands is subdivided into 124 single carrier channels of 200 KHz each. A guard band of 200 KHz is left on both sides of uplink and the downlink bands (See Figure 2.10).

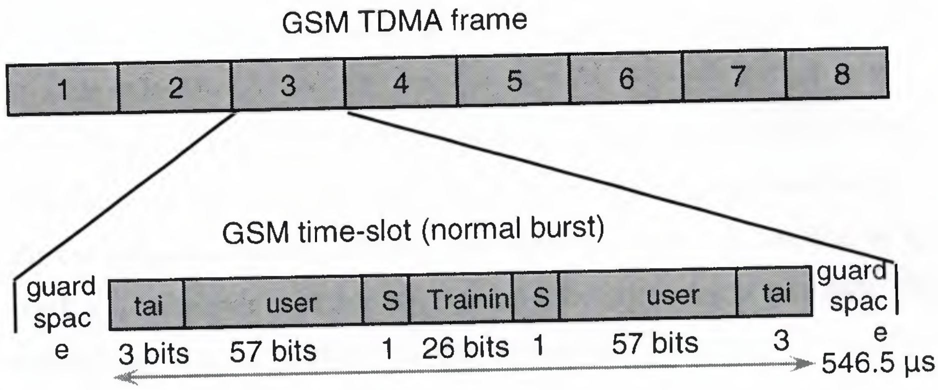


Figure 2.10: Burst Formats in GSM.

In Figure 2.11, each of the 200 KHz channels is divided into eight time slots and carries eight TDMA channels. The eight time slots form a TDMA frame. A MS uses the same time slot number in the uplink and the downlink. Each time slot of a TDMA frame lasts for 576.9 microseconds and contains data, which is also called a burst. The transmission rate is 270.83 kbps per carrier frequency.

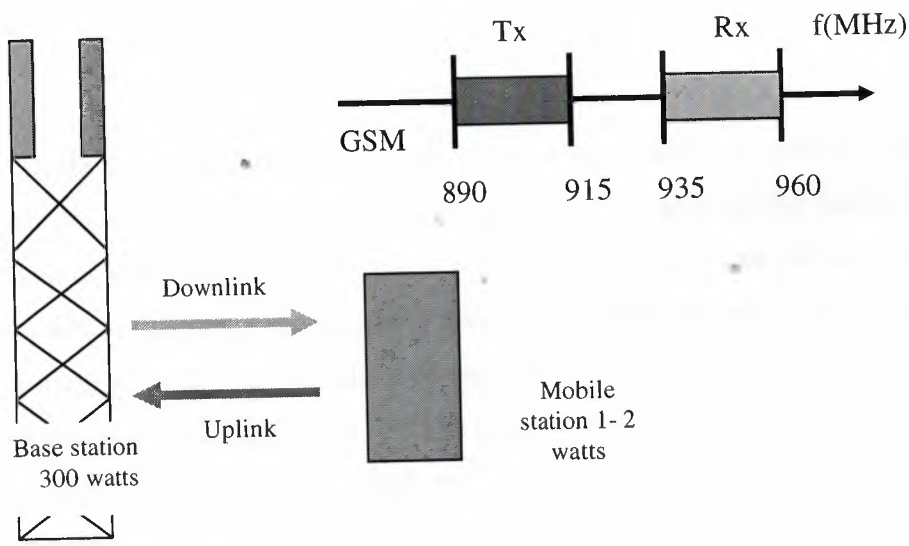


Figure 2.11: GSM900 Frequency Allocation.

2.8.1 Normal Burst

A normal burst consists of 148 bits, which is a mix of information bits and signaling bits.

Normal bursts are used to transmit information on both traffic and control channels. A normal burst contains 26 training sequence bits termed Equalization Bits. This sequence is one of the fixed combinations of bits that are known to the MS and the BS. The existence of multipath causes the original signal to fade. Therefore, it is difficult to exactly identify the transmitted signal at the receiver. The training bit sequence helps in estimating the channel response, optimize reception and reduce inter-symbol interference caused by propagation time differences in the multipath channel. The structure of a normal burst is represented in Figure 2.12

3	57	1	26	1	57	3
Tail bits	Data bits	CTRL bit	Equalization bits	CTRL bit	Data bits	Tail bits
{000}		{1/0}		{1/0}		{000}

Figure 2.12: Normal Burs.

Stealing flag bits reside on both sides of the training sequence. They can either be '1' or '0'. They are signaling bits that indicate whether the burst contains traffic data or signaling data. In addition to the stealing flags there are two blocks of 57 bits each containing channel-coded user data. Therefore, the actual voice data consists of 114 bits of the 148 total bits transmitted. Toward the end there are 3 tail bits on each side, which are set to '0'. The tail bits are required for the demodulation process. There is also a guard period that equals a bit period of 8.25 bits or 30.4 microseconds.

2.9 Ciphering

The purpose of ciphering is to encode the burst so that it cannot be interpreted by any device other than the intended receiver. The ciphering algorithm in GSM is called A5 algorithm. It does not add bits to the burst, meaning that the input and the output to the ciphering process are the same as the input: 456 bits per 20ms.

The security mechanisms of GSM are implemented in three different system elements; the Subscriber Identity Module (SIM), the GSM handset or MS, and the GSM network. The SIM contains the IMSI, the individual subscriber authentication key (Ki), the ciphering key generating algorithm (A8), the authentication algorithm (A3), as well as a Personal Identification Number (PIN). The GSM handset contains the ciphering algorithm (A5). The encryption algorithms (A3, A5, and A8) are present in the GSM network as well.

Distribution of security information is among the three system elements, the SIM, the MS, and the GSM network. Within the GSM network, the security information is further distributed among the authentication centre (AUC), the home location register (HLR) and the visitor location register (VLR). The AUC is responsible for generating the sets of RAND, SRES, and Kc, which are stored in the HLR and VLR for subsequent use in the authentication and encryption processes.

2.10 Modulation

GSM uses GMSK modulation, which stands for Gaussian Minimum Shift Keying; GMSK uses a Gaussian filter from which its name was derived. The frequency separation utilized is the minimum frequency separation required for the two modulating signals to be orthogonal over a signaling interval of length T. One of the advantages of GMSK modulation is that it does not produce any Intersymbol Interference (ISI). GMSK modulation is continuous phase modulation, which achieves a good compromise between power and bandwidth efficiency while maintaining a specified level of BER at the expense of only reasonable increase of system complexity.

The relation between premeditation bandwidth, B, and the bit period, T, defines bandwidth of the system. GSM uses BT as 0.3 with the modulating symbol rate of 270.833 Kbps. Prior to the first bit entering the modulator, the modulator resides in an internal state that represents the modulation bit stream consisting of consecutive ones had been entered to a differential encoder. After the last bit of the time slot the modulator returns to an internal state that represents the modulation of bit stream consisting of ones. These bits are called dummy bits and define the start and the stop of the active and useful part of the burst as presented in Figure 2.12.

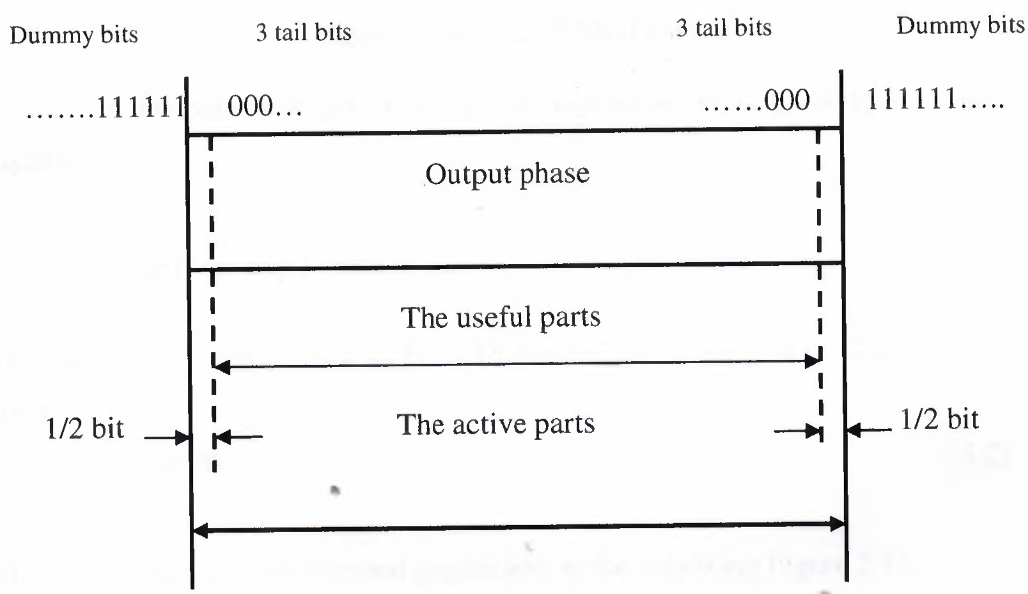


Figure 2.13: Relation between Active Part of Burst, Tail Bits and Dummy Bits.

Before the burst enters the modulator, it is differentially coded. The differential encoder sequence is then applied to the Gaussian filter in the modulator. The GMSK modulator is shown in Figure 2.14.

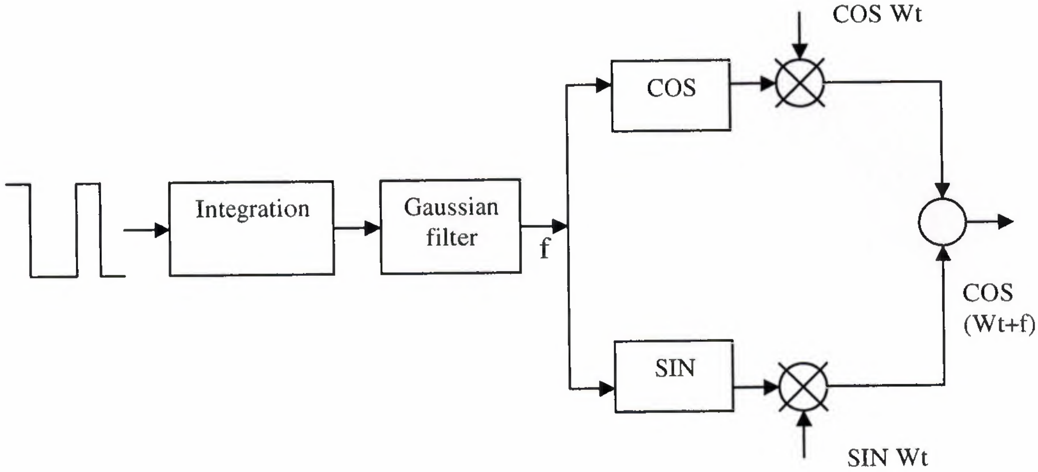


Figure 2.14: GMSK Modulation.

The Gaussian low pass filter has an impulse response given by the following equation

$$h(t) = (\pi)^{1/2}/a \ \exp(- (\pi^2 / a^2) t^2) \quad (3.1)$$

The parameter (a) is related to B, 3-dB bandwidth of base band Gaussian shaping filter,

$$a = 0.5887/B \quad (3.2)$$

The filter response is represented graphically in the following Figure 2.14.

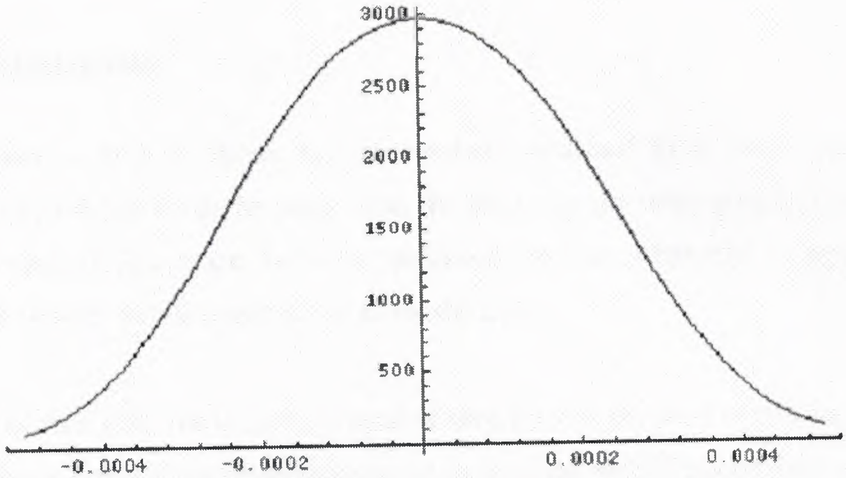


Figure 2.15: The Response of Gaussian Filter.

The output of the modulator is given by

$$m(t) = SIN(2\pi f_c t)I(t) + COS(2\pi f_c t)Q(t) \tag{3.3}$$

Where f_c is the carrier frequency used from the oscillator. The final output of the modulator, which is sent through the channel, is shown in figure 2.15.

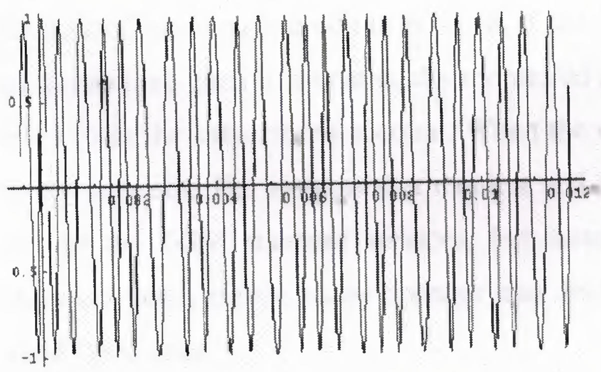


Figure 2.16: Final Output of the Modulator.

2.11 Voice call

2.11.1 Outgoing calls

Once a mobile phone has successfully attached to a GSM network as described above, calls may be made from the phone to any other phone on the global Public Switched Telephone Network assuming the subscriber has an arrangement with their "home" phone company to allow the call.

The user dials the telephone number then presses the send or talk key, and the mobile phone sends a call setup request message to the mobile phone network via the mobile phone mast (BTS) it is in contact with.

The element in the mobile phone network that handles the call request is the Visited Mobile Switching Center (Visited MSC). The MSC will check against the subscriber's temporary record held in the Visitor Location Register to see if the outgoing call is allowed. If so, the MSC then routes the call in the same way that a telephone exchange does in a fixed network.

If the subscriber is on a Pay As You Go tariff, then an additional check is made to see if the subscriber has enough credit to proceed. If not, the call is rejected. If the call is allowed to continue, then it is continually monitored and the appropriate amount is decremented from the subscriber's account. When the credit reaches zero, the call is cut off by the network. The systems that monitor and provide the prepaid services are not part of the GSM standard services, but instead an example of intelligent network services that a mobile phone operator may decide to implement in addition to the standard GSM ones.

2.11.2 Incoming calls

Step One: Contact the Gateway MSC

When someone places a call to a mobile phone, they dial the telephone number (also called a MSISDN) associated with the phone user and the call is routed to the mobile phone operator's Gateway Mobile Switching Centre. The Gateway MSC, as the name suggests, acts as the "entrance" from exterior portions of the Public Switched Telephone Network onto the provider's network.

As noted above, the phone is free to roam anywhere in the operator's network or on the networks of roaming partners, including in other countries. So the first job of the Gateway MSC is to determine the current location of the mobile phone in order to connect the call. It does this by consulting the Home Location Register (HLR), which, as described above, knows which Visitor Location Register (VLR) the phone is associated with, if any.

Step Two: Determine how to route the call

When the HLR receives this query message, it determines whether the call should be routed to another number (called a divert), or if it is to be routed directly to the mobile.

If the owner of the phone has previously requested that all incoming calls be diverted to another number, known as the Call Forward Unconditional (CFU) Number, then this number is stored in the Home Location Register. If that is the case, then the CFU number is returned to the Gateway MSC for immediate routing to that destination.

If the mobile phone is not currently associated with a Visited Location Register (because the phone has been turned off or is not in range) then the Home Location Register returns a number known as the Call Forward Not Reachable (CFNRc) number to the Gateway MSC, and the call is forwarded there. Many operators may set this value automatically to the phone's voice mail number, so that callers may leave a message. The mobile phone may sometimes override the default setting.

Finally, if the Home Location Register knows that the phone is in the jurisdiction of a particular Visited Location Register, then it will request a temporary number (called an MSRN) from that VLR. This number is relayed to the Gateway MSC, which uses it to route the call to another Mobile Switching Center, called the Visiting MSC.

Step Three: Ringing the phone

When the call is received by the Visiting MSC, the MSRN is used to find the phone's record in the Visited Location Register. This record identifies the phone's location area. Paging occurs to all mobile phone masts in that area. When the subscriber's mobile responds, the exact location of the mobile is returned to the Visited MSC. The VMSC then forwards the call to the appropriate phone mast, and the phone rings. If the subscriber answers, a speech path is created through the Visiting MSC and Gateway MSC back to the network of the person making the call, and a normal telephone call follows.

It is also possible that the phone call is not answered. If the subscriber is busy on another call (and call waiting is not being used) the Visited MSC routes the call to a pre-determined Call Forward Busy (CFB) number. Similarly, if the subscriber does not answer the call after a period of time (typically 30 seconds) then the Visited MSC routes the call to a pre-determined Call Forward No Reply (CFNRy) number. Once again, the operator may decide to set this value by default to the voice mail of the mobile so that callers can leave a message....

2.11.3 Encoding speech during the calls

During a GSM call, speech is converted from analogue sound waves to digital data by the phone itself, and transmitted through the mobile phone network by digital means (Though older parts of the fixed Public Switched Telephone Network may use analog transmission.)

The digital algorithm used to encode speech signals is called a codec. The speech codec's used in GSM are called Half-Rate (HR), Full-Rate (FR), Enhanced Full-Rate (EFR) and Adaptive Multirate (AMR). All codec's except AMR operate with a fixed data rate and error correction level.

2.12 Summary

In this chapter, we have seen the GSM transmission process and we have discussed all the working principles of every part contains in it, and we have known that GSM system uses GSMK to modulate the signals into a suitable form that matches the channel conditions and we also have discussed the outgoing call and incoming call with the steps of each one

3. Reception Process

3.1 Overview

In this chapter we going to talk about the reception process in the global system for mobile communication (GSM) and we will explain some main parameters and operations that will be important to the reception procedure such as demodulation, Deciphering, Burst Format and deinterleaving, after that we will talk about the candidate receivers use in GSM and configuration of four candidate GSM receivers employing various iterative schemes. And we will make comparison between fore different receivers.

3.2 Reception Procedure

3.2.1 Demodulation

Demodulation is the act of removing the modulation from an analog signal to get the original baseband signal back. Demodulating is necessary because the receiver system receives a modulated signal with specific characteristics and it needs to turn it to base-band.

There are several ways of demodulation depending on what parameters of the base-band signal are transmitted in the carrier signal, such as amplitude, frequency or phase. For example, if we have a signal modulated with a lineal modulation, like AM (Amplitude Modulated), we can use a synchronous detector. On the other hand, if we have a signal modulated with an angular modulation, we must use a FM (Frequency Modulated) demodulator or a PM (Phase Modulated) demodulator respectively. There are different kinds of circuits that make these functions.

An example of a demodulation system is a modem, which receives a telephone signal (electrical signal) and turns this signal from the wire net into a binary signal for the computer.

3.2.1.1 AM Demodulation

An AM signal can be rectified without requiring a coherent demodulator. For example, the signal can be passed through an envelope detector (a diode rectifier). The output will follow the same curve as the input baseband signal.

3.2.1.2 FM Demodulation

There are several ways to demodulate an FM signal. The most common is to use a discriminator. This is composed of an electronic filter which decreases the amplitude of some frequencies relative to others, followed by an AM demodulator. If the filter response changes linearly with frequency, the final analog output will be proportional to the input frequency, as desired. Another one is to use two AM demodulators; one tuned to the high end of the band and the other to the low end, and feed the outputs into a difference amp. Another is to feed the signal into a phase-locked loop and use the error signal as the demodulated signal.

3.2.1.3 PM Demodulation

Main article Phase modulation

3.2.1.4 QAM Demodulation

Main article QAM demodulation

3.2.1.5 Demodulation in GSM

The Demodulation used to recover the baseband signal that was modulated using by GMSK modulation. We used the GMSK Demodulator Baseband block shown in figure 3.2 demodulates a signal that was modulated using the Gaussian minimum shift keying method. The output of demodulator is shown in the figure below.

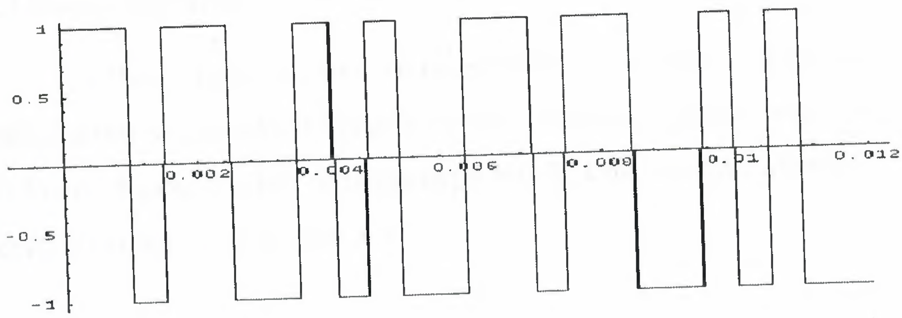


Figure 3.1: The Output of Demodulator.

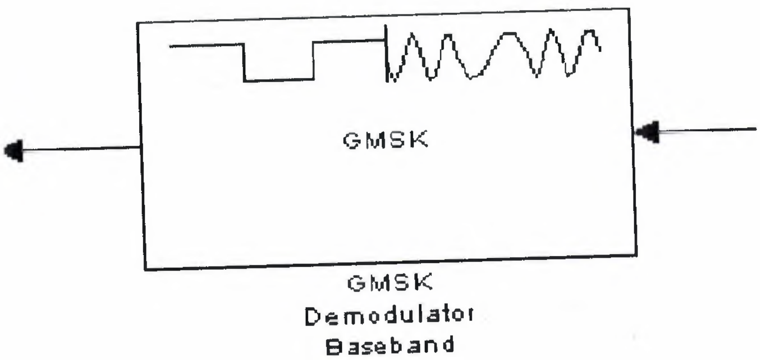


Figure 3.2: GMSK Demodulator.

3.2.2 Deciphering

A protection has been introduced in GSM by means of transmission ciphering. The ciphering method does not depend on the type of data to be transmitted (speech, user data or signaling) but is only applied to normal bursts.

Ciphering is achieved by performing an ``exclusive or" operation between a pseudo-random bit sequence and 114 useful bits of a normal burst (i.e. all information bits except the 2 stealing flags). The pseudo-random sequence is derived from the burst number and a key session established previously through signaling means. Deciphering follows exactly the same operation.

3.2.3 Burst Deformat

It is the reverse operation of Burst Format it is used to remove the additional bits which added in burst format (32 bits) for each frame. We used Sub-Matrix block to select only the main bits 114 bits (before addition operation) as shown in Figure 3.3

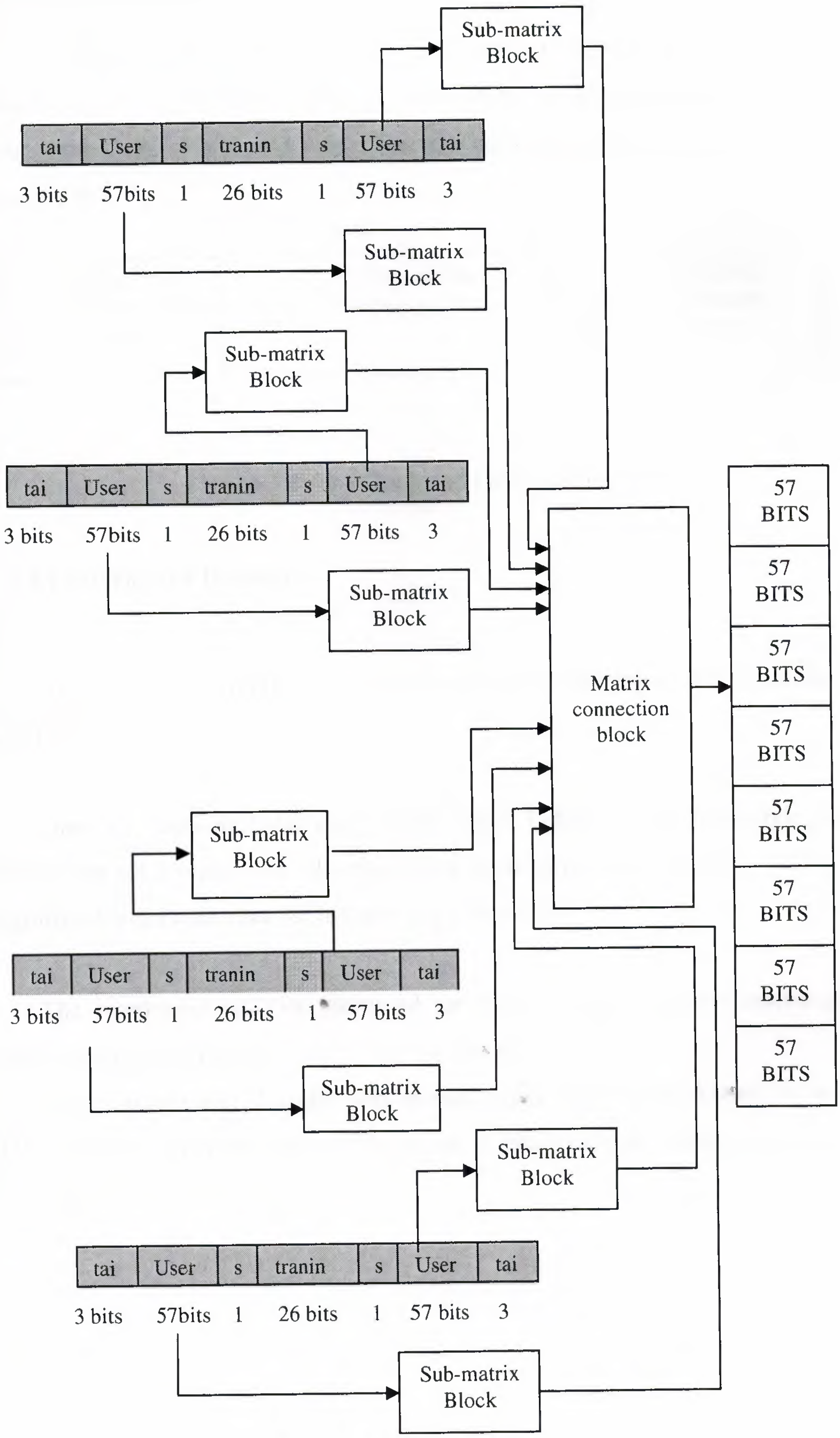


Figure 3.3 Burst Deformat.

3.2.4 DeInterleaving

In this stage the order of the 456 bits returns to the original order before Interleaving. This process is done by using Random deinterleaver Block that returns the order of bits as it was before the interleaving process, as shown in Figure 3.4.

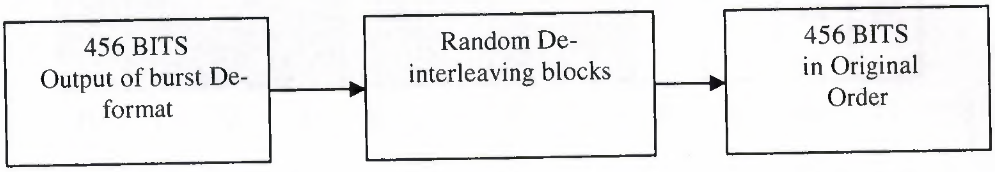


Figure 3.4: The Output of Deinterleaving.

3.2.4.1 Interleaver Decoder

The purpose of interleaving is to help the convolutional code to correct burst errors.

Since the convolutional code works much better for random single bit errors than for a burst error, the possibility of uncorrected burst errors will be significantly reduced after the interleaving process.

The interleaver encoder dispersed the bits from two 456-bit frames of speech data over 8 consecutive 114-bit sub frames.

So the interleaver decoder will reorder eight 114-bit sub frames to two 456-bit frames. Figure 4.5 shows the details of interleaver decoding process.

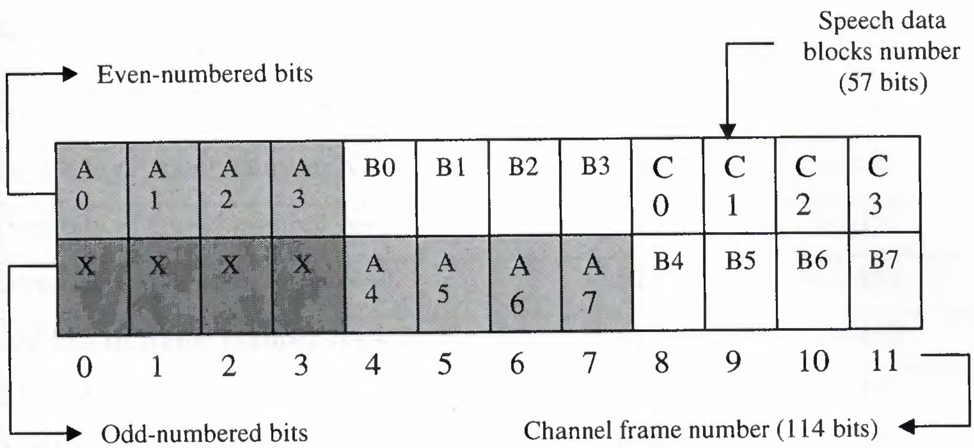


Figure 3.5: Diagonal interleaver decoding of speech data.

Two separate memory blocks are used, each containing eight 114-bit sub-frames (two frames of data after decoding).

The decoder operates by first reading 8 sub-frames of consecutive received data into the first memory block. For example in Figure 3.4, A0~A7, B0~B3 and four X ("padding bits") is the coded data in the first 8 sub-frames, this data will be reordered and stored in the second memory block. Then the first memory block is ready to receive new data. The decoder starts outputting data from the second Memory block after two frames of delay (Initial delay). Simultaneously, the first memory block starts receiving new data (B4~B7, C0~C7, D0~D3 in this case). The decoder will operate continuously following an initial delay to process the first 8 sub-frames.

Table 3.1 shows the locations of the bits of one 456-bit speech data frame in 8 consecutive received sub-frames. The decoder extracts these bits following the decoding algorithm.

Table 3.1 Reordering scheme for a traffic channel TCH.

Position within 26-frame structure (sub frame)	
Even bits of frame number N	08------------------448
Even bits of frame number N+1	19------------------449
Even bits of frame number N+2	210------------------450
Even bits of frame number N+3	311------------------451
Odd bits of frame number N+4	412------------------452
Odd bits of frame number N+5	513------------------453
Odd bits of frame number N+6	614------------------454
Odd bits of frame number N+7	715------------------455

The original received data is arranged in two frames (456-bits each) that each contains 8 sub frames (114-bits each) and stored in the first memory block (MEM) and the decoded data is stored in the second memory block (De-interleaved data).

The memory block (MEM_2D) is a 8 by 57 memory array and stores the table contents. The variable "framecount" could be 0 or 1 in order to indicate the first frame or the second frame.

So the algorithm divides the received data into 57-bit sub blocks (total of 8 sub blocks in 1 frame). First, it picks the first bit in each sub block and puts them in proper order into memory block two, and then it picks the second bit in each sub block and puts them into memory block two and so on. Each set of eight 57-bit sub blocks is decoded to one 456-bit frame.

3.2.5 Channel Decoding

The Viterbi decoder block is used to remove the addition bits that added by the convolutional coder at the transmission process. The input to this block is 378 bits from 456 bits that get out from deinterleaver. Also, the Sub-Matrix Block is used to split the 456 bits into 378 and 78 bits (Not Important Bits) as shown in Figure 3.6 Figure 3.7 represent the output of this decoder which is 189 bits.

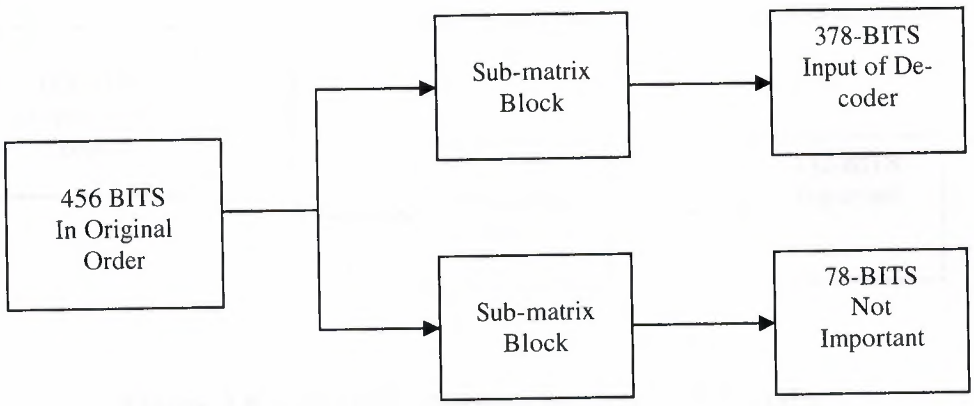


Figure 3.6: The Split Operation before Decoder.

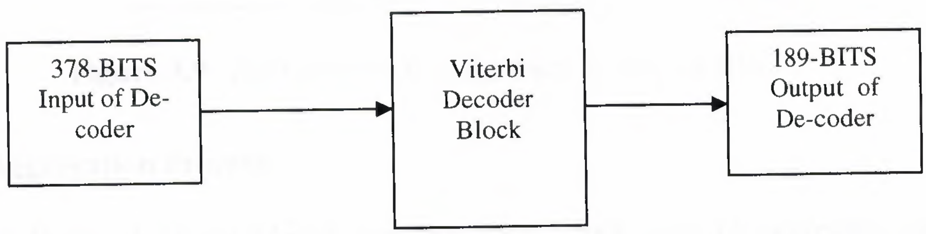


Figure 3.7: The Output of Decoder.

3.2.6 CRC Error Detector

The general CRC syndrome error detector block is used to detect the errors that occurs for the very important 50-bits and corrects these errors. The Sub Matrix divide the 189-bits output of Decoder to 50 bits(Very Important Bits) and 132 bits(Important Bits) as shown in Figure 3.8. In the Figure 3.9 the 50 bits entered to the CRC detector to fix errors that occur to these bits.

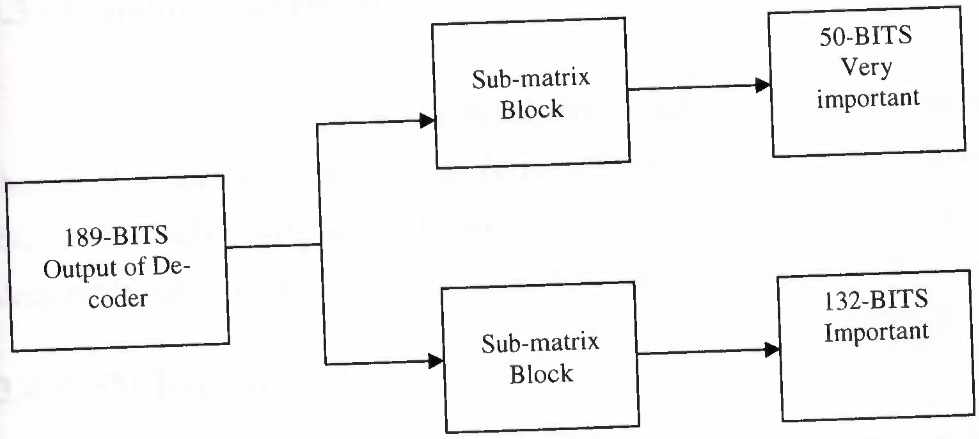


Figure 3.8: The Split Operation for Output of Decoder

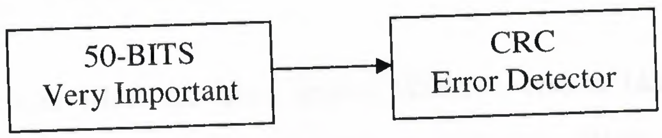


Figure 3.9: The Correction of the Very Important Bits.

3.2.7 Aggregation Process

In Figure 3.10 the Matrix Concatenation Block used to aggregate the three blocks to produce 260 received bits.

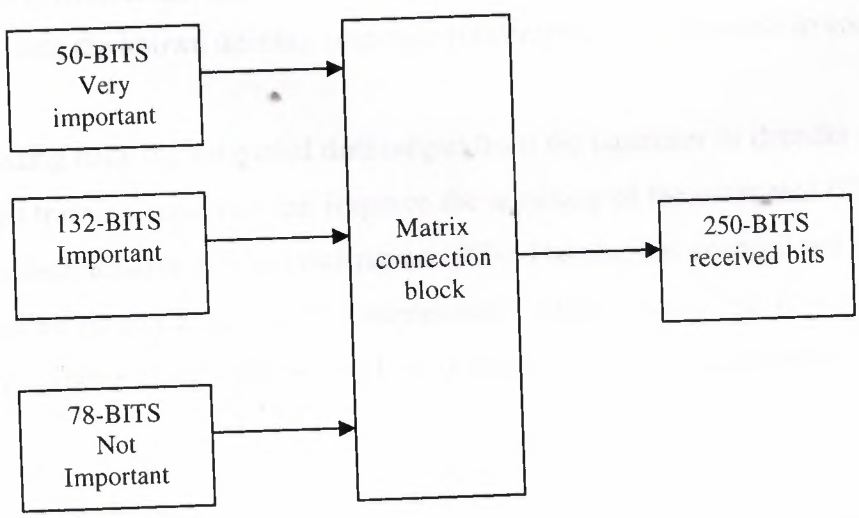


Figure 3.10: The Aggregation Operation.

3.3 Discontinuous reception

It is a method used to conserve the mobile station's power. The paging channel is divided into sub channels corresponding to single mobile stations. Each mobile station will then only 'listen' to its sub channel and will stay in the sleep mode during the other sub channels of the paging channel.

3.4 GSM Recevires

Intersymbol Interference (ISI) caused by multi-path propagation is a significant problem in digital mobile communication system. In addition, the channel characteristic is time varying due to the movement of the mobile station relative to its surrounding.

In a time division multiplex access (TDMA) system like GSM, the receiver must be able to estimate the channel and compensate for channel distortion adaptively.

The performance of this compensation is largely determine by the accuracy of the channel estimation.

In a conventional GSM receiver, the channel impulse response (CIR) is estimated using the known training sequence transmitted as a midamble in each burst.

Feeding back the estimated data output from the equalizer or decoder as an extended training sequence can improve the accuracy of the estimated CIR. Which is called iterative channel estimation (ICE) The channel encoder and ISI channel can be treated as a serially concatenated coding scheme, the decoding principle of Turbo Code can be used to perform iterative equalization and decoding.

Four different types of GSM receiver with these techniques were introduced. The complexity of implemented these receivers and the corresponding performances are compared here in terms of number of operations required in receiving one GSM block

3.3.1 Candidate GSM Receivers

The typical structure of the GSM digital receiver is shown in Figure 1. The channel impulse response (CIR) is estimated by a known training sequence or data sequence transmitted as a midamble in each GSM burst.

The channel is estimated using channel sounding (CS) or least square (LS) techniques, and this estimated CIR is used in equalization process to eliminate the impact of Intersymbol Interference (ISI). After deinterleaving, the decoder estimates the original message sequence. In this paper, we consider receivers which employ maximum a posteriori probability (MAP) algorithm for equalization and either maximum likelihood sequence estimation (MLSE) or MAP estimation in the decoder. The MAP algorithm is a softin soft-out (SISO) algorithm, which is well suited to iterative receiver schemes.

3.3.1.1 Conventional Training Sequence Receiver (TS)

In conventional training sequence receiver, the channel impulse response was estimated by the known training sequence stored at the receiver. MLSE or MAP is used to perform equalization and MLSE implemented by Viterbi algorithm to perform decoding. The output from the MAP equalizer will be soft value, which can improve the performance of decoder

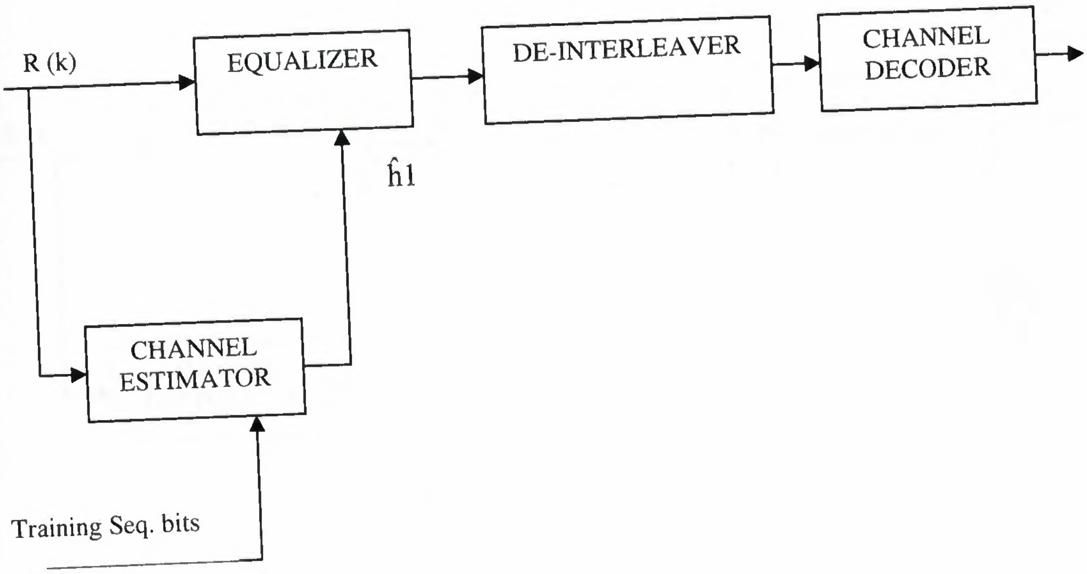


Figure3.11 conventional GSM receiver

3.3.1.2 Iterative Channel Estimation from Equalizer (EQ-ICE)

The effectiveness of the Equalization depends on the accuracy of the reliability of the estimated CIR. In order to get more accurately estimation, the equalization output can be feedback to the channel estimator to perform Iterative Channel Estimation (ICE), the performance is further improved using soft decision feedback.This configuration is shown in Figure 3.12.

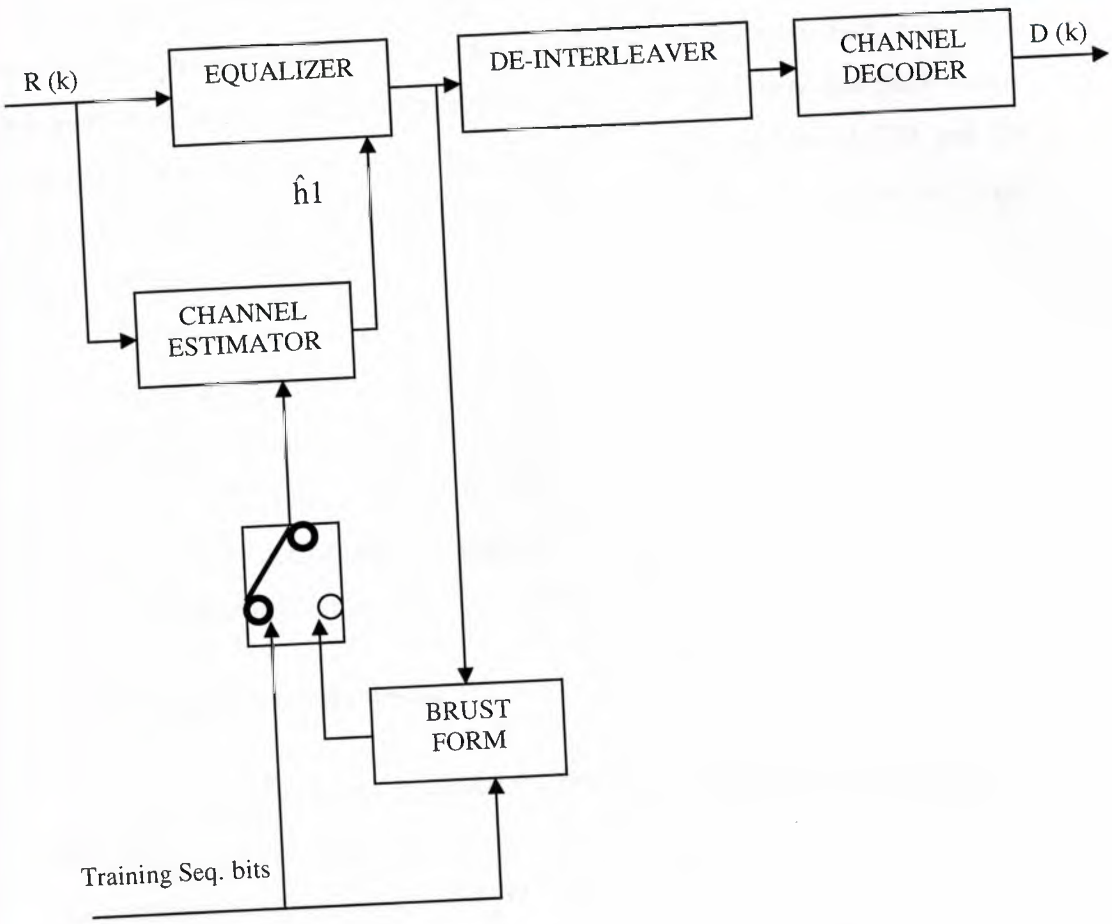


Figure 3.12 iterative channel estimation feedback from equalizer

3.3.1.3 Iterative Channel Estimation plus Iterative Equalization and Decoding (EQ-ICE+Turbo)

In GSM, the channel encoder and ISI channel can be viewed as a serially concatenated coding scheme, so the decoding principle of Turbo Code can be used to perform Iterative Equalization and Decoding. So in this configuration there are two iterative processes: One for channel estimation and the other for equalization and decoding.

3.3.1.4 Turbo Iterative Channel Estimation (Turbo-ICE)

As an extension to the EQ-ICE+Turbo receiver as described above, these two iterative processes can be combined to perform Turbo Iterative Channel Estimation. The output from the APP decoder were fed back to the channel estimator to re-estimate the channel, using the improved CIR and the extrinsic information as a priori knowledge, the APP equalizer will get improved data estimation.

MAP algorithm is used to perform these iterative processes. In order to reduce the complexity, a simplification of MAP algorithm: Max-Log-MAP [9] algorithm is used instead MAP in these receiver schemes. All the iterative processes can be performed any times without delay restriction. One iterative process is performed for each in our comparison.

3.3.2 Complexity Comparison

The complexity is evaluated by the total operation needed in receiving one GSM block. The interleaving and de-interleaving processes are ignored in this complexity calculation. Channel sounding is used as channel estimation algorithm for both training sequence estimation and iterative channel estimation. The operation needed for one GSM burst is shown in Table 3.2. For soft decision feedback, a look up table used to convert the log likelihood ration (LLR) of each transmitted data into soft values.

Tablel 3.2 the complexity of channel sounding

Operation	Channel sounding		
	Training Sequence Estimation	Interleaving Channel Estimation	
		Hard decision feedback	Hard decision feedback
Multiplication	MCIR Lt	MCIR (LB-MCIR)	MCIR (L-MCIR)
Addition	MCIR (Lt-1)	MCIR (LB-MCIR-1)	MCIR (LB-MCIR-1)
Looks-ups			LB-MCIR-26

M_{CIR}: number of estimated channel impulse response taps.

L_t: number of training sequence bits used in channel estimation.

L_B: length of GSM normal burst excludes ground bits.

Equalization is performed burst by burst and the decoding is taken when it receives the whole speech block. So the equalization and decoding complexity are estimated in term of the number of operation needed for equalizing one burst and decoding one speech block. Table 3.3 gives the complexity of Viterbi equalizer and decoder used in conventional training sequence receiver.

Table 3.3. Complexity of Viterbi equalization and decoding

Operation	Viterbi Equalizer from GSM	Viterbi decoder from GSM
Multiplication	$(M+2)\,2^{M}\,(L_B+M)$	$8\,2^{K}\,(L_{C1}+K)$
Addition	$(M+1)\,2^{M}\,(L_B+M)$	$4\,2^{K}\,(L_{C1}+K)$
Max Operation	$2^{M}\,(L_B+M)$	$2^{K}\,(L_{C1}+K)$

M: memory length of ISI channel, $M = M_{CIR} - 1$

LB: number of bits in one GSM normal burst excludes ground bits

K: Memory length of the channel encoder for GSM.

LC1: Length of Class1a bits in one GSM speech block.

Max-Log-Map is used to perform iterative channel estimation, equalization and decoding. The operation required for equalizer and decoder is presented in Table 3.4

Table 3.4 Complexity of Max-Log-MAP Equalizer and Decoder

Operation	Max-Log-MAP Equalizer				
	Branch Transition Probability γ Calculation	Forward Process α Calculation	Backward Process β Calculation	Log Likelihood Function	Total (Approximation)
Multiplication	2(M+ 3) M ,2 (LB+M)	M 2 ,2 (LB+M)	M 2 ,2 (LB+M)	M 5 ,2LB	2 (2 ,M+ 10M+10L B+ 2MLB) M ,2
Addition	2(M+ 3) M ,2 (LB+M)				(2M+6) (LB+M) M ,2
Max operation		M ,2 (LB+M)		M 2 ,2LB	(4LB+2M) M ,2

LB: Number of bits in one GSM normal burst excludes guard bits.
M: Memory Length of the ISI channel. M = MCIR – 1

Operation	Max-Log-MAP Decoder					
	Branch Transition Probability γ Calculation	Forward Process α Calculation	Backward Process β Calculation	Log Likelihood Function (Info Bits)	Log Likelihood Function (Coded Bits)	Total (Approximation)
Multiplication	k 4 ,2 (LC+1+K)	k 2 ,2 (LC1+K)	k 2 ,2 (LC1+K)	k 5 ,2 LC1	k 2 ,2 LC1	(13LC1 +6K) k ,2
Addition	k 6 ,2 (LC1+K)					6(LC1+ k K) ,2
Max operation		k ,2 (LC1+K)	k ,2 (LC1+K)	k 2 ,2 LC1	k 4 ,2 LC1	(8LC1+ 2K) k ,2

K: Memory Length of Channel Encoder for GSM

L_{C1}: Length of Class1a bits in one GSM speech block

The number of process times required in receiving one GSM speech block is shown in Table 3.5. One iteration is investigated for each configuration because the simulation result shows that most of the improvement is achieved by the first iteration. The GSM speech block are divided into 8 sub-block and interleaved into 8 GSM burst, only receiving the first speech block required equalization of 8 burst. For the other speech block, 4 sub-block data have been equalized in the previous 4 bursts due to the interleaving process. In our simulation:

- Estimated channel impulse response taps: = *5 CIR M*
- Memory Length of the ISI channel = −1 = 4 CIR M M
- Channel Encoder Memory of GSM: $K = 4$
- Length of one GSM Normal Burst: =148 *B L*
- Length of training sequence used in channel estimation:= 16 *t L*

- Length of coded Class 1 bits in one speech block: 378 1 = C L

In Table 3.6 we give the exact number of operation required for each type of GSM receiver in receiving one GSM speech block. Max-log-MAP is used in conventional GSM receiver to perform equalization.

Table 3.5 Configuration of different receiver schemes

RECEIVER TYPE	Number of Operation in Receiving one GSM Speech Block					
	TS Channel Estimation	Iterative Channel Estimation	Equalization		Decoding	
			Viterbi	Max-Log MAP	Viterbi	Max-Log MAP
TS	4	0		4	1	
EQ-ICE	4	4		8		1
EQICE+ Turbo (N iteration)	4	4		4(2+N)		(1+N)
Turbo-ICE (N-iteration)	4	4N		4(1+N)		(1+N)

TS: Training Sequence Only Receiver using Max-log-MAP equalization.

EQ-ICE: Iterative Channel Estimation Feedback from Equalizer.

EQ-ICE + Turbo: Iterative Channel Estimation Feedback from Equalization plus Iterative Equalization and Decoding.

Turbo-ICE: Iterative Channel Estimation Feedback from Decoder with Iterative Equalization and Decoding.

Table 3.6 Complexity of different receiver schemes

Operation	RECEIVER TYPE						
	TS	EQ-ICE		EQ-ICE+TURBO (1 Iteration)		TURBO-ICE (1 Iteration)	
		Hard Decision	Soft Decision	Hard Decision	Soft Decision	Hard Decision	Soft Decisi on
Addition	238652	432536	432536	686468	686468	511364	51136 4
Multiplicati on	160960	312236	312236	485100	485100	426732	42673 2
Max Operation	44512	125312	125312	212224	212224	173824	17382 4
Look-ups			976		976		976

3.4 Summary

In this chapter, we have seen the GSM reception process and we have seen the component of the reception system and we explained the procedure of each one in the reception process sequence and then we have seen different kind of receiver used in the global system for mobile communication and we also saw comparison between complexity and performance of 4 different types of GSM receiver: Conventional training sequence channel estimation receiver, iterative channel estimation feedback from equalizer, iterative channel estimation plus iterative equalization and decoding, and combination of iterative channel estimation and Iterative equalization and decoding. The combination of iterative channel estimation and iterative equalization and decoding has the best trade off among the receivers with iterative processes.

4. GSM INTERFACE

4.1 overview

In this chapter we going to talk about the radio interface, the radio interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces of the GSM system.

The spectrum efficiency depends on the radio interface and the transmission, more particularly in aspects such as the capacity of the system and the techniques used in order to decrease the interference and to improve the frequency reuse scheme. The specification of the radio interface has then an important influence on the spectrum efficiency.

And we will talk about the Air-interface, The Air-interface is the central interface of every mobile system and typically the only one to which a customer is exposed. The physical characteristics of the Air-interface are particularly important for the quality and success of a new mobile standard. For some mobile systems, only the Air-interface was specified in the beginning, like IS-95, the standard for CDMA. Although different for GSM, the Air-interface still has received special attention. Considering the small niches of available frequency spectrum for new services

4.2 The GSM RADIO INTERFACE

4.2.1 Overview

The radio interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces of the GSM system.

One of the main objectives of GSM is roaming. Therefore, in order to obtain a complete compatibility between mobile stations and networks of different manufacturers and operators, the radio interface must be completely defined.

The spectrum efficiency depends on the radio interface and the transmission, more particularly in aspects such as the capacity of the system and the techniques used in order to decrease the interference and to improve the frequency reuse scheme. The specification of the radio interface has then an important influence on the spectrum efficiency.

4.2.2 Frequency allocation

Two frequency bands, of 25 Mhz each one, have been allocated for the GSM system:

1-The band 890-915 Mhz has been allocated for the uplink direction (transmitting from the mobile station to the base station).

2-The band 935-960 Mhz has been allocated for the downlink direction (transmitting from the base station to the mobile station).

But not all the countries can use the whole GSM frequency bands. This is due principally to military reasons and to the existence of previous analog systems using part of the two 25 Mhz frequency bands.

4.2.3 Multiple access scheme

The multiple access scheme defines how different simultaneous communications, between different mobile stations situated in different cells, share the GSM radio spectrum. A mix of Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA), combined with frequency hopping, has been adopted as the multiple access schemes for GSM.

4.2.3.1 FDMA and TDMA

Using FDMA, a frequency is assigned to a user. So the larger the number of users in a FDMA system, the larger the number of available frequencies must be. The limited available radio spectrum and the fact that a user will not free its assigned frequency until he does not need it anymore, explain why the number of users in a FDMA system can be "quickly" limited.

On the other hand, TDMA allows several users to share the same channel. Each of the users, sharing the common channel, are assigned their own burst within a group of bursts called a frame. Usually TDMA is used with a FDMA structure.

In GSM, a 25 Mhz frequency band is divided, using a FDMA scheme, into 124 carrier frequencies spaced one from each other by a 200 kHz frequency band. Normally a 25 Mhz frequency and can provide 125 carrier frequencies but the first carrier frequency is used as a guard band between GSM and other services working on lower frequencies.

Each carrier frequency is then divided in time using a TDMA scheme. This scheme splits the radio channel, with a width of 200 kHz, into 8 bursts. A burst is the unit of time in a TDMA system, and it lasts approximately 0.577 ms. A TDMA frame is formed with 8 bursts and lasts, consequently, 4.615 ms. Each of the eight bursts, that form a TDMA frame, are then assigned to a single user.

4.2.4 GSM Channel Structure

The GSM standard not only specifies then "when" of different channels in those different types of information is transmitted in different burst periods, frames, multi-frames super-frames etc.

It also distinguish the "why" of the information under the phrase of "logical channels", For example, it is not sufficient to identify between TCH and CCH. The GSM standard identifies the different types of CCH and TCH that are used.

Depending on the kind of information transmitted (user data and control signaling), we refer to different logical channels, which are mapped under physical channels (slots).

Digital speech is sent on a logical channel named TCH, which during the transmission can be an allocated to a certain physical channel. In a GSM system no RF channel and no slot is dedicated to a priori to the exclusive use of anything (any RF channel can be used for number of different uses).

Logical channels are divided into two categories:

I) Traffic Channels (TCHs)

ii) Control Channels.

A channel corresponds to the recurrence of one burst every frame. It is defined by its frequency and the position of its corresponding burst within a TDMA frame. In GSM

There are two types of channels:

1-The traffic channels used to transport speech and data information.

2-The control channels used for network management messages and some channel maintenance tasks, We have already introduced the physical channels used in GSM, namely 8 burst periods per frame on an FDMA carrier.

We have also seen the need for the transmission of two distinct types of information between MS and BS, namely control (signaling) and user traffic information, This leads to the concept of two types of channels: Traffic Channel (TCH) used to convey user traffic information, Control Channels (CCH) used to convey signaling information between MS and network

Typically, burst period 0 in a frame is used (in both directions) as a CCH, Remaining seven burst periods in the TDMA are "nominally" TCHs, However, and this simple picture is not the complete picture.

We have already seen that the normal burst in a burst period which carries TCH can be "stolen" to carry specific types of "urgent" signalling information, Up to four consecutive frames can be stolen for this Fast Associated Control Channel (FACCH), For example, the 26 channel multi-frame structure applies to burst periods used as TCH, in this multi-frame structure, in frames 0 to 11; the burst period acts as a TCH, In frame 12, it acts as a means of transmitting specific type of control information (Slow Associated Control Channel - SACCH). In frames 13 to 24, it again acts as a TCH, in frame 25; it is actually unused to allow the MS to do other tasks.

Similarly, the 51 frame multi-frame used on burst period carrying certain CCH (e.g. burst period 0) is used in a similarly manner to separate when different "types" of signalling information (or channels) are transmitted

4.2.4.1 Traffic channels (TC)

A traffic channel (TCH) is used to carry speech and data traffic. Traffic channels are defined using a 26-frame multiframe, or group of 26 TDMA frames. The length of a 26-frame multiframe is 120 ms, which is how the length of a burst period is defined (120 ms divided by 26 frames divided by 8 burst periods per frame). Out of the 26 frames, 24 are used for traffic, 1 is used for the Slow Associated Control Channel (SACCH) and 1 is currently unused. TCHs for the uplink and downlink are separated in time by 3 burst periods, so that the mobile station does not have to transmit and receive simultaneously, thus simplifying the electronics.

TCHs carry either encoded speech or user data in both up and down directions in a point-to-point communication.

There are two types of TCHs that are differentiated by their traffic rates.

1-Full Rate TCH

2-Half Rate TCH

Full Rate TCH (Also represented as Bm) It carries information at a gross rate of 22.82 Kbps, Half Rate TCH carries information with half of full rate channels.

Full-rate traffic channels (TCH/F) are defined using a group of 26 TDMA frames called a 26-Multiframe. The 26-Multiframe lasts consequently 120 ms. In this 26-Multiframe structure, the traffic channels for the downlink and uplink are separated by 3 bursts. As a consequence, the mobiles will not need to transmit and receive at the same time, which simplifies considerably the electronics of the system.

The frames that form the 26-Multiframe structure have different functions:

1- 24 frames are reserved to traffic.

2- 1 frame is used for the Slow Associated Control Channel (SACCH).

3- the last frame is unused. This idle frame allows the mobile station to perform other functions, such as measuring the signal strength of neighboring cells.

Half-rate traffic channels (TCH/H), which double the capacity of the system, are also grouped in a 26-Multiframe but the internal structure is different, TCH are also classified accord to the type of traffic that they are carrying

The main ones are:

1-TCH/F: Full rate speech codec traffic channel (1 per burst period)

2-TCH/H: Half rate speech codec traffic channel (2 per burst period)

3-TCH/n: n (e.g. 9.6, 4.8) kbps data traffic channel (1 per burst period).

4.2.4.2 Control Channels

Basic structure of Control channel

1	2	3	4	.	.	.	.	.	10	11	.	.	.	.	.					21						26

F	S	x	X	X	X	X	X	X	X	F	S	X	X	X	X	X	X	X	X	F	S	X	X

Figure 4.1 Basic structure of Control channel

Actually in the above diagram S will be at slot 1 of next frame, F is frequency correction channel, which occurs every 10^{th} burst. The next frame to S contains service operator's information. There are four important different classes of control channels defined:

1-Broadcast Channels (BCH)

2-Common Control Channels (CCCH)

3-Dedicated Control Channels (DCCH)

4-Associated Control Channels (ACCH)

Each class is further subdivided to identify specific "logical channels", The mapping of these "logical" channels onto "physical" channels is quite complex but

Some examples have already been mentioned

- Broadcast Channels

Which gives to the mobile station the training sequence needed in order to demodulate the information transmitted by the base station, Broadcast channels are transmitted by the base station to convey "information" to ALL MS in the cell Three different "logical" BCH exist information necessary for the MS to register in the system.

1- The Broadcast Control Channel (BCCH)

This gives to the mobile station the parameters needed in order to identify and access the network. BCCH is a point-to-multipoint unidirectional control channel from the fixed subsystem to MS that is intended to broadcast a variety of information to MSs, BCCH has 51 bursts. BCCH is dedicated to slot1 and repeats after every 51 bursts.

Broadcast Control Channel (BCCH) continually broadcasts, on the downlink, information including base station identity, frequency allocations, and frequency hopping sequences. This provides general information per BTS basis (cell specific information) including information necessary for the MS to register at the system. After initially accessing the mobile, the BS calculates the requires

MS power level and sets a set of power commands on these channels. Other information sent over these channels includes country code network code, local code, PLMN code, RF channels used with in the cell where the mobile is located, and surrounding cells, hopping sequence number, mobile RF channel number for allocation, cell selection parameters, and RACH description. One of the important messages on a BCCH channel is CCCH_CONF, which indicates the organization of the CCCHs. This channel is used to down link point-to-multipoint communication and is unidirectional; there is no corresponding uplink. The signal strength is continuously measured by all mobiles which may seek a hand over from its present cell and thus it is always transmitted on designated RF channel using time slot 0(zero). This channel is never kept idle-either the relevant messages are sent or a dummy burst is sent.

2- Frequency correction channel (FCCH)

The Frequency-Correction Channel (FCCH), which supplies the mobile station with the frequency reference of the system in order to synchronize it with the network (FCCH), is used to allow an MS to accurately tune to a BS. The FCCH carries information for the frequency correction of MS downlink. It is required for the correct operation of radio system. This is also a point-to multipoint communication. This allows an MS to accurately tune to a BS.) conveys all information required by the MS to access and identify the network - transmitted in burst period 0 on only one (non-hopping) carrier in a cell The BCCH is a point-to-multipoint unidirectional control channel from the fixed subsystem to MS that is intended to broadcast a variety of information to MSs, including information necessary for the MS to register in the system. BCCH has 51 bursts. BCCH is dedicated to slot1 and repeats after every 51 bursts.

3- Synchronization channel (SCH)

Which gives to the mobile station the training sequence needed in order to demodulate the information transmitted by the base station (SCH) is used to provide TDMA frame oriented synchronization data to a MS. When a mobile recovers both FCCH and SCH signals, the synchronization is said to be complete.

SCH repeats for every 51 frames. SCH carries information for the frame synchronization (TDMA frame number of the MS

And the identification of BTS). This is also required for the correct operation of the mobile.

The Synchronization Channel contains 2 encoded parameters:
1-BTS identifications code (BSIC)
2- Reduced TDMA frame number (RFN).

- Common Control Channels (CCCH)

A CCCH is a point-to-multipoint (bi-directional control channel) channel that is primarily intended to carry signaling information necessary for access management functions (e.g., allocation of dedicated control channels). The CCCH channels help to establish the calls from the mobile station or the network. Three different types of CCCH can be defined:

The CCCH includes:
1- Paging channel (PCH)

Which is used to search (page) the MS in the downlink direction, The Paging Channel (PCH). It is used to alert the mobile station of an incoming cal

2- Random access channel (RACH)

The Random Access Channel (RACH), which is used by the mobile station to request access to the network which is used by MS to request of an SDCCH either as a page response from MS or call origination/ registration from the MS. This is uplink channel and operates in point-point mode (MS to BTS).This uses slotted ALOHA protocol. This causes a possibility of contention. If the mobiles request through this channel is not answered with in a specified time the MS assumes that a collision has occurred and repeats the request. Mobile must allow a random delay before re-initiating the request to avoid repeated collision. It is used by MS when it attempts to request access to the network

3- Access grant channel (AGCH)

Which is a downlink channel used to assign a MS to a specific SDCCH or a TCH. AGCH operates in point-to-point mode. A combined paging and access grant channel is designated as PAGCH. The Access Grant Channel (AGCH). It is used, by the base station, to inform the mobile station about which channel it should use. This channel is the answer of a base station to a RACH from the mobile station _Access Grant Channel (AGCH) is used by BS to tell MS which DCH to use after it has sent a message over the RACH

- Dedicated Control Channels (DCCH)

The Standalone Dedicated Control Channels (SDCCH) are allocated to specific mobiles to exchange information with the network for a specific duration A typical use of the SDCCH would be to exchange signalling relating to a call set up.

A DCCH is a point-to-point, directional control channel. The DCCH channels are used for message exchange between several mobiles or a mobile and the network. Two different types of DCCH can be defined:

Two types of DCCHs used are:

1- Standalone DCCH (SDCCH)

Is used for system signaling during idle periods and call setup before allocating a TCH, for example MS registration, authentication and location updates through this channel.

When a TCH is assigned to MS this channel is released. Its data rate is one-eighth of the full rate speech channel which is achieved by transmitting data over the channel once every eighth frame. The channel is used for uplink and downlink and is meant for point-to-point usage, it is used in order to exchange signaling information in the downlink and uplink directions.

2- The slow associated control channels (SACCH)

Is data channel carrying information such as measurement reports from the mobile of received signal strength for a serving cell as well as the adjacent cells, This is necessary channel for the assisted over hand over function, is also used for power regulation of MS and time alignment and is meant for uplink and down link. It is used for point-to-point communication. SACCH can be linked to TCH or an SDCCH.

- Associated Control Channels

Two types of ACH, which have already been mentioned:

1-Slow ACH (SACCH) which is transmitted in the TCH burst period once every TCH multi-frame and is used for signalling of a non-urgent nature relating to the call (e.g. supplementary service and call related requests)

2-Fast ACH (FACCH) which is formed by "stealing" up to four consecutive TCH bursts (frames) to convey "urgent" signalling information (e.g. handover, power control, timing advance) The Fast Associated Control Channels (FACCH) replace all or part of a traffic channel when urgent signaling information must be transmitted. The FACCH channels carry the same information as the SDCCH channels.

It is a DCCH whose allocation is linked to the allocation of a CCH. A FACCH or burst stealing is a DCCH obtained by pre-emptive dynamic multiplexing on a TCH.

A FACCH is also associated to TCH. FACCH works in a stealing mode. This means that if suddenly during a speech transmission it is necessary to exchange signaling information with the system at a rate much higher than the SACCH can handle, then 20 ms speech (data) bursts are stolen for signaling purposes. This is the case at the case at the hand over. The user will not hear the interruption of the speech since it lasts only for 20 ms and cannot sensed by human ears.

4.2.5 Structure of TDMA Slot with a Frame

There are five different kinds of bursts in the GSM system. They are:

1-Normal Burst

2- Synchronization Burst

3-Frequency Correction Burst

4- Access Burst

5- Dummy Burst

4.2.5.1 Normal Burst

This burst is used to carry information on the TCH and on control channels. The lowest bit number is transmitted first. The encrypted bits are 57 bits of data or (speech + 1 bit stealing flag) indicating whether the burst was stolen for FACCH signaling or not. The reason why the training sequence is placed in the middle is that the channel is constantly changing. By having it there, the chances are better that the channel is not too different when it affects the training sequence compared to when the information bits were affected. If the training sequence is put at the beginning of the burst, the channel model that is created might not be valid for the bits at the end of a burst there are 8 training sequences shown at the diagram. The 26 bits equalization patterns are determined at the time of the call setup.

Tail Bits (TB) always equal (0,0,0), which has bit location from 0 to 2 and 145 to 147.

The Guard Period are the empty spaced bits and are used to synchronize the burst with exact accuracy and makes sure that different time.

4.2.5.2 Synchronization Burst

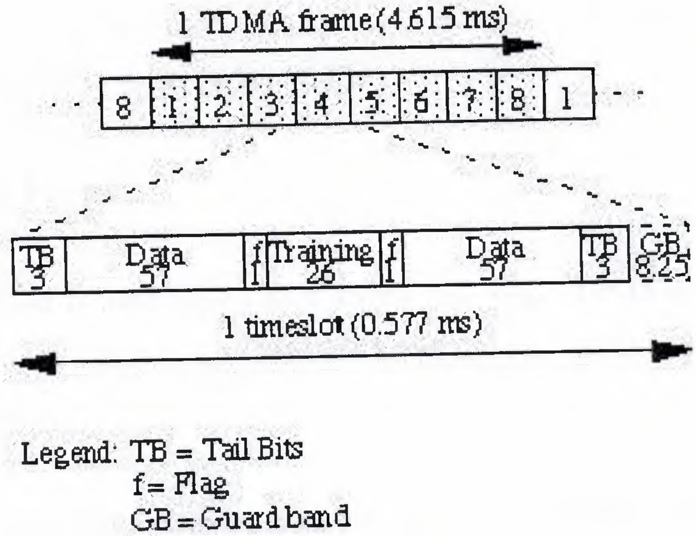


Figure 4.2 GSM TDMA structure and normal burst number of bits per field below the field legend

This burst is used for the time synchronization of the mobile. It contains 64 bit synchronization sequence. The encrypted 78 bits carry information of the TDMA frame number along with the BSIC. It is broadcast together with the correction burst. The TDMA frame is broadcast over SCH, in order to protect the user information against eavesdropping, which is accomplished is ciphering the information before transmitting. The algorithm that calculates the ciphering key uses a TDMA frame number as one of the parameters and therefore, every frame must have a frame number. By knowing the TDMA frame number, the mobile will know what kind of logical channel is being transmitted on the control channel TS0. BSIC is also used by the mobile to check the identity of the BTS when making signal strength measurements (to prevent measurements on co-channel cells).

4.2.5.3 Frequency Correction Burst

This burst is used for frequency synchronization of the mobile. It is equivalent to an un-modulated channel with a specific frequency offset. The repetitions of these bursts are called FCCH.

4.2.5.4 Access Burst

This burst is used for random access and longer GP to protect for burst transmission from a mobile that does not know the timing advance when it must access the system.

This allows for a distance of 35 km from base to mobile. Incase the mobile is far away from the BTS, the initial burst will arrive late since there is no timing advance on the first burst. The delay must be shorter to prevent it from overlapping a burst in the adjacent time-slot following this.

4.2.5.5 Dummy Burst

It is sent from BTS on some occasions as discussed previously which carries no information and has the format same as the normal burst. The normal burst is used to carry speech or data information. It lasts approximately 0.577 ms and has a length of 156.25 bits. Its structure is presented in figure 3.

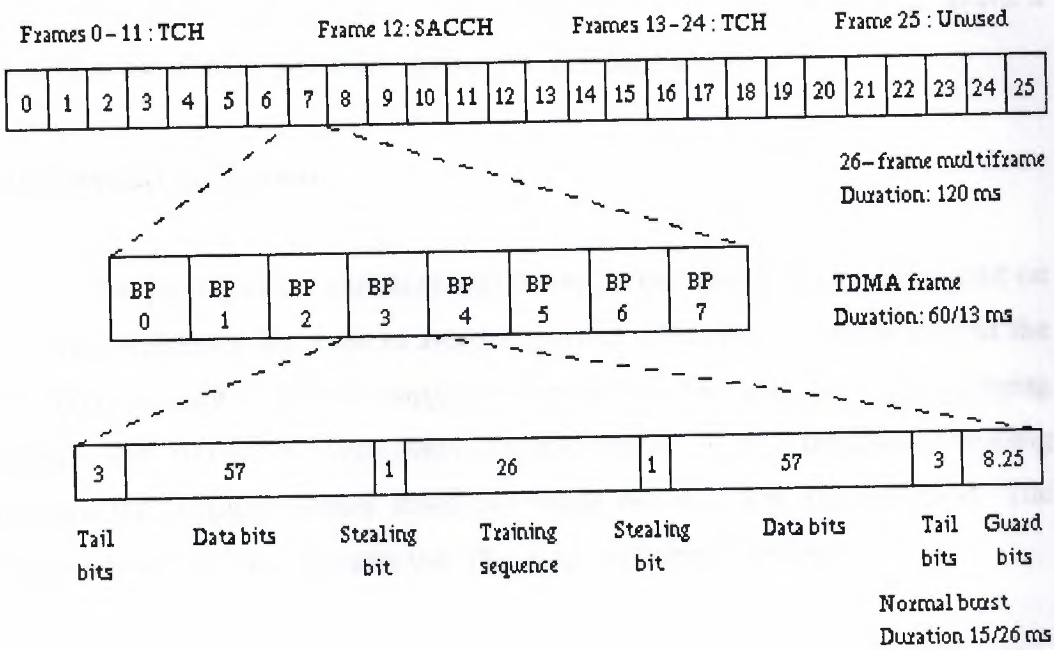


Figure 4.3: Structure of the 26-Multiframe, the TDMA frame and the normal burst

This figure has been taken, with the corresponding authorization, from "An Overview of GSM" by John Scourias (see Other GSM sites)

The tail bits (T) are a group of three bits set to zero and placed at the beginning and the end of a burst. They are used to cover the periods of ramping up and down of the mobile's power.

The coded data bits correspond to two groups, of 57 bits each, containing signaling or user data.

The stealing flags (S) indicate, to the receiver, whether the information carried by a burst corresponds to traffic or signaling data.

The training sequence has a length of 26 bits. It is used to synchronize the receiver with the incoming information, avoiding then the negative effects produced by a multipath propagation.

The guard period (GP), with a length of 8.25 bits, is used to avoid a possible overlap of two mobiles during the ramping time.

4.2.6 Frequency Hopping

The propagation conditions and therefore the multipath fading depend on the radio frequency. In order to avoid important differences in the quality of the channels, the slow frequency hopping is introduced. The slow frequency hopping changes the frequency with every TDMA frame. A fast frequency hopping changes the frequency many times per frame but it is not used in GSM. The frequency hopping also reduces the effects of co-channel interference.

There are different types of frequency hopping algorithms. The algorithm selected is sent through the Broadcast Control Channels, Even if frequency hopping can be very useful for the system, a base station does not have to support it necessarily On the other hand, a mobile station has to accept frequency hopping when a base station decides to use it.

4.3 THE AIR-INTERFACE OF GSM

The Air-interface is the central interface of every mobile system and typically the only one to which a customer is exposed.

The physical characteristics of the Air-interface are particularly important for the quality and success of a new mobile standard. For some mobile systems, only the Air-interface was specified in the beginning, like IS-95, the standard for CDMA. Although different for GSM, the Air-interface still has received special attention. Considering the small niches of available frequency spectrum for new services, the efficiency of frequency usage plays a crucial part. Such efficiency can be expressed as the quotient of transmission rate (kilobits per second) over bandwidth (kilohertz). In other words, how much traffic data can be squeezed into a given frequency spectrum at what cost?

The answer to that question eventually will decide the winner of the recently erupted battle among the various mobile standards.

4.3.1 Structure of the Air-Interface in GSM

4.3.1.1 The FDMA/TDMA Scheme

GSM utilizes a combination of frequency division multiple access (FDMA) and time division multiple access (TDMA) on the Air-interface. That results in a two-dimensional channel structure, which is presented in Figure 3.1. Older standards of mobile systems use only FDMA (an example for such a network is the C-Nets in Germany in the 450 MHz range). In such a pure FDMA system, one specific frequency is allocated for every user during a call. That quickly leads to overload situations in cases of high demand. GSM took into account the overload problem, which caused most mobile communications systems to fail sooner or later, by defining a two-dimensional access scheme. In fullrate configuration, eight time slots (TSs) are mapped on every frequency; in a half rate configuration there are 16 TSs per frequency.

In other words, in a TDMA system, each user sends an impulse like signal only periodically, while a user in a FDMA system sends the signal permanently. The difference between the two is illustrated in Figure 3.2. Frequency 1 (fl) in the figure represents a GSM frequency with one active TS, that is, where a signal is sent once per TDMA frame. That allows TDMA to simultaneously serve seven other channels on the same frequency (with full rate configuration) and manifests the major advantage of TDMA over FDMA (f2).

The spectral implications that result from the emission of impulses are not discussed here. It needs to be mentioned that two TSs are required to support duplex service, that is, to allow for simultaneous transmission and reception. Considering that Figures 4.4 and 4.5 describe the downlink, one can imagine the uplink as a similar picture on another frequency.

GSM uses the modulation technique of Gaussian minimum shift keying (GMSK). GMSK comes with a narrow frequency spectrum and theoretically no amplitude modulation (AM) part.

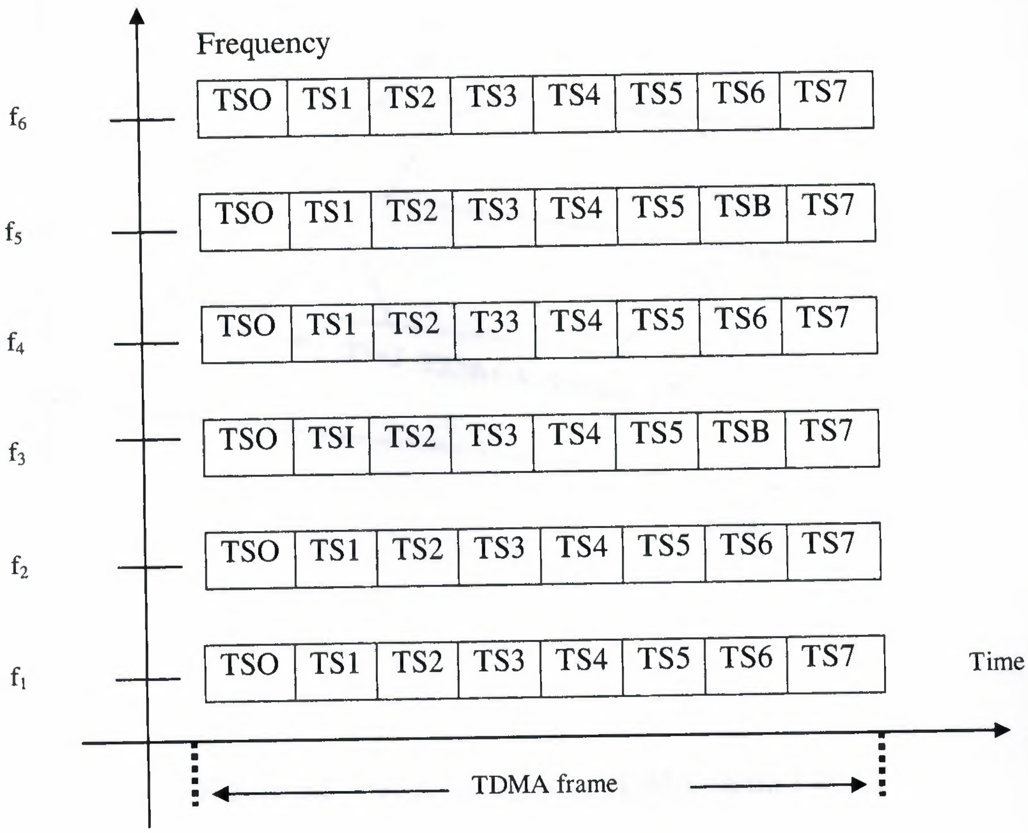


Figure 4.4 The FDMA/TDMA structure of GSM

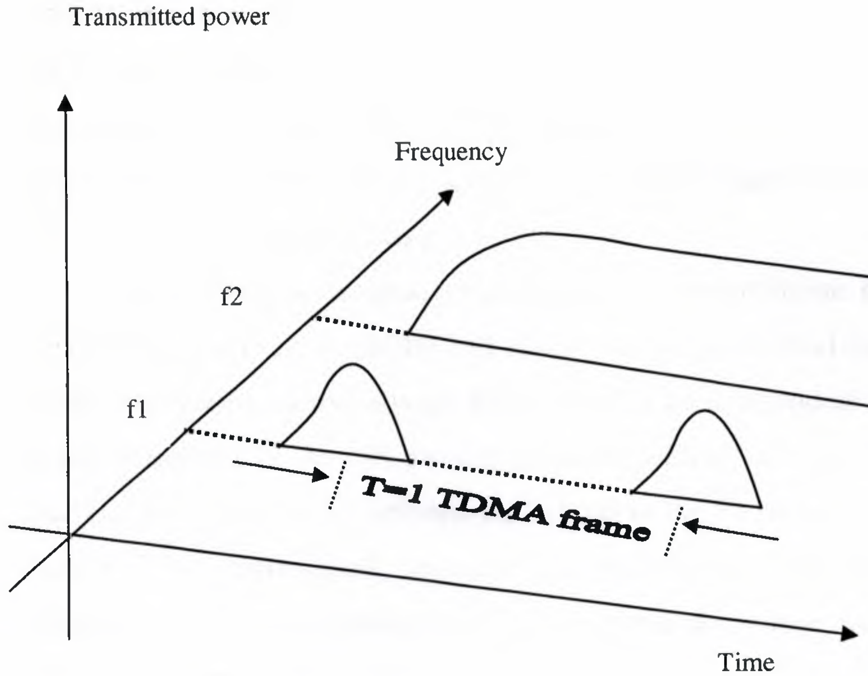


Figure 4.5 Spectral analysis of TDMA versus FDMA

4.3.1.2 Frame Hierarchy and Frame Numbers

In GSM, every impulse on frequency 1, as shown in Figure3.2, is called a burst. Therefore, every burst shown in Figure 3.2 corresponds to a TS. Eight bursts or TSs, numbered from 0 through 7, form a TDMA frame.

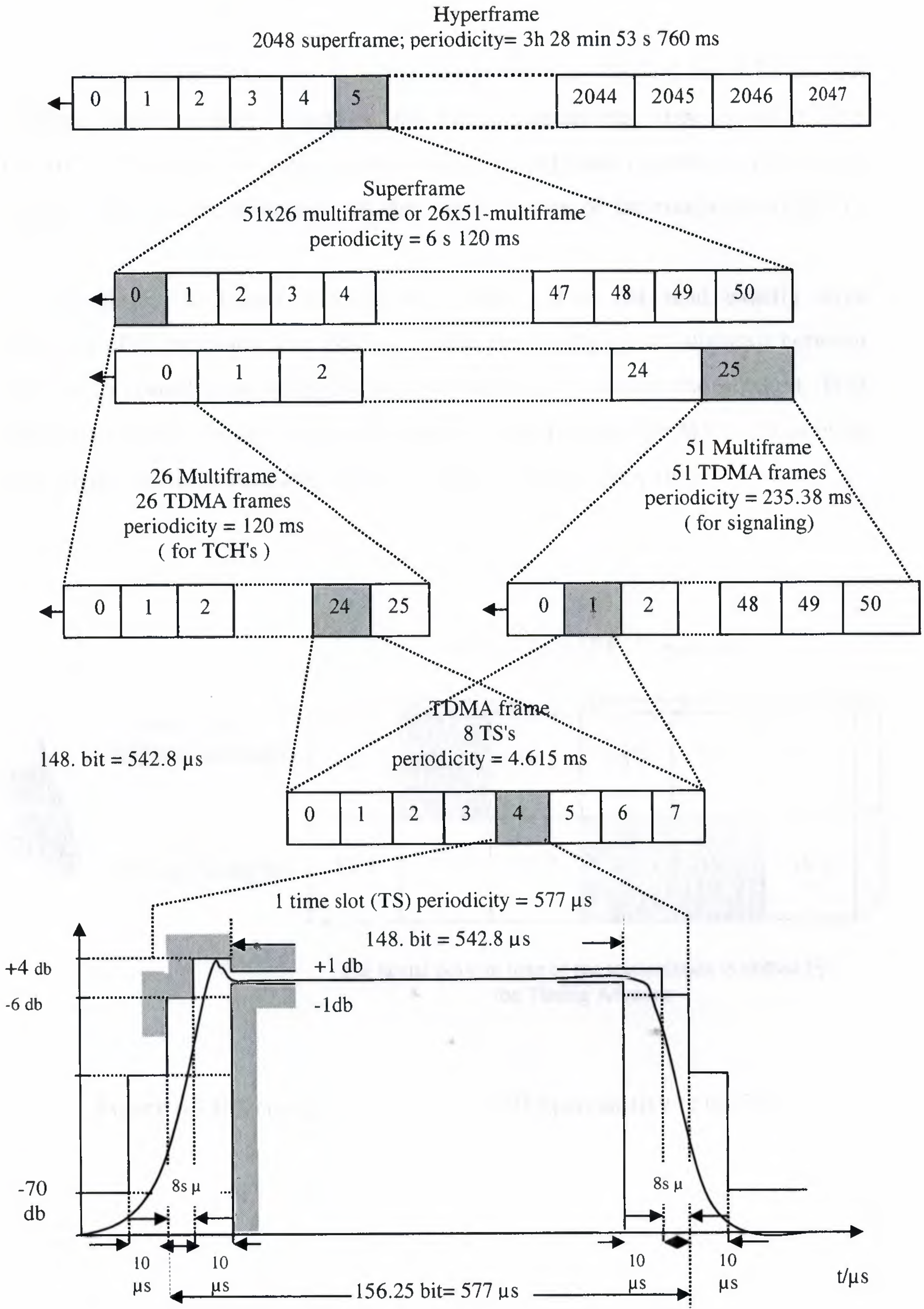
In a GSM system, every TDMA frame is assigned a fixed number, which repeats itself in a time period of 3 hours, 28 minutes, 53 seconds, and 760 milliseconds. This time period is referred to as hyperframe. Multiframe and super frame are layers of hierarchy that lie between the basic TDMA frame and the hyper frame. Figure 4.6 presents the various frame types, their periods, and other details, down to the level of a single burst as the smallest unit.

Two variants of multiframes, with different lengths, need to be distinguished. There is the 26-multiframe, which contains 26 TDMA frames with duration of 120 ms and which carries only traffic channels and the associated control channels. The other variant is the 51 -multiframe, which contains 51 TDMA frames with duration of 235.8 ms and which carries signaling data exclusively. Each super frame consists of twenty-six 51-multiframes or fifty-one 26-multiframes. This definition is purely arbitrary and does not reflect any physical constraint. The frame hierarchy is used for synchronization between BTS and MS, channel mapping, and ciphering.

Every BTS permanently broadcasts the current frame number over the synchronization channel (SCH) and thereby forms an internal clock of the BTS. There is no coordination between BTSs; all have an independent clock, except for synchronized BTSs. An MS can communicate with a BTS only after the MS has read the SCH data, which informs the MS about the frame number, which in turn indicates the chronologic sequence of the various control channels. That information is very important, particularly during the initial access to a BTS or during handover.

Consider this example: an MS sends a channel request to the BTS at a specific moment in time, let's say frame number Y (t = FN Y). The channel request is answered with a channel assignment, after being processed by the BTS and the BSC. The MS finds its own channel assignment among all the other ones, because the channel assignment refers back to frame number Y.

The MS and the BTS also need the frame number information for the ciphering process. The hyper frame with its long duration was only defined to support ciphering, since by means of the hyper frame, a frame number is repeated only about every three hours. That makes it more difficult for hackers to intercept a call.



-30 db

Figure 4.6 Hierarchy of frames in GSM

4.3.1.3 Synchronization between Uplink and Downlink

For technical reasons, it is necessary that the MS and the BTS do not transmit simultaneously. Therefore, the MS is transmitting three timeslots after the BTS. The time between sending and receiving data is used by the MS to perform various measurements on the signal quality of the receivable neighbor cells.

As shown in Figure 4.7, the MS actually does not send exactly three timeslots after receiving data from the BTS. Depending on the distance between the two, a considerable propagation delay needs to be taken into account. That propagation delay, known as timing advance (TA), requires the MS to transmit its data a little earlier as determined by the "three timeslots delay rule."

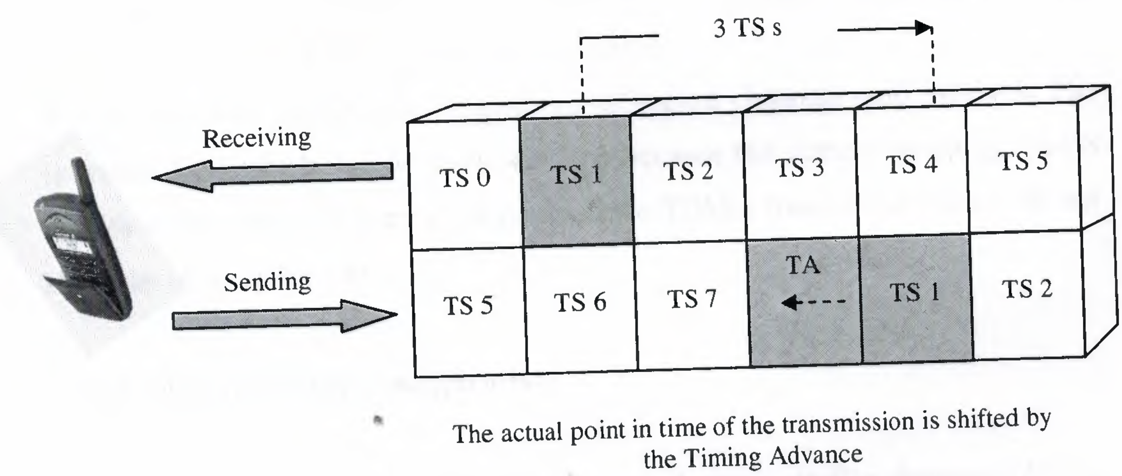


Figure 4.7 Receiving and sending from the perspective of the MS.

4.3.2 Physical versus Logical Channels

Because this text frequently uses the terms physical channel and logical channel, the reader should be aware of the differences between them.

- Physical channels are all the available TSs of a BTS, whereas every TS corresponds to a physical channel. Two types of channels need to be distinguished, the half rate channel and the full rate channel. For example, a BTS with 6 carriers, as shown in Figure 4.4, has 48 (8 times 6) physical channels (in full rate configuration).
- Logical channels are piggybacked on the physical channels. Logical channels are, so to speak, laid over the grid of physical channels. Each logical channel performs a specific task.

Another aspect is important for the understanding of logical channels: during a call, the MS sends its signal periodically, always in a TDMA frame at the same burst position and on the same TS to the BTS (e.g., always in TS number3). The same applies for the BTS in the reverse direction.

It is important to understand the mapping of logical channels onto available TSs (physical TSs)-which will be discussed later-because the channel mapping always applies to the same TS number of consecutive TDMA frame (The figures do not show the other seven TSs.).

4.3.3 Logical-Channel Configuration

Firstly, the distinction should be made between traffic channels (TCHs) and control channels (CCHs). Distinguishing among the different TCHs is rather simple, since it only involves the various bearer services. Distinguishing among the various CCHs necessary to meet the numerous signaling needs in different situations, however, is more complex. Table 4.1 summarizes the CCH types. Note that, with three exceptions, the channels are defined for either downlink or uplink only.

Table 4.1 Signaling Channels of the Air-Interface

Name	Abbreviation	Task
Frequency channel (DL) Correction.	FCCH	The 'lighthouse" of a BTS. PLMN/base station identifier of a BTS plus
Synchronization Channel (DL).	SCH	synchronization information (frame number).
Broadcast common control channel (DL).	BCCH	To transmit system information 1-4, 7-8 (differs in GSM, DCS1800, and PCS1900).
Access grant channel (DL).	AGCH	SDCCH channel assignment (the ASCH carries IMM_ASS_CMD).
Paging channel (DL).	PCH	Carries the FAG_REQ message.
Cell broadcast channel (DL).	CBCH	Transmits cell broadcast messages.
Standalone dedicated control channel.	SDCCH	Exchange of signaling information between MS and BTS when no TCH is active.
Slow associated control channel.	SACCH	Transmission of signaling data during a connection (one SACCH TS every 120 ms).
Fast associated control Channel.	FACCH	Transmission of signaling data during a connection (used only if necessary).
Random access channel (UL).	RACH	Communication request from MS to BTS

Note: DL = downlink direction only; UL = uplink direction only.

4.3.3.1 Mapping of Logical Channels Onto Physical Channels

In particular, the downlink direction of TS 0 of the BCCH-TRX is used by various channels. The following channel structure can be found on TS 0 of a BCCH-TRX, depending on the actual configuration:

- FCCH;
- SCH;
- BCCH information 1-4;
- Four SDCCH sub channels (optional);
- CBCH (optional).

This multiple use is possible because the logical channels can time-share TS 0 by using different TDMA frames. A remarkable consequence of the approach is that, for example, the FCCH or the SCH of a BTS is not broadcast permanently but is there only from time to time. Time sharing of the same TS is not limited to FCCH and SCH but is widely used. Such an approach naturally results in a lower transmission capacity, which is still sufficient to convey all necessary signaling data. Furthermore, it is possible to combine up to four physical channels in consecutive TDMA frames to a block, so that is possible for the same SDCCH to use the same physical channel in four consecutive TDMA frames, as illustrated in Figure 4.8. On the other hand, an SDCCH sub channel has to wait for a complete 5 1-multiframe before it can be used again.

That clarifies another reason for the frame hierarchy of GSM. The structure of the 5 1-multiframe defines at which moment in time a particular control channel (logical channel) can use a physical channel (it applies similarly to the 26-multifrarme)

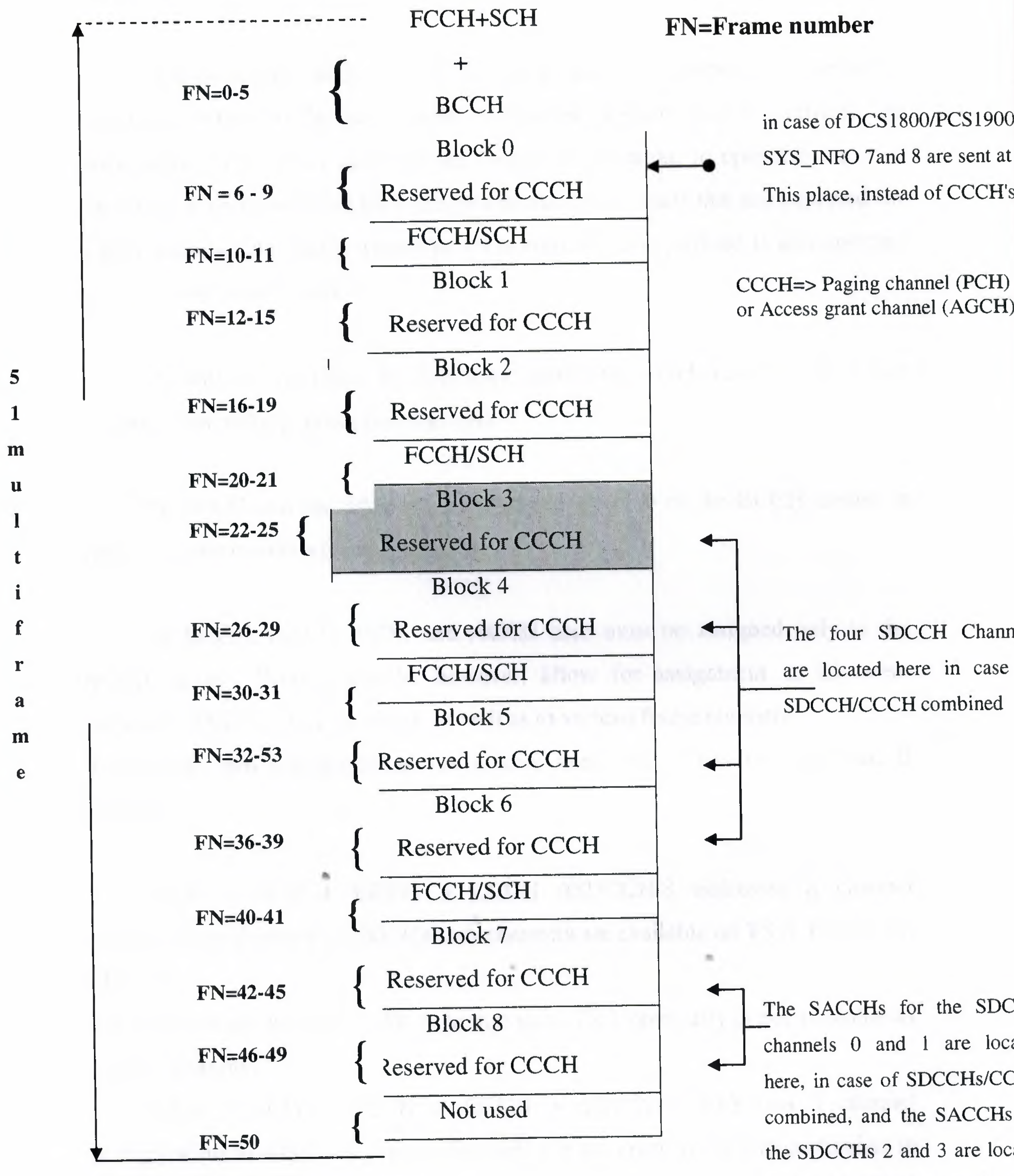


Figure 4.8 Example of the mapping of logical channels.

4.3.3.2 Possible Combinations

The freedom to define a channel configuration is restricted by a number of constraints when configuring a cell; a network operator has to consider the peculiarities of a service area and the frequency situation, to optimize the configuration. Experience with the average and maximum loads that are expected for a BTS and how the load is shared between signaling and payload is an important factor for such consideration.

GSM05.02 provides the following guidelines which need to taken into account when setting up control channels.

- The FCCH and the SCH are always sent in TS 0 of the BCCH carrier at specific frame numbers (see Figure 4.8).

- The BCCH, RACH, PCH, and AGCH also must be assigned only to the BCCH carrier. These channels, however, allow for assignment to all even-numbered TSs, e.g., 0, 2, 4, and 6, as well as to various frame numbers.

In practice, two configurations are mainly used, which can be combined if necessary:

- FCCH + SCH + BCCH + CCCH //SDCCH/8 addresses a channel configuration in which no SDCCH sub channels are available on TS 0. Eight such SDCCH
Sub channels are defined on TS 1. In that case, TS 1 obviously is not available as a traffic channel.
- FCCH + SCH + BCCH + CCCH + SDCCH/4 addresses a channel configuration in which all control channels are assigned to TS 0,in particular, to have TS 1 available to carry payload traffic. Because TS 0 needs to be used by the other control
Channels, too, it is possible to establish only four SDCCH subchannels, that is, only half the number compared to the preceding configuration.

A channel configuration is always related to single TS and not to single TS and not to a complete TRX. It is not possible to combine traffic channels and SDCCHs. If necessary
a TS can be "sacrificed" to allow for additional SDCCHs.

4.3.4 Signaling on the Air-Interface

4.3.4.1 Layer 2 LAPDm Signaling

The only GSM-specific signaling of OSI Layers 1 and 2 can be found on the Air-interface, where LAPDm signaling is used. The other interfaces of GSM use already defined protocols, like LAPD and SS7.
The abbreviation LAPDm suggests that it refers to a protocol closely related to LAPD, which is correct. The "m" stands for "modified" and the frame structure already shows the closeness to IAPD.

The modified version of LAPD is an optimized version for the GSM Air-interface and was particularly tailored to deal with the limited resources and the peculiarities of the radio link. All dispensable parts of the LAPD frame were removed to save resources. The LAPDm frame, in particular, lacks the TEl, the FCS, and the flags at both ends. The IAPDm frame does not need those parts, since their task is performed by other GSM processes. The task of the FCS, for instance, to a large extent,is performed by channel coding/decoding.

4.3.4.1.1 The Three Formats of the LAPDm Frame

Three different formats of identical length (23 bytes) are defined; their respective uses depend on the type of information to be transferred.

- A-format. A frame in the A-format generally can be sent on any DCCH in directions, uplink and downlink. The A-format frame is sent as a fill frame when no payload is available on an active connection, for example, in the short time period immediately after traffic channel is connected.
- B-format. The B-format is used on the Air-interface to transport the actual signaling data; hence, every DCCH and every ACCH use this format. The

maximum length of the Layer 3 information to be carried is restricted, depending on the channel type (SDCCH, FACCH.SACCH). This value is defined per channel type by the constant N201. If the information to be transmitted requires less space, this space has to be filled with fill-in octets.

- Bbis-format. For transmission of BCCH, PCH, and AGCH. There is no header in the Bbis-format that would allow for addressing or frame identification. Addressing is not necessary, since BCCH, PCH and AGCH are CCCHs, in which addressing is not required. In contrast to the DCCH, the CCCH transports only point-to multipoint messages Both frame types, the A-format and the B-format, are used in both directions uplink and downlink. The Bbis format is required for the downlink only. Also noteworthy is the relationship for signaling information between the maximum frame length of a LAPDm frame (= 23 byte 184 bit) and the number of input bits for channel coding (= 184 bit).

4.3.4.1.2 The Header of a LAPDm Frame

The Address Field

The address field starts with the bits EA and C/R, which perform the same tasks as the parameters with the same names in an LAPD frame. The same applies for SAPI, which takes on different values over the Air-interface than on the Abis-interface. Table2.2 lists the possible values for SAPIs on the air-interface and their uses. SAPI = 0 is used for all messages that deal with CC, MM, and RR, while SAPI= 3 is used for messages related to supplementary services and the SMS.
Furthermore, the address field of an LAPDm frame contains the 2-bit long link protocol discriminator (LPD), which in GSM is, with one exception, always coded with 00bin.
The exception is the cell broadcast service (CBS), where LPD = 0lbin.

Control Field

The control field of an LAPDm frame is identical to that of an LAPD frame modulo 8. It defines the frame type and contains, in the case of Iframes,the counters for N(S) and N(R); in the case of supervisory frames,it contains only N(R).

The frame length indicator field consists of three parts:

- Bit 0, the EL-bit. The EL-bit indicates if the current octet is the last one of the frame length indicator field. When this bit is set to 1,then another length indication octet follows, if set to 0, this octet is the last one. GSM does not allow the frame length indicator field to exceed one octet, and hence, the value of the EL-bit is always zero. GSM may change this restriction, if future applications require a different length.

- Bit 1, the M-bit. If entire messages are longer than the data field of the LAPDm frames allows, the information has to be partitioned and transmitted in consecutive frames. The M-bit is used in such a situation to indicate that the message was segmented and that further frames belonging to the same messages have to be expected. The M-bit of the last segment is set to zero, as illustrated in Figure 4.10.

- Bits 2-7, the length indicator. This field indicates the actual length of the information field. The value range is from zero to N201

Information Field

For all three frame formats, the information field that carries signaling data consists of N2Ol octets, where N201 represents a value that is different for the various channel types. How many of the octets—in the case of a B-format—are actually part of Layer 3 depends on the data to be transported. It is important to note that all unused octets in case of the B-format and all octets of the A-format are so-called fill-in octets, which are coded in a precisely defined pattern. This bit pattern is different for uplink and downlink. If, for example, an SDCCH frame contains only 18 bytes of data, the remaining two bytes are occupied with fill-in octets (note that N201 for the SDCCH has a value of 20).

Table 4.2 Possible Values of SAPI on the Air-Interface

SAPI (Decimal)	**Meaning**
0	RR, MM, CC
3	SMS,SS

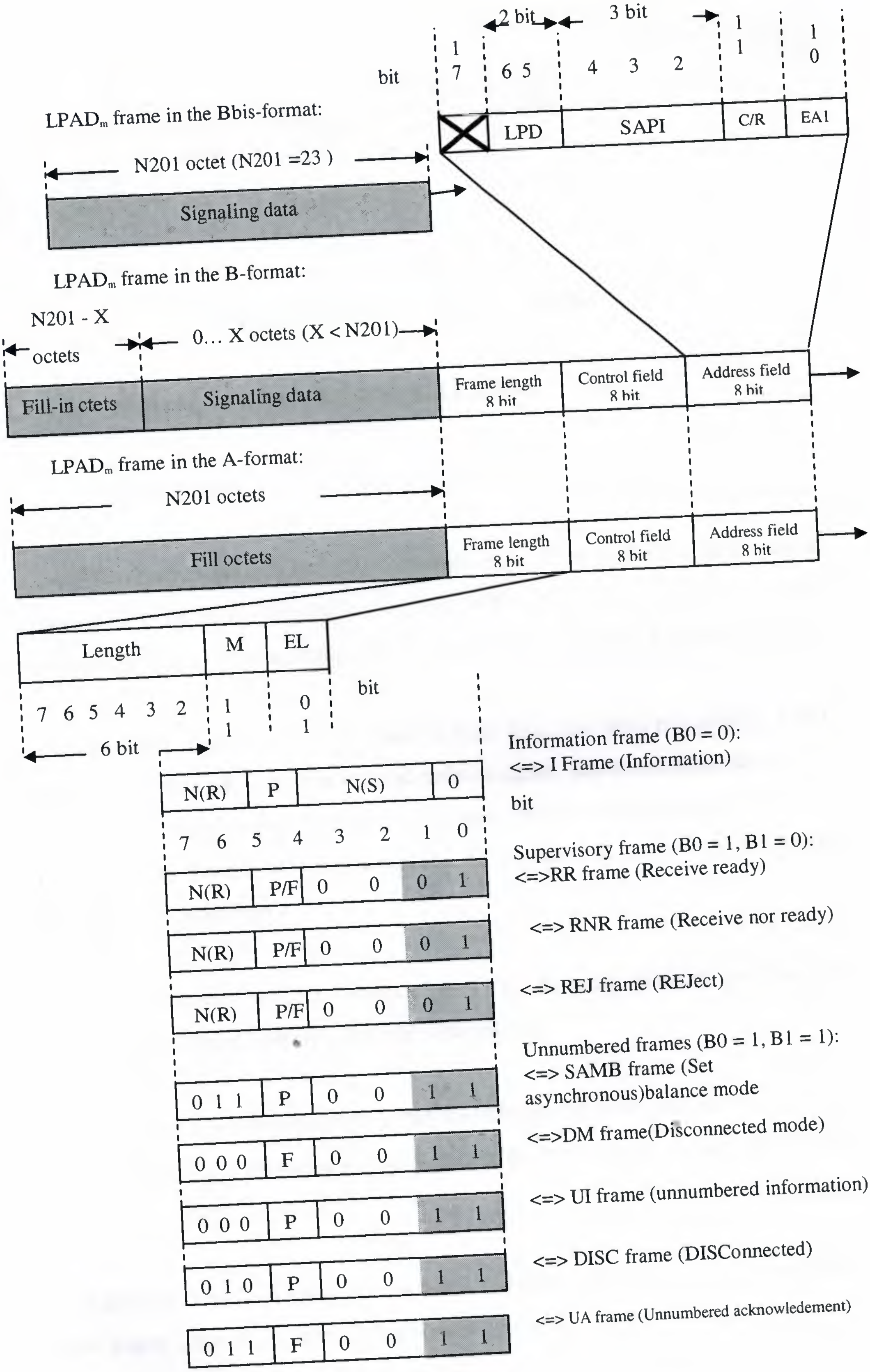


Figure 4.9 Frame format and frame type of LAPDm

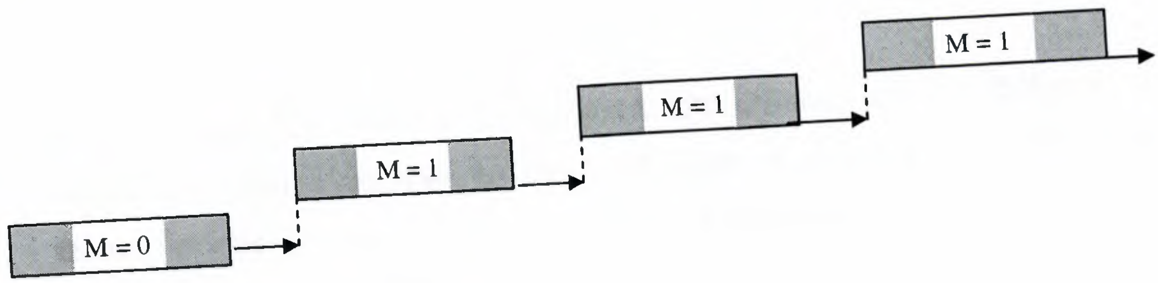


Figure 4.10 Segmentation in LAPDm

4.3.4.1.3 Differences Between LAPD and LAPDm

The differences between IAPD and LAPDm are as follows:

- IAPDm frames exist in modulo 8 formats only. Their control field, therefore, is always 1 octet long. The N(S) and the N(R) are in the range 0 to 7. That theoretically restricts the maximum number of unacknowledged I frames to seven.

- The address field of IAPDm, is only 1 octet long and does not contain a TEl. The reason is that when a channel is already assigned, the connection on the Air-interface is always a point-to-point connection. Several simultaneous users, for example, on a terrestrial point-to-multipoint connection, do not exist, which makes the TEl superfluous.

- IAPDm frames do not contain an FCS, because channel coding and interleaving of Layer 1 already provide data security.

- LAPDm frames do not have a flag to indicate the start and end of a frame. That functionality is provided on the Air-interface by Layer 1, in particular by the burst segmentation.

- Unlike in LAPD, SABM frames and UA frames of LAPDm, may even carry Layer 3 data. That saves time during connection setup.

- The maximum lengths of LAPD and LAPDm frames are very different. While IAPD frames can transport up to 260 octets of signaling data, LAPDm allows for only 23 octets. If a larger amount of data needs to be transported, segmentation has to be applied.

- LAPDm frames do not contain a length indicator (Layer 2).

- In LAPD, no fill-in octets are used when the data area is not completely occupied with signaling data.

4.3.4.1.4 Frame Types of LAPDm

Fewer frame types are defined for the IAPDm protocol than for LAPD. The XID frame and the FRMR frame are missing in LAPDm. Both frames are used for specific tasks and are not necessary in LAPDm. Table 3.3 lists the frame types of LAPDm and their specific uses. As for IAPD, it is distinguished whether a frame is used to carry a command, a response, or both. LAPD follows the definition of LAPD, that is, the P/F bit and the C/R bits are used the same way for both protocols.

Table 4.3 Frame Types of the Air-Interface

Name	Command Frame?	Answer Frame?	Possible Values of Control Field (Hex)
1-frame group:			
I	Yes	No	(OX), (2X),(4X),(6X),(8X) if even, then I frame
Supervisory-frame group			
RR RNR REJ	Yes Yes Yes	Yes Yes Yes	(1X) (5X) (9X)
Unnumbered-frame group			
DISC UI DM SABM UA	Yes Yes Yes Yes Yes	No No Yes No Yes	(53) because P bit is always 1 (03) because P bit always 0 (0F), (1F) (7F) because P bit always 1 (73) because F bit always 1

4.3.4.2 Layer 3

Figure 4.11 illustrates the Layer 3 format on the Air-interface.

4.3.4.2.1 Protocol Discriminator

The 4-bit-long protocol discriminator (PD) is used on the Air-interface to classify all messages into groups and allows, within Layer 3, the addressing of various users, just as the message discriminator does on the Abis-interface. Every message is no ambiguously assigned to a PD or service class. A distinction between transparent and nontransparent services is possible at the same time. Supplementary services and the SMS are special, because they do not belong to CC but are still sent with the same PD. Table 4.4 lists all PDs and their service classes.

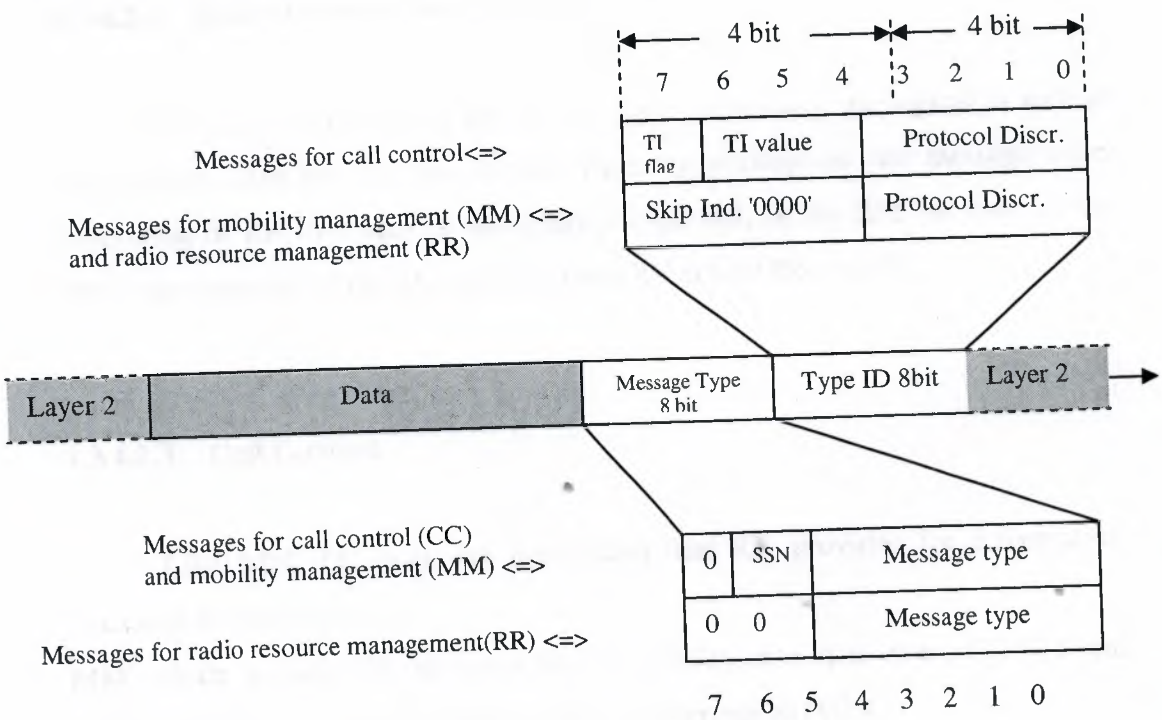


Figure 4.11 The Layer 3 format on the Air-interface.

Table 4.4 Protocol Discriminators on the Air-Interface

PD	Service Class
06 05 03	RB (radio resource management) MM(mobility-management) CC(call control) SS(supplementary services) SMS(short-massage service)

4.3.4.2.2 Radio Resource Management

Messages in the area of RR are necessary to manage the logical as well as the physical channels on the Air-interface. Depending on the message type, processing of RR messages is performed by the MS, in the BSS, or even in the MSC. Involvement of the BSS distinguishes RR from MM and CC.

4.3.4.2.3 Call Control

Like MM, CC uses the connection that RR provides for information exchange In contrast to
MM, which is used only to maintain the mobility of a subscriber, CC is a real application that at the same time provides an interface to ISDN.

4.4 Summery

In this chapter, we have seen the GSM radio interface and we have discussed all the working principles of every part contains in it, and we have discussed also the Air-interface with every part it contains such as structure of the Air-interface, the FDMA/TDMA scheme, Physical versus Logical Channels, Logical-Channel Configuration, Signaling on the Air-Interface with details.

CONCLUSION

GSM is the global system for mobile communication, GSM utilizes a cellular structure, the basic idea of a cellular network is to partition the available frequency range, to assign only parts of that frequency spectrum to any base transceiver station, and to reduce the range of a base station in order to reuse the scarce frequencies as often as possible. One of the major goals of network planning is to reduce interference between different base stations.

Besides the advantage of reusing frequencies, a cellular network also: comes with the following disadvantages:

- An increasing number of base stations increase the cost of infrastructure and access lines.
- All cellular networks require that, as the mobile station moves, an active call is handed over from one cell to another, a process known as handover.
- The network has to be kept informed of the approximate location of the mobile station, even without a call in progress, to be able to deliver an incoming call to that mobile station.
- The second and third items require extensive communication between the mobile station and the network, as well as between the various network elements. That communication is referred to as signaling and goes far beyond the extent of signaling that fixed networks use. The extension of communications requires a cellular network to be of modular or hierarchical structure. A single central computer could not process the amount of information involved.

Call process have two main processes first it will deal with transmission component:

- At first it will be converted from analog to digital signal and the outputs will be a collection of bits: binary ones and zeros, that by using a process called Pulse Code Modulation (PCM). PCM involves the three main steps which are sampling, quantization, and coding.
- This output will deal with speech coding; the goal of all speech coding systems is to transmit speech with the highest possible quality using the

least possible channel capacity, Speech coders classified into two categories:

1. Waveform coders and
2. Vocoders.

- Channel coding refers to the class of signal information's designed to improve communications performance by enabling the transmitted signal to better withstand the effects of various channel impairment, such as noise, interference.
- Interleaving is meant to de-correlate the relative positions of the bits respectively in the code words and in the modulated radio bursts. The aim of the interleaving algorithm is to avoid the risk of loosing consecutive data bits.
- Burst format GSM uses both FDMA and TDMA for multiple accesses. GSM900 uses the 890 MHz to 960 MHz frequency band for communications (Uplink 890-915 MHz, Downlink 935-960 MHz). GSM900 has two frequency bands 45 MHz. The 890 MHz to 915 MHz frequency band is used for uplink from a MS to a BS and 935 MHz to 960 MHz frequency band is used for the downlink from a BS to a MS. Each of the two bands is subdivided into 124 single carrier channels of 200 KHz each. A guard band of 200 KHz is left on both sides of uplink and the downlink bands
- Ciphering, the purpose of ciphering is to encode the burst so that it cannot be interpreted by any device other than the intended receiver. The ciphering algorithm in GSM is called A5 algorithm. It does not add bits to the burst
- Modulation GSM uses GMSK modulation, GMSK uses a Gaussian filter from which its name was derived. The frequency separation utilized is the minimum frequency separation required for the two modulating signals to be orthogonal over a signaling interval of length T. the advantages of GMSK modulation is:

1. It does not produce any Intersymbol Interference (ISI).

2. GMSK modulation is continuous phase modulation, which achieves a good compromise between power and bandwidth efficiency while maintaining a specified level of BER at the expense of only reasonable increase of system complexity.

After modulation the process will continue but with reception component, it will start with demodulation, the Demodulation used to recover the baseband signal that was modulated using by GMSK modulation. We used the GMSK Demodulator Baseband.

Then it will deal with deciphering and then with burst deformat, it is the reverse operation of Burst Format it is used to remove the additional bits which added in burst format (32 bits) for each frame, the process will continue with deinterliveing, channel decoding and speech decoding and it's all opposite to the transmission process

The radio interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces of the GSM system. The spectrum efficiency depends on the radio interface and the transmission, more particularly in aspects such as the capacity of the system and the techniques used in order to decrease the interference and to improve the frequency reuse scheme. The specification of the radio interface has then an important influence on the spectrum efficiency.

The Air-interface is the central interface of every mobile system and typically the only one to which a customer is exposed.

The physical characteristics of the Air-interface are particularly important for the quality and success of a new mobile standard. For some mobile systems, only the Air-interface was specified in the beginning, like IS-95, the standard for CDMA. Although different for GSM, the Air-interface still has received special attention. Considering the small niches of available frequency spectrum for new services

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