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BASE STATIONS AND AIR INTERFACE

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DEDICATIONS

Dedicated to my parents

ABSTRACT

GSM, the Global System for Mobile communications, is a digital cellular communications system, which has rapidly gained acceptance and market share world wide, although it was initially developed in a European context. In addition to digital transmission, GSM incorporates many advanced service and features, Including ISDN compatibility and worldwide roaming in other GSM networks. The advanced services and architecture of GSM have made it a model for future third-generation cellular systems, such as UMTS.These papers cover the material about the GSM internal architecture such as mobile stations and base stations. Also some information about the air interface between them.

1. INTRODUCTION TO WIRELESS COMMUNICATION

This century has seen great development in communication technology. We have a sound global telephone network, which is both affordable and reliable. We have powerful cable television networks in every urban area, which could provide a relatively broadband width. And we have the dominating Internet, which is able to provide various rate of service. And there are also many research and industry going on all over this world about these fueling wire lined networks.

But why do we need wireless communication systems? The answer is simple: there do exit a group of people who need to be on line all of the time. Business people wish to be able to have their meeting at any time and in any place; stock dealers wish to know what is going on in the stock exchange all the time and to respond as soon as possible; networms tend to access their email boxes every 10 minutes, the same time that they are chatting with other networms. That explains why we need wireless communication system, and it is becoming into the most profitable and appealing area in computer network.

1.1 Solutions To Wireless Communication Network

The fact is there are already lots of wireless communication network running currently. Generally speaking, they could roughly be classified into three types: cordless telephony, cellular system, and satellite communication. The first solution is cordless telephony. The idea in cordless telephony is straightforward. Since we could exploit electromagnetic wave for radio broadcasting, which is a simplex communication mode, why not use it for duplex communication, such as dialog transmitting? Sure we can and so we did. Cordless communication in a relative large area used to be running in some places, but it turned out to be not so practical, since in order to reach a larger area. Let's say 100km in radius, we need to do and suffer much. We have to build high towers to place the antenna, because of both the shape of the earth surface and barriers on the ground; we have to apply great power to transmission so that signals would not be unrecognizable when they reach the receivers; handsets would have to use a huge battery for receiving and transmitting signals (it is really funny to carry a battery 10 times larger than the handset itself); other wireless communication, such as communication at airports where precision and reliability is vital, would be interference badly; the worse thing is that the capacity of such a paradigm is low in that there is no way of frequency reusing. Therefore, wide area cordless communication was aborted because of its great drawbacks. Fortunately, it is not a totally useless invention. On the contrary, we have lots of small-area cordless products, like the cordless telephones that we are using in kitchen when chatting on the phone the same time as cooking. These cordless telephones have relatively small coverage's, usually 100 meters, and less radio interferences.

We have been making various sorts of uses of satellites since the former Soviet Union launched the first man-made satellite in 1957, and one of the most remarkable one is in communication. There are basically two classes of satellite communication, where the basic ideas differ a lot. One is Geo synchronous Earth Orbit (GEO) satellites, which is now very matured and quite of common sense, and the other is Low Earth Orbit (LEO), in which several consortia are quite involved. And unlike GEO, where we only need a geo stationery satellite that is always above us for communication purposes, in LEO we do not have predetermined satellite. Instead, we might use a certain satellite that is passing overhead. What is more, during the whole process of communication, the communication might be switched to another satellite if the previously used one has got far away. LEO is much more complicated than GEO, but the round trip delay is decreased greatly since GEO satellites have to be placed 36,000km above the equator, which is teens as much as GEO satellites. While round trip delay is one of the most important criteria for evaluating the performance of a communication system. Both GEO and LEO would thrive in future, and the only difference between them is that they are serving in communication with different requirements. There are several GEO paradigms going on, like Iridium, TeleDesic, Global Star.

Cellular systems are one of the most fashionable words in telecommunication and electronic industries. And it could be looked on as a deviation of cordless communication system, which no longer has the drawbacks of wide-area cordless communication. The basic idea in cellular systems is that a geographic area is divided up into small cells and frequencies could be reused in different cells as long as these cells are not neighboring with one another. Cellular communication mode is has many advantages, which cordless telephony used to be lack of, such as high capacity, low interference, less power needed, and etc. Actually satellite communication researchers exploited the idea from cellular, and that is how LEO became so popular in current communication research work.

1.2 Electromagnetic Spectrum and Its Uses For Communication

Some radical people think that the future holds only two kinds of communication: fiber and wireless, so before taking a close look at cellular communication systems themselves, let's take a quick glance at electromagnetic spectrum. British scientist James Clerk Maxwell predicted the behavior of electromagnetic wave in 1865 and then began our uses of such a natural resource from then on. From figure 1 we can see that we are making a pretty good use of such resource nowadays. The radio, microwave, infrared, and visible light portions of the spectrum can all be used for transmitting information by modulating the amplitude, frequency, phase, or a combination of them. Higher frequencies, like ultraviolet light, X-rays, and gamma rays would be even better, but they are hard to produce, control and modulate, and do not propagate well through buildings, since they behave more like particle rather than wave, and dangerous to living things.

1.3 Wireless LANs and WANs

Wireless Local Area Networks (LANs) and Wide Area Networks (WANs) provide the means for the mobile user to access information stored in remote peripherals or databases. Wireless LANs enable communications within a single building or campus. Wireless WANs fulfill the same function but have a greater reach, often spanning an entire continent, by interconnecting with a wide area service provider.

An extensive array of wireless LAN and WAN hardware and software underscores IBM's recognition of the importance of communications sans wires. Wireless LAN adapter cards use radio frequencies to communicate with each other, thus allowing the user to converse, retrieve or enter data without having his or her personal computer, notebook or pen-tablet physically attached to a wired LAN or telephone line. Wireless WANs communicate in a similar fashion but use wireless PCMCIA modems to connect across radio or cellular networks owned and operated by service providers. A look at LAN and WAN configurations and components

1.3.1. Wireless LANs

There are three types of wireless LAN configurations: base-to-remote, peer-to-peer or micro cellular. In base-to-remote, distant workstations and personal computers are linked to a central base workstation. Because the base workstation is located at the center of the LAN, excellent coverage, good range, high security levels and more management functions across the network are the result. For commercial uses such as those found within a single building or integrated set of buildings, base-to-remote works well either as a standalone wireless LAN or as an extension to a wired LAN. Peer-to-peer wireless LANs is self-defining since they permit direct communication between devices without going through a base station. As such these networks are quick to install and therefore well suited to ad hoc networking. The downside to peer-to-peer is that security and network management concerns are not easily addressed and the range of communications is limited. This latter concern can be resolved, however, by adding an access point product for attachment of the wireless peer-to-peer LAN to a wired LAN.

Micro cellular wireless LANs use access points on a wired backbone to permit client devices to communicate to any backbone-connected device or to any other wireless device connected to the backbone through the same or another access point. This configuration also permits seamless roaming from cell to cell when the coverage of those cells via the access points provides sufficient overlap. This configuration is analogous to the technique used in cellular telephones—now on a local area basis due to the restricted range of LAN radios.

1.3.2. Wireless WANs

The key components in wireless WANS are PCMCIA adapters that represent the latest in wireless communication. Currently, IBM offers systems with integrated WAN modems for Cellular/CDPD (Cellular Digital Packet Data), ARDIS (U.S. and Canada only) and Mobitex[1]. Each modem has a different business application.

CDPD is unique to the Advanced Mobile Phone Service (AMPS) cellular network, the largest in the United States. IBM's 2489-600 with the optional wireless modem for CDPD includes an internal PCMCIA radio modem and radio antenna. The use of this radio modem requires the purchase of CDPD services from a service provider. Advanced Radio Data Information Service (ARDIS) provides interactive, real-time[2] data communications throughout the U.S. and Canada. The IBM 2489-600 with integrated Wireless Modem for ARDIS supports automatic nationwide "roaming," which means users can move seamlessly from one city to another and still communicate. The use of this radio modem requires the purchase of ARDIS services from a service provider. Mobitex runs on the MOBITEX network that serves some European countries and about 8,000 cities across the United States with fax, e-mail, two-way messaging and server applications. The IBM 2489-600 with integrated Wireless Modem for Mobitex consists of an integrated PCMCIA adapter[1] with an integrated antenna. The use of this radio modem requires the purchase of MOBITEX services from a service provider.

Due to distinct country differences in communications standards, it is currently impossible to say one network provides wireless WAN services globally. In most cases, analog data is transmitted using a cellular-enabled modem with a hand held phone. GSM/DCS 1800 data wireless networks are further made up of GSM, the digital equivalent of AMPS, and DCS 1800, an 1800MHz system with similar protocols to GSM and a data adapter. CT2 (Cellular Telephone) is a short-range campus and public network. It requires an integrated adapter/transceiver connected to a local base station for "campus" work that is connected to a PSTN for WAN communications.

Public safety agencies (police, fire, rescue) can use wireless WANs for immediate access to state and national databases. Real estate agencies find that WANs bring timely information to realtors "in the field" through updates to the online version of the Multiple Listings Book. The book was formerly updated once every two weeks, it now can be updated as soon as there is a change to a listing. Another WAN application is in the mobile nursing field. Now, using a wireless WAN, nurses at patients' homes can enter current status and last treatment information for immediate update in the healthcare database. The risk of error is greatly reduced and time is saved (locating and pulling charts, etc.) because critical information is only a few keystrokes away.

What Wireless can mean to our business

1. Support for short- and long-term requirements without cabling

2.Support for mobile and roaming users in a campus or across a continent

3.Reduction or elimination of cabling costs

4.Provide the ability to keep employees informed and responsive to customer needs anywhere anytime

5.Save time and money by simplifying processes

1.4 What is personal communications?

The PCS buzzword is not new in the worldwide wireless industry. The PCS term, in particular, is used and misused in North America. Since 1995, the notion of personal communications was coupled with the words systems and services. In fact, we should go back LO the end of the 1980s, when the first personal communications concepts were discussed in the United Kingdom, still one of the most competitive cellular markets in the world. Can we define what personal communications stands for? What communications system offer that other systems do not? When we gather together the different definitions, advertisements, explanations, and technology offerings of the mobile marketplace, we find some common problems, requirements, design targets, and technical solutions.

Changes in lift-style and work habits have increased the mobility of people in recent years. People in industrialized countries are forced 10 increase their productivity by managing more information relative to things. Remote working and longer work hours have blurred the border between personal time and work time in the information society. The blending of work and home is enabled with supplements to plain old voice telephony.

Now, many workers routinely add video, fax, and data communications to their voice communications tools. The process feeds on itself. Additional low-cost, non-voice communications services make it easier for employers to expect more performance out of their busy employees, and for customers to expect even faster service from their suppliers. The communicating society finds itself using more and more value-added services that did not exist a few years ago. Point-to-multipoint information systems such as cell broadcast, short messaging, and supplementary (personalized) paging services seem ideal for alerting people to sports scores, business events, weather, and stock price changes. Packet data services, in addition to conventional point-to-point telecommunications, add interactive features. The convergence of communications equipment and computers continues to create new products and new services for mobile users. The challenge is to make these products reliable and simple to use, but affordable. The cycle is an enticing trap that guarantees further innovation, for the very tools and services that force people to carry their work around with them are the same things that seem to offer a way out of the trap. Look at how things have escalated. Fifteen years ago an airmail letter was fast enough for most needs. Today, a fax is seldom fast enough. The more features and gadgets people can use with their phones, the greater will be the number of people who feel they cannot function without at least one of them. The picture is not yet clear. We have listed some features and services people often associate with PCS and other personal communications terms, but we still don't know what it is. Let's try a more orderly approach

1.5 Defining the requirements

Boiling the requirements down to a few visionary buzzwords would mean to say: receive, revise, and originate calls or messages...

With one small terminal;

Everywhere (in the world);

Under one number;

In any form;

At any time.

Communications and related services become personalized, customized, and location independent in PCS. We connect to people, not places. We dial one number to reach one person, regardless of where this person is. The communicator, and all the things the user does with it, becomes part of the person. The person and the output of her labor become ubiquitous, in both time and space, to a growing number of people. What processes and services make this happen?

1.6 Cost of ownership

Subscriptions have to be affordable to a wide variety of mass users. We are talking about individual and personal services for the mass markets that compete with, and to some degree replace, the wired phone services. To be successful, even in developing countries, such services must be priced at levels matching the wealth structures of the customers. Service revenues per subscriber are shrinking, because the majority of new entrants to wireless network services are private users and not business users. Western Europe will see a drop in annual revenues per subscriber of approximately 30% from 1995 to the year 2000. Similar decreases in the range of 20% to 40% are to be expected in all other regions worldwide.

1.7 Access, mobility, connectivity, and services

Access to both wireless and fixed networks will be supported with mobility. The user wants to be able to originate and receive calls that are routed through public switches, to and from practically anywhere in the world, without any restrictions. Furthermore, in order to be reachable all the time and at any location (with the option of some private imitations), some kind of sophisticated mobility management has to be employed. This should be transparent to the user without the need to call in with a roaming code for the area in which the user is currently located. Users do not want to participate directly in the mobility management processes. Services and applications have to be state of the art and scalable to the user. Bearer services should allow for high data rates, up to 144 Kbps (basic rate ISDN) and fast packet oriented delivery, say, for fast Internet access. The portfolio is completed with voice mailboxes, messaging, and alerting—each of which has to function well when users do not need to be reached or prefer not to be disturbed. Combinations of business and private use, overlaid on even more combinations of office, residential, and wide-area use, challenge the system and its services. One terminal, one number (or explicitly two numbers: one for incoming private and another for business calls) will be possible. Different billing and even different radio access may be provided for different applications of the same communicator. The concept is to have a single terminal for each user. The user can be sitting still at home or walking in the street, at work or at leisure, alone in the office or with a client in a meeting. The user does not have the time or the skill to assist in mobility management tasks or to deal with the details of his private branch exchange (PBX), roaming, and handover.

1.8 Coverage and capacity

Network coverage and capacity must be able to cope with a high user density, a broad mix of traffic, and sharp spikes in peak usage. Peaks will occur during daytime business use and evening private use. In conventional cellular networks operators cleverly exploited this peak business capacity by marketing the off-peak periods to the private user at "moonshine" tariffs. Adequate coverage to the private user means that she can use her phone anywhere in a given area she may find herself most of the time say, 90% of her time. The service should be available everywhere and anytime: at home, on the road, on the street, in the office, outside, or inside. Roaming agreements between operators, or the fact that a particular operator's service is nationwide, enhance the chances of being granted access to service all the time.

1.9 Voice quality

Voice quality has to be high. The trend is toward fixed-line toll quality without any compromises that are characteristic of first- and second-generation speech trans coder technologies. As a benchmark, the CCITT/ITU G721/726 adaptive differential pulse code modulation (ADPCM) algorithm delivers adequate quality, but at a relatively high rate of 32 Kbps. An abundance of speech coding algorithms is available that have been designed for use in communication systems with a requirement for low-bandwidth (baud rate) voice data, such as spectrally efficient wireless transmission systems, with few sacrifices in voice quality. Voice quality is determined by system design. Mobile systems dedicated to voice traffic can tolerate relatively high error rates on the channel relative to the lower error rates demanded of data services. Operators have to perform yet another balancing act as they configure their network densities in accordance with the mix of voice and data services their business plans require. The "communicators" Phones have to be conveniently small, lightweight, secure, and easy to use. They must guarantee long standby times in excess of 1 week and talk times of several hours on each battery charge. There are practical and technical limits to these requirements. If one has to move an ultra small phone from ear to mouth in order to listen and speak in a quasi-half-duplex mode, a tangible limit of size have been exceeded, which, in this case, is "undercut". Phones cannot be made as small as the technology allows. We still have to be able to use the phone with comfort.

Small, low-power phones also need additional support from the network architecture. The network's cell structure must be built densely enough to accommodate low RF power transmissions from the phone at the required grade of service. New data features that are supported by modem ports and larger displays need to be reliable and obvious in their use. Finally, security features, such as user authentication and encryption, must be supported without the hassle and confusion that usually accompany security measures.

1.10 The technical solutions to the requirements

Under the defined PCS circumstances, the basic demands are sufficiently met by a migration of current digital cellular technology into a technology that can handle higher capacity yet still allows for an upgrade path for services and applications. Existing digital technology has to be modified to accommodate more capacity while also being able to support additional services and applications. Cordless telephone-based technologies might be enough for some applications. DECT or PHP technology may work in dense office environments. Because nothing solved all the problems at the time, PCSs were initially regarded as "next" or "third"-generation systems. Apparently, the feature-rich second-generation digital cellular systems such as GSM are well suited for use in systems characterized as PCSs, but there is a lot of room for improvement in compatibility, worldwide coverage, cost, effectiveness against user expectations, and service features. Such improvements are currently tackled in projects defining and designing the next generation.

2. HISTORY OF THE CELLULAR MOBILE RADIO AND GSM

The idea of cell-based mobile radio systems appeared at Bell Laboratories (in USA) in the early 1970s. However, mobile cellular systems were not introduced for commercial use until the 1980s. During the early 1980s, analog cellular telephone systems experienced a very rapid growth in Europe, particularly in Scandinavia and the United Kingdom. Today cellular systems still represent one of the fastest growing telecommunications systems. But in the beginnings of cellular systems, each country developed its own system, which was an undesirable situation for the following reasons: The equipment was limited to operate only within the boundaries of each country. The market for each mobile equipment was limited.

In order to overcome these problems, the Conference of European Posts and Telecommunications (CEPT) formed, in 1982, the Groupe Spécial Mobile (GSM) in order to develop a pan-European mobile cellular radio system (the GSM acronym became later the acronym for Global System for Mobile communications). The standardized system had to meet certain criterias:

1. Spectrum efficiency

2. International roaming

3. Low mobile and base stations costs

4. Good subjective voice quality

5. Compatibility with other systems such as ISDN (Integrated Services Digital Network)

6. Ability to support new services

Unlike the existing cellular systems, which were developed using an analog technology, the GSM system was developed using a digital technology. In 1989 the responsibility for the GSM specifications passed from the CEPT to the European Telecommunications Standards Institute (ETSI). The aim of the GSM specifications is to describe the functionality and the interface for each component of the system, and to provide guidance on the design of the system. These specifications will then standardize the system in order to guarantee the proper inter working between the different elements of the GSM system. In 1990, the phase I of the GSM specifications was published but the commercial use of GSM did not start until mid-1991.

The most important events in the development of the GSM system are presented in the table 2.1

Year	Events
1982	CEPT establishes a GSM group in order to develop the standards for a European cellular mobile system
1985	Adoption of a list of recommendations to be generated by the group
1986	Field tests were performed in order to test the different radio techniques for the air interface
1987	TDMA is chosen as access method (in fact, it will be used with FDMA) initial Memorandum of Understanding (MoU) signed by telecommun-cation operators (representing 12 countries)
1988	Validation of the GSM system
1989	The responsibility of the GSM specifications is passed to the ETSI
1990	Appearance of the phase 1 of the GSM specifications
1991	Commercial launch of the GSM service
1992	Enlargement of the countries that signed the GSM- MoU> Coverage of cities/airports
1993	Coverage of main roads GSM services start outside Europe
1995	Phase2 of the GSM specifications Coverage of rural areas

Table 2.1: Events in the development of GSM

From the evolution of GSM, it is clear that GSM is not anymore only a European standard. GSM networks are operational or planned in over 80 countries around the world. The rapid and increasing acceptance of the GSM system is illustrated with the following figures:

1. 1.3 million GSM subscribers worldwide in the beginning of 1994.
2. Over 5 million GSM subscribers worldwide in the beginning of 1995.
3. Over 10 million GSM subscribers only in Europe by December 1995.

Since the appearance of GSM, other digital mobile systems have been developed. The table 2.2 charts the different mobile cellular systems developed since the commercial launch of cellular systems.

Year	Mobile Cellular System
1981	Nordic Mobile Telephony (NMT), 450>
1983	American Mobile Phone System (AMPS)
1985	Total Access Communication System (TACS) Radiocom 2000 C-Netz
1986	Nordic Mobile Telephony (NMT), 900>
1991	Global System for Mobile communications> North American Digital Cell (NADC)
1992	Digital Cellular System (DCS) 1800
1994	Personal Digital Cellular (PDC) or Japanese Digital Cellular (JDC)
1995	Personal Communications Systems (PCS) 1900- Canada>
1996	PCS-United States of America>

Table 2.2: Mobile cellular systems

2.1. Cellular systems

2.1.1.The cellular structure

In a cellular system, the covering area of an operator is divided into cells. A cell corresponds to the covering area of one transmitter or a small collection of transmitters. The size of a cell is determined by the transmitter's power. The concept of cellular systems is the use of low power transmitters in order to enable the efficient reuse of the frequencies. In fact, if the transmitters used are very powerful, the frequencies cannot be reused for hundred of kilometers as they are limited to the covering area of the transmitter. The frequency band allocated to a cellular mobile radio system is distributed over a group of cells and this distribution is repeated in all the covering area of an operator. The whole number of radio channels available can then be used in each group of cells that form the covering area of an operator. Frequencies used in a cell will be reused several cells away. The distance between the cells using the same frequency must be sufficient to avoid interference. The frequency reuse will increase considerably the capacity in number of users. In order to work properly, a cellular system must verify the following two main conditions:

The power level of a transmitter within a single cell must be limited in order to reduce the interference with the transmitters of neighboring cells. The interference will not produce any damage to the system if a distance of about 2.5 to 3 times the diameter of a cell is reserved between transmitters. The receiver filters must also be very per formant.

Neighboring cells cannot share the same channels. In order to reduce the interference, the frequencies must be reused only within a certain pattern. In order to exchange the information needed to maintain the communication links within the cellular network, several radio channels are reserved for the signaling information.

2.1.2. Cluster

The cells are grouped into clusters. The number of cells in a cluster must be determined so that the cluster can be repeated continuously within the covering area of an operator. The typical clusters contain 4, 7, 12 or 21 cells. The number of cells in each cluster is very important. The smaller the number of cells per cluster is, the bigger the number of channels per cell will be. The capacity of each cell will be therefore increased. However a balance must be found in order to avoid the interference that could occur between neighboring clusters. This interference is produced by the small size of the clusters (the size of the cluster is defined by the number of cells per cluster). The total number of channels per cell depends on the number of available channels and the type of cluster used.

Types of cells

The density of population in a country is so varied that different types of cells are used:

1. Macro cells
2. Micro cell
3. Selective cells
4. Umbrella cells

Macro cells

The macro cells are large cells for remote and sparsely populated areas.

Micro cells

These cells are used for densely populated areas. By splitting the existing areas into smaller cells, the number of channels available is increased as well as the capacity of the cells. The power level of the transmitters used in these cells is then decreased, reducing the possibility of interference between neighboring cells.

Selective cells

It is not always useful to define a cell with a full coverage of 360 degrees. In some cases, cells with a particular shape and coverage are needed. These cells are called selective cells. A typical example of selective cells is the cells that may be located at the entrances of tunnels where coverage of 360 degrees is not needed. In this case, a selective cell with coverage of 120 degrees is used.

Umbrella cells

A freeway crossing very small cells produces an important number of handovers among the different small neighboring cells. In order to solve this problem, the concept of umbrella cells is introduced. An umbrella cell covers several micro cells. The power level inside an umbrella cell is increased comparing to the power levels used in the micro cells that form the umbrella cell. When the speed of the mobile is too high, the mobile is handed off to the umbrella cell. The mobile will then stay longer in the same cell (in this case the umbrella cell). This will reduce the number of handovers and the work of the network. A too important number of handover demands and the propagation characteristics of a mobile can help to detect its high speed.

2.2.The transition from analog to digital technology

In the 1980s most mobile cellular systems were based on analog systems. The GSM system can be considered as the first digital cellular system. The different reasons that explain this transition from analog to digital technology are presented in this section.

2.2.1 .The capacity of the system

As it is explained in section 1, cellular systems have experienced a very important growth. Analog systems were not able to cope with this increasing demand. In order to overcome this problem, new frequency bands and new technologies were proposed. But the possibility of using new frequency bands was rejected by a big number of countries because of the restricted spectrum (even if later on, other frequency bands have been allocated for the development of mobile cellular radio). The new analog technologies proposed were able to overcome the problem to a certain degree but the costs were too important. The digital radio was, therefore, the best option (but not the perfect one) to handle the capacity needs in a cost-efficiency way.

2.2.2.Compatibility with other systems such as ISDN

The decision of adopting a digital technology for GSM was made in the course of developing the standard. During the development of GSM, the telecommunications industry converted to digital methods. The ISDN network is an example of this evolution. In order to make GSM compatible with the services offered by ISDN, it was decide that the digital technology was the best option. Additionally, a digital system allows, easily than an analog one, the implementation of future improvements and the change of its own characteristics.

2.2.3. Aspects of quality

The quality of the service can be considerably improved using a digital technology rather than an analog one. In fact, analog systems pass the physical disturbances in radio transmission (such as fades, multi path reception, spurious signals or interferences) to the receiver. These disturbances decrease the quality of the communication because they produce effects such as fadeouts, cross talks, hisses, etc. On the other hand, digital systems avoid these effects transforming the signal into bits. This transformation combined with other techniques, such as digital coding, improves the quality of the transmission. The improvement of digital systems comparing to analog systems is more noticeable under difficult reception conditions than under good reception conditions.

2.3. The GSM Network

2.3.1. Architecture of the GSM network

The GSM technical specifications define the different entities that form the GSM network by defining their functions and interface requirements.

The GSM network can be divided into four main parts:

1. The Mobile Station (MS).
2. The Base Station Subsystem (BSS).
3. The Network and Switching Subsystem (NSS).
4. The Operation and Support Subsystem (OSS).

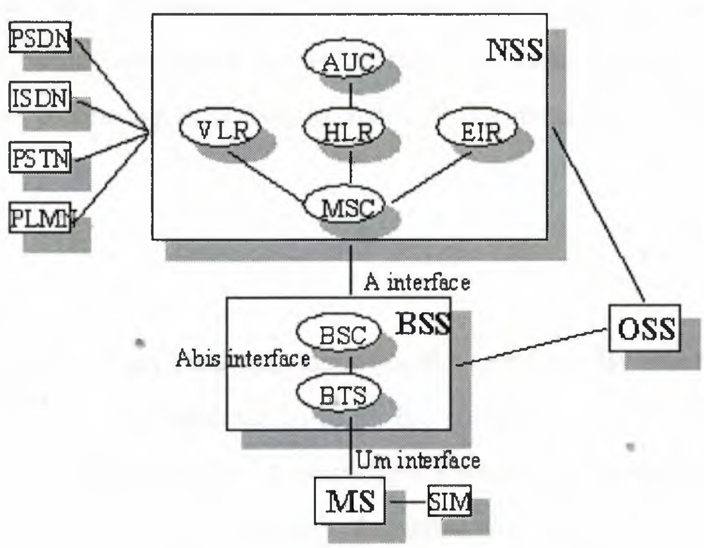

Figure 2.3: Architecture of the GSM network

2.3.2 Mobile Station

A Mobile Station consists of two main elements:

1. The mobile equipment or terminal.

2. The Subscriber Identity Module (SIM).

2.3.3 The Terminal

There are different types of terminals distinguished principally by their power and application:

- The `fixed' terminals are the ones installed in cars. Their maximum allowed output power is 20 W.

- The GSM portable terminals can also be installed in vehicles. Their maximum allowed output power is 8W.

- The handhels terminals have experienced the biggest success thanks to the weight and volume, which are continuously decreasing. These terminals can emit up to 2 W. The evolution of technologies allows to decrease the maximum allowed power to 0.8 W.

2.3.4 The SIM

The SIM is a smart card that identifies the terminal. By inserting the SIM card into the terminal, the user can have access to all the subscribed services. Without the SIM card, the terminal is not operational. A four-digit Personal Identification Number (PIN) protects the SIM card. In order to identify the subscriber to the system, the SIM card contains some parameters of the user such as its International Mobile Subscriber Identity (IMSI). Another advantage of the SIM card is the mobility of the users. In fact, the only element that personalizes a terminal is the SIM card. Therefore, the user can have access to its subscribed services in any terminal using its SIM card.

2.3.5.The Base Station Subsystem

The BSS connects the Mobile Station and the NSS. It is in charge of the transmission and reception. The BSS can be divided into two parts: The Base Transceiver Station (BTS) or Base Station. he Base Station Controller (BSC).

2.3.5.1.The Base Transceiver Station

The BTS corresponds to the transceivers and antennas used in each cell of the network. A BTS is usually placed in the center of a cell. Its transmitting power defines the size of a cell. Each BTS has between one and sixteen transceivers depending on the density of users in the cell.

2.3.5.2 .The Base Station Controller

The BSC controls a group of BTS and manages their radio resources. A BSC is principally in charge of handovers, frequency hopping, exchange functions and control of the radio frequency power levels of the BTSs.

2.4. The Network and Switching Subsystem

Its main role is to manage the communications between the mobile users and other users, such as mobile users, ISDN users, fixed telephony users, etc. It also includes data bases needed in order to store information about the subscribers and to manage their mobility. The different components of the NSS are described below.

2.4.1 The Mobile services Switching Center (MSC)

It is the central component of the NSS. The MSC performs the switching functions of the network. It also provides connection to other networks.

2.4.2 The Gateway Mobile services Switching Center (GMSC)

A gateway is a node interconnecting two networks. The GMSC is the interface between the mobile cellular network and the PSTN. It is in charge of routing calls from the fixed network towards a GSM user. The GMSC is often implemented in the same machines as the MSC.

2.4.3.Home Location Register (HLR)

The HLR is considered as a very important database that stores information of the subscribers belonging to the covering area of a MSC. It also stores the current location of these subscribers and the services to which they have access. The location of subscriber corresponds to the SS7 address of the Visitor Location Register (VLRassociated to the terminal.

2.4.4.Visitor Location Register (VLR)

The VLR contains information from a subscriber's HLR necessary in order to provide the subscribed services to visiting users. When a subscriber enters the covering area of a new MSC, the VLR associated to this MSC will request information about the new subscriber to its corresponding HLR. The VLR will then have enough information in order to assure the subscribed services without needing to ask the HLR each time a communication is established. The VLR is always implemented together with a MSC; so the area under control of the MSC is also the area under control of the VLR.

2.4.5. The Authentication Center (AuC)

The AuC register is used for security purposes. It provides the parameters needed for authentication and encryption functions. These parameters help to verify the user's identity.

2.4.6 The Equipment Identity Register (EIR)

The EIR is also used for security purposes. It is a register containing information about the mobile equipments. More particularly, it contains a list of all valid terminals. Its International Mobile Equipment Identity (IMEI) identifies a terminal. The EIR allows then to forbid calls from stolen or unauthorized terminals (e.g., a terminal which does not respect the specifications concerning the output RF power).

2.4.7.The GSM Interworking Unit (GIWU)

The GIWU corresponds to an interface to various networks for data communications. During these communications, the transmission of speech and data can be alternated.

2.4.8.The Operation and Support Subsystem (OSS)

The OSS is connected to the different components of the NSS and to the BSC, in order to control and monitor the GSM system. It is also in charge of controlling the traffic load of the BSS. However, the increasing number of base stations, due to the development of cellular radio networks, has provoked that some of the maintenance tasks are transferred to the BTS. This transfer decreases considerably the costs of the maintenance of the system.

2.5. The geographical areas of the GSM network

The figure 2.4 presents the different areas that form a GSM network.

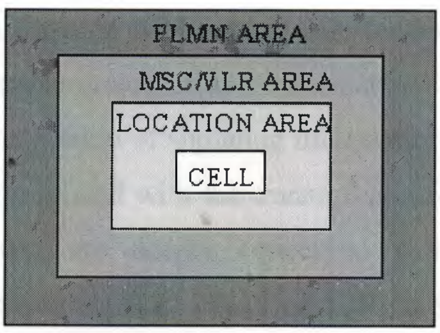

Figure: 2.4

As it has already been explained a cell, identified by its Cell Global Identity number (CGI), corresponds to the radio coverage of a base transceiver station. A Location Area (LA), identified by its Location Area Identity (LAI) number, is a group of cells served by a single MSC/VLR. A group of location areas under the control of the same

MSC/VLR defines the MSC/VLR area. A Public Land Mobile Network (PLMN) is the area served by one network operator.

2.6 The GSM functions

In this paragraph, the description of the GSM network is focused on the different functions to fulfill by the network and not on its physical components. In GSM, five main functions can be defined:

1.Transmission.

2. Radio Resources management (RR).

3. Mobility Management (MM).

4. Communication Management (CM).

5. Operations, Administration and Maintenance (OAM).

2.6.1 Transmission

The transmission function includes two sub-functions: The first one is related to the means needed for the transmission of user information. The second one is related to the means needed for the transmission of signaling information. Not all the components of the GSM network are strongly related with the transmission functions. The MS, the BTS and the BSC, among others, are deeply concerned with transmission. But other components, such as the registers HLR, VLR or EIR, are only concerned with the transmission for their signaling needs with other components of the GSM network.

2.6.2 Radio Resources management (RR)

The role of the RR function is to establish, maintain and release communication links between mobile stations and the MSC. The elements that are mainly concerned with the RR function are the mobile station and the base station. However, as the RR function is also in charge of maintaining a connection even if the user moves from one cell to another, the MSC, in charge of handovers, is also concerned with the RR functions. The RR is also responsible for the management of the frequency spectrum and the reaction of the network to changing radio environment conditions. Some of the main RR procedures that assure its responsibilities are:

Channel assignment, change and release.

Handover.

Frequency hopping.

Power-level control.

Discontinuous transmission and reception.

Timing advance.

Some of these procedures are described in section 5. In this paragraph only the handover, which represents one of the most important responsibilities of the RR, is described.

2.6.3.Handover

The user movements can produce the need to change the channel or cell, especially when the quality of the communication is decreasing. This procedure of changing the resources is called handover. Four different types of handovers can be distinguished:

Handover of channels in the same cell.

Handover of cells controlled by the same BSC.

Handover of cells belonging to the same MSC but controlled by different BSCs.

Handover of cells controlled by different MSCs.

Handovers are mainly controlled by the MSC. However in order to avoid unnecessary signaling information, the first two types of handovers are managed by the concerned BSC (in this case, the MSC is only notified of the handover). The mobile station is the active participant in this procedure. In order to perform the handover, the mobile station controls continuously its own signal strength and the signal strength of the neighboring cells. The base station gives the list of cells that must be monitored by the mobile station. The power measurements allow decide which is the best cell in order to maintain the quality of the communication link. Two basic algorithms are used for the handover:

1. The minimum acceptable performance' algorithm. When the quality of the transmission decreases (i.e. the signal is deteriorated), the power level of the mobile is increased. This is done until the increase of the power level has no effect on the quality of the signal. When this happens, a handover is performed.

2. The `power budget' algorithm. This algorithm performs a handover, instead of continuously increasing the power level, in order to obtain a good communication quality.

2.6.4. Mobility Management

The MM function is in charge of all the aspects related with the mobility of the user, specially the location management and the authentication and security.

2.6.5. Location management

When a mobile station is powered on, it performs a location update procedure by indicating its IMSI to the network. The first location update procedure is called the IMSI attach procedure. The mobile station also performs location updating, in order to indicate its current location, when it moves to a new Location Area or a different PLMN. This location-updating message is sent to the new MSC/VLR, which gives the location information to the subscriber's HLR. If the mobile station is authorized in the new MSC/VLR, the subscriber's HLR cancels the registration of the mobile station with the old MSC/VLR. A location updating is also performed periodically. If after the updating time period, the mobile station has not registered, it is then deregistered. When a mobile station is powered off, it performs an IMSI detach procedure in order to tell the network that it is no longer connected.

2.6.6. Authentication and security

The authentication procedure involves the SIM card and the Authentication Center. A secret key, stored in the SIM card and the AuC, and a ciphering algorithm called A3 are used in order to verify the authenticity of the user. The mobile station and the AuC compute a SRES using the secret key, the algorithm A3 and a random number generated by the AuC. If the two computed SRES are the same, the subscriber is authenticated. The different services to which the subscriber has access are also checked. Another security procedure is to check the equipment identity. If the IMEI number of the mobile is authorized in the EIR, the mobile station is allowed to connect the network. In order to assure user confidentiality, the user is registered with a Temporary Mobile Subscriber Identity (TMSI) after its first location update procedure. Enciphering is another option to guarantee a very strong security but this procedure is going to be described in section 5.

2.7. Communication Management (CM)

The CM function is responsible for:

Call control.

Supplementary Services management.

Short Message Services management.

2.7.1 Call Control (CC)

The CC is responsible for call establishing, maintaining and releasing as well as for selecting the type of service. One of the most important functions of the CC is the call routing. In order to reach a mobile subscriber, a user dials the Mobile Subscriber ISDN (MSISDN) number, which includes:

1. A country code

2. A national destination code identifying the subscriber's operator

3. A code corresponding to the subscriber's HLR

The call is then passed to the GMSC (if the call is originated from a fixed network), which knows the HLR corresponding to a certain MISDN number. The GMSC asks the HLR for information helping to the call routing. The HLR requests this information from the subscriber's current VLR. This VLR allocates temporarily a Mobile Station Roaming Number (MSRN) for the call. The MSRN number is the information returned by the HLR to the GMSC. Thanks to the MSRN number, the call is routed to subscriber's current MSC/VLR. In the subscriber's current LA, the mobile is paged.

2.7.2 Supplementary Services management

The mobile station and the HLR are the only components of the GSM network involved with this function. The different Supplementary Services (SS) to which the users have access are presented in section 6.3.

2.7.3 Short Message Services management

In order to support these services, a GSM network is in contact with a Short Message Service Center through the two following interfaces:

The SMS-GMSC for Mobile Terminating Short Messages (SMS-MT/PP). It has the same role as the GMSC.

The SMS-IWMSC for Mobile Originating Short Messages (SMS-MO/PP).

2.7.4. Operation, Administration and Maintenance (OAM)

The OAM function allows the operator to monitor and control the system as well as to modify the configuration of the elements of the system. Not only the OSS is part of the OAM, also the BSS and NSS participate in its functions as it is shown in the following examples: The components of the BSS and NSS provide the operator with all the information it needs. This information is then passed to the OSS, which is in charge of analyze it and control the network. The self-test tasks, usually incorporated in the components of the BSS and NSS, also contribute to the OAM functions. The BSC, in charge of controlling several BTSs, is another example of an OAM function performed outside the OSS.

3.GSM BASE STATIONS

3.1 Physical and logical blocks of a GSM mobile station

As in any communications systems, a digital cellular system's purpose is to provide reliable, cost-effective transmission and reception of the user's information. Digital cellular systems carry user information, including speech, computer or message data, and even facsimile information in digital form. Even though we talk broadly about a digital communications system, the physical transmission from a digital radio transmitter to a digital receiver is analog, and so is the audio interface to the users on both sides of the channel. The digital processes are confined within the phone. If we look inside a GSM MS we can identity a number of building blocks that process audio signals and user information before transmitting it.

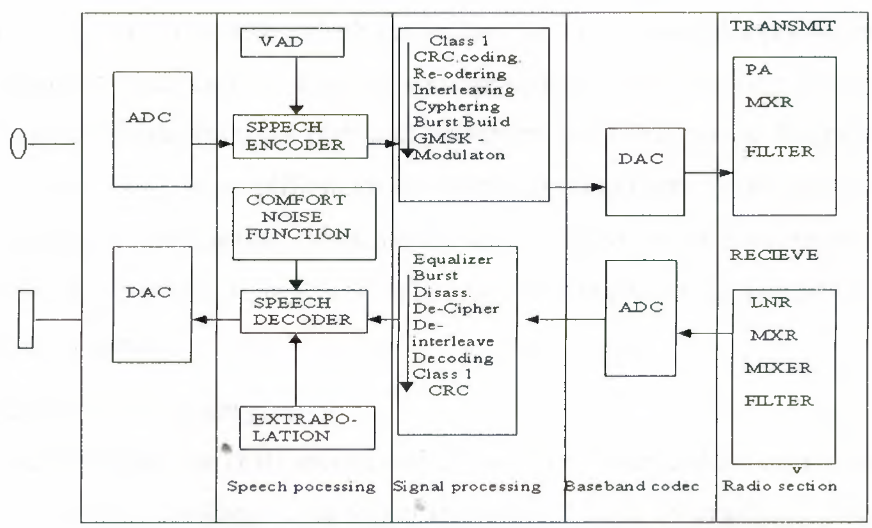

Figure 3.1 Block diagram of a GSM mobile station signal path

Some other blocks work in the opposite direction because they convert the received signals back into the form in which they entered the transmitter of the other end of the link. A measure of a phone's performance can be how "invisible" the phone appears to the user: the voce from the receiver's speaker should sound exactly as it did at the transmitter's microphone, and computer data should arrive at the receiver free of errors. Even as we strive to make the phone invisible, we need some kind of control and interface functions to do so. Figure 3.1 shows a logical physical and logical signal processing flow. The blocks in Figure 3.1 represent functions, not necessarily actual devices. As we will see because many of the signals processing functions share similar processes and techniques, only a few physical devices perform the tasks.

3.2.The voice codec

The voice codec performs the conversions between the radio's speech processing functions, which are digital processes, and the analog audio domain outside the phone. We find two converters in the voice codec: (1) the analog-to-digital convener (ADC) for the analog input signal from microphone and (2) the-digital-to-analog converter DAC) for the recovered analog output signal to speaker. The output of the voice codec's transmit path is the digital input to the GSM speech codec. The input to the speech codec is a 13-bit representation, (13-bit resolution) of the voice signal sampled at 8 kHz. This 13-bit linear pulse code modulation (PCM) is represented by a 104-Kbps data stream (13 x 8,000 = 104,000), which means there are 8,000 samples per second, where 13 bits represent each sample. The voice codec contains band-limiting low-pass filters for anti-aliasing in the transmit path and waveform reconstruction in the receive path. There are also some gain amplifiers for the signals coming from the microphone and the signals that drive the speaker. Multiplexers can be used in order to switch acoustic signal paths, for example, from a built-in handset microphone or speaker to a hands-free microphone or speaker.

3.3.GSM speech processing

The GSM full rate (FR) speech codec (speech encoder and decoder) compresses The transmit side's 104-Kbps voice signal into only 13 Kbps by drastically reducing the amount of redundant information from the voice codec. The 13-Kbps speech data rate is more suited for transmission over the limited radio resource, which is a sharp contrast to the typical ISDN (fixed-line) PCM signal, which represents speech at a relatively high rate of 64 Kbps .For GSM speech encoding/decoding on the fixed network side, a conversion between the fixed-line 64-Kbps (8-kHz sampling rate x 8-bit resolution) and the GSM 104-Kbps speech codec (8-kHzx I 3-bit resolution) is done by companding and expanding tie digital representation of the audio in accordance with the A-law or -law scales.

A compander converts the higher resolution 13-bit values to 8-bit values by quantizing low-amplitude signals more precisely than high-amplitude values. Low-amplitude samples are given a higher resolution with smaller step sizes, and high-amplitude values get a lower resolution with larger step sizes. The reverse process of compressing is called expanding, hence, companding. Further significant reductions in redundant speech information and data rates call for a much more intricate processes beyond the simple quantization process; the final reduction in payload data occurs in the speech coder. The counterparts of the mobile terminal's speech codec can be found in the networks in different physical places, for example, with the BTS, BSC, or MSC, although it is logically associated with the BSS. The placement depends on network topology and line traffic considerations. The transport of speech data on the network takes place either in encoded logical 16-Kbps sub channels (13 Kbps + synchronization data. sub multiplexed from 64-Kbps channels), which is the most economical way in terms of line traffic, or in standard 64-Kbps channels.

The GSM full-rate speech coder is a type of hybrid coder called a regular pulse excited long-term prediction codec (RPE-LTP). The output of the GSM speech coder occurs in frames of 260 bits at a rate of one every 20 ms (260 bits/20 ms = 13,000 bps). This speech data (separated into class 1 and class 2 bits) is passed on to the channel coding process, which prepares the codec speech bits for transmission on the radio channel. The reverse side of the process in the receiver sees 260-bit frames coming from the channel decoder, which are used by the speech decoder to recreate the speech sounds that specified the bits the coder generated. Auxiliary functions such as voice activity detection (VAD), extrapolation, and a comfort noise function support features such as discontinuous transmission (DTX) and substitutions for lost terms, DTX is an effective way to save battery capacity by minimizing transmit cycles when no voice activity is detected. This feature also reduces interference in adjacent cells.

Two more speech coders are specified for use in GSM networks. One is the half-rate (HR| coder for use in half-rate channels [4]. GSM half-rate frees up the use of every other time slot in a single full-rate traffic channel, thus doubling the channel capacity. The enhanced fun-rate (EFR) coder uses the same rate as normal full rate |5]. It was required by [lie North American personal communications industry for use in PCS 1900 networks-

The EFR algorithm that was chosen (known as the US-1 codec), and also adopted by ETSI (through GSM Phase 2), offers much better perceived voice quality.

3.4. Signal processing

The signal processing function works with the user data (speech or data traffic), as well as signaling and control information. Payload data must be well protected before transmission over the radio channel, and the protective measures must be "undone" on the received data. The GSM channel coding block protects speech coder information with several processes: cyclic redundancy check (CRC) for the most important class I bits, followed by convolutional coding, and interleaving. User data (other than voice data) are transmitted over GSM data channels, which arc convolutionally coded and interleaved with their own unique schemes, depending on the data rate, that are different from those used on voice data. Signaling data are block coded, convolutionally coded, and block interleaved according to their own rules. A common ciphering scheme can be applied to the channel coded data. The channel decoding block reverses the transmitter's coding for the received data. The burst building and multiplexing process adds a midamble (training sequence code), tail bits, and guard bits to the encoded bits after the interleaving, and makes sure that the burst data (radio bits) are delivered to the modem and the radio at the right time. The radio bits are differentially encoded and modulated through a Gauss Ian filter function as a part of modem function .The reverse side of the modem function is dominated by the channel equalizer, which adaptively compensates for the adverse effects (inter-symbol interference) applied to the original signal through time-variant multi path interference, delay spread, and Doppler shifts.

3.5. Baseband Codec

The baseband codec converts the transmit and receive data into analog or digital signals, respectively. The transmit process delivers analog baseband I- and Q-signals (from two DACs). Which modulate the radio carrier. The receive process converts the analog I- and Q-signals from the radio back to a digital data stream, which is filtered, sampled, and quantized (again, two ADCs) before presentation to the equalizer. A typical value for the convener's resolution is 10 bits for both directions.

3.6.The radio section

The radio section, which includes the IF stages, is one of the interfaces to the outside world; the other is the audio interface, functions such as frequency synthesis, a local oscillator, up-and-down con version with mixers (MXR in Figure), analog RF and IF filtering, transmitter power amplification (PA in the transmit path of Figure 2.1), pre amplification in the receive path (LNA in Figure 2.1), RF pulse shaping, automatic gain control, and frequency correction and control are handled in the radio section..

Other functions

Apart from routine signal routing and conditioning tasks between audio, baseband, and radio blocks, there are some other functions that complete the GSM mobile station. Some control units are required to organize and schedule logical and physical entities. On top of the physical layer functions, which should be viewed as the physical conditioning and processing of payload data (coding and decoding), we need to accommodate some higher level tasks such as < I) synchronization, (2) frequency and time acquisition, and tracking, (3) monitoring (serving cell and adjacent cells), (4) received signal strength measurements, and (5) radio control. Above these radio tasks we need to add the higher protocol layers (layers 2 and 3) with signaling functions, which allow the phone to operate in an orderly way within a network. Other control functions within a GSM mobile station handle the user interfaces (keypad, display, and the user menu). The SIM interface, and other auxiliary (data) interfaces.

3.7.Physical and logical blocks of a GSM base station

A GSM base station is the mobile's counterpart, but not in al! Its aspects. The base station does not provide for most of the layer 3 signaling functions, and none of the higher layer functions, because it is transparent to those functions coming from the network. Because the base station converts the network's wire environment to an air interface, base stations exercise their own control over the layer 1 and layer 2 processes. The signal processing functions in a GSM base station is similar to those in a mobile station. Additional logical channel structures, not found in the mobile station's uplink, must be supported by a base station: (1) common control channels (CCCH), which comprise the pacing channel (PCH) and the access grant channel (AGCH); (2) tile broadcast control channel (BCCH|; (3) the frequency correction channel (FCCH); and (4) the synchronization channel (SCH). All of these are unidirectional channels; a mobile does not support these channels beyond being able to read and understand them. The interface to the fixed network requires some wire line physical and logical entities that provide transmission and reception capabilities for the user data, signaling data, and control information. One of these entities is the Transcoder rate adaptation unit (TRAU), which supports the multiplexing of speech data to and from the Abis interface. Just as there are

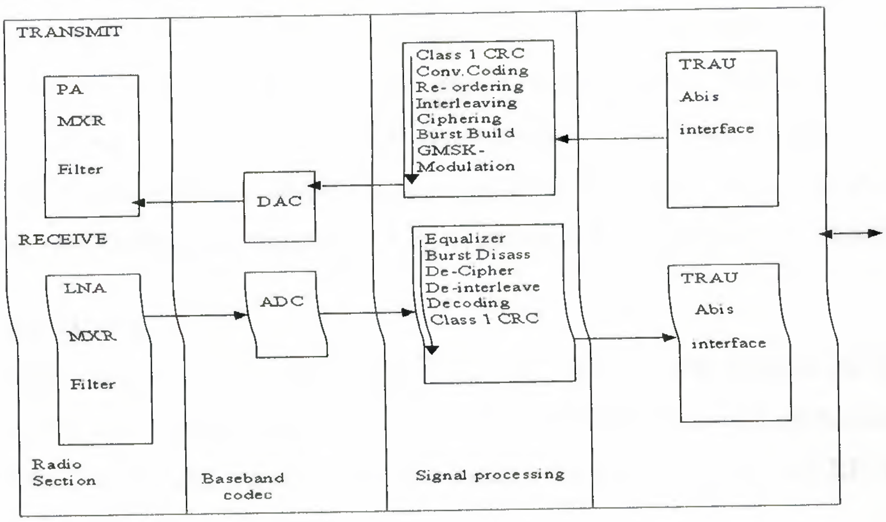

FIGURE 3.2 Sample block diagram of GSM base station signal

many configurations possible within a GSM network, so there are much architecture of GSM base stations. There are different applications; different numbers of channels, different configurations for standby channels, and different network interface configurations. Fig 3.2 shows an example, which assumes that speech codec, is placed remotely in the network (e.g., within the mobile switching center), and shows the architecture for a single transceiver. The appropriate signal processing and radio sections would exist in parallel in multiple-transceiver and architecture.

3.8.Transmitters and receivers

The radio sections in cellular phones and base stations require an enormous design effort, Circuits need to meet specifications even as they arc manufactured in large quantities at low cost. In particular, this means that analog circuitry must be produced with an absolute minimum of individual "tweaking" (tuning and calibration) exercises such that millions of "equal" radios are the result. Radio design, which is strictly separated from the digital design task, is sometimes regarded as "black magic" or "alchemy" rather than a disciplined and orderly engineering task.

These kinds of references arise from myths surrounding RF work that are given prolonged life as "lab legend" by those unfamiliar with the techniques and skills of RT" practice. Radio design is a balancing act in which the successful practitioners find the-optimum combination of technology, technique, and cost, and performance. The skill and experience of the attending engineer is the key to meeting design goals, Figure 3.3 shows one sample block diagram of a digital radio structure that may be used in a GSM. Transceiver. Here we find a single conversion transmitter and a dual conversion, super heterodyne receiver with quadrature (I/O) modulation and demodulation, respectively.

3.8.1.Transmitters

There are two kinds of radio transmitters in GSM: (1) the ones in the BTS, which create the physical forward side of the RF link to the mobile station (downlink), and (2) those we find in the MS, which creates the other---the reverse side of the RF link back to the base site (uplink)

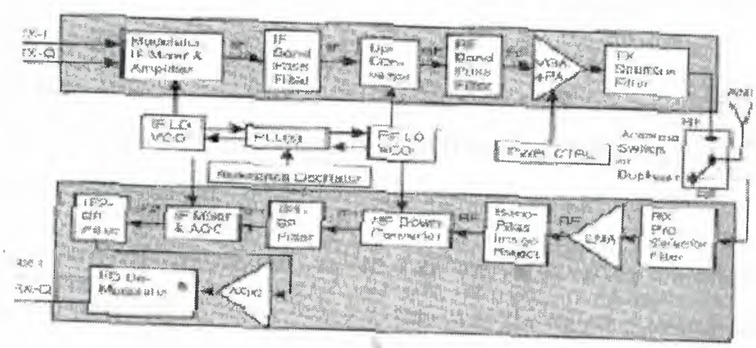

Fig 3.3 A GSM radio transceiver architecture example

The chief difference between these are (1) the much greater RF power, relative to the MS, generated in the BTS, and (2) the tiny size of the MS's receiver. The similarities outnumber The differences. Those who design transmitters for GSM service are obliged to consider the following parameters.

1 .The power output is seldom more than 2W for an MS, but it can be more than an order of magnitude greater in a RTS. The amplification job usually needs to he distributed over several stages in the high-power applications in base stations. The transmitter's output power needs to be managed over temperature as well as with changes in the source impedance of the power source for the RF amplifier stages, and the RF output components (including the antenna).

2. Power consumption (efficiency of amplifiers) is critical in the MS.

3. Load pull is the transmitter's tendency to change its output frequency with impedance changes at the antenna, or in the power source of the transceiver.

4. Modulation accuracy (linearity) is compromised by attempts to increase the efficiency of the transmitter.

5. SNR, hum, and noise are general terms for those factors that limit the best signal-to-noise ratio (SNR) that we can expect from a transmitter. Linearity problems in the modulation conspire with noise sources and unwanted RF coupling to put a limit on our best efforts.

6. Spurious outputs can be either harmonic ones or all of those undesired outputs that are not harmonics of the desired output |9]. The former usually come from linearity problems in the RF amplification stages of the transmitter, and the latter find their origin in mixers and frequency synthesizers.

7. Excessive adjacent channel power comes from too much single sideband (SSB) phase noise from the transmitter's oscillator, or poor fractional frequency stability performance of the frequency synthesizer.

8. Intermodulation distortion appears when a strong signal from outside the transceiver is coupled into the transmitter, through the antenna, and mixing products are created in one of the many nonlinear mixing sizes anywhere in the transmitter's signal path. This is a particularly troublesome problem in base sites, because base stations often share site space with other high-powered transmitters such as TV transmitters.

Seven of the transmitter's functional components that are not a part of the digital baseband functions are listed next in order in which they appears as the transmit path approaches to antenna

1. An oscillator is one of the reference inputs to a phase-locked loop (PLL) synthesizer.

2. The frequency synthesizer functions as an intermediate/radio-frequency voltage-

controlled oscillator (IF/RF VCO), which is a means to add frequency agility to the fixed reference oscillator. The relatively high RF levels of the transmitter's output reduce the required noise performance of the synthesizer compared to what is required of the same function in the receiver's local oscillator (LO).

3. The modulator imparts intelligence to the carrier. GSM's GMSK modulation affords many possibilities for modulators, but some kind of I/Q process is the general method . 1/Q modulators are sometimes referred to as "universal modulators." We could, for example, use a more direct technique than an I/Q method to generate MSK, such as a pair of oscillators operating 67 kHz on either side of The assigned channel frequency and then use the modulator's data stream input to enable one or the other oscillator, chief among the many reasons such direct methods are not used in digital radio is that the frequency shifts are-sloppy and abrupt. The more complicated I/Q method can be made to generate a wide variety of digital modulation schemes including GMSK. They work by enlisting two double-balanced mixers into which we apply alternate halves of the symbol data stream; first into one mixer, and the next into the other mixer. Each mixer has a reference input of the transmitter's carrier frequency shifting from each other by 90 degrees. The outputs are summed and passed into the up-converter. The utility of this quadrature technique is that the phase and the amplitude of both parts of the data input can be adjusted and filtered through look-up tables before they are applied to the mixers.

4. An up-converter (mixer) brings the transmitter's output up to the assigned frequency. There arc many kinds of mixers, and all of them produce odd order products that have to be filtered from the transmitter's output [10]. The double balanced variety is often used, because it isolates the frequency synthesizer from the variable load of the RP power amplifier.

5. Amplification raises the transmitter's output to the required RF level specified for the task. The job is made particularly difficult in the GSM mobile station, because the power has to be adjusted over a wide range and intermittently keyed within strict burst power-time templates without the AM splash that usually accompanies intermittent duty transmitters. All of this has to be accomplished in a small volume with a limited power source (battery).

6. Filters attenuate most kinds of spurious outputs before they reach the transmitter's antenna, but they have none of the effects of unwanted outputs caused by linearity

problems in the modulator. Filter design has become a very specialized skill, and the possibilities are enormous.

7. Though the half-duplex operating mode of GSM may seem to eliminate the need for a duplexer, it is often included in mobile stations. The duplexer is a filter than is common to the transmitter and the receiver, and it is included to (1) add receiver front-end filtering to reduce the harmful influences of nearby transmitters in other phones and (2) increase the isolation between the transmitter and the receiver.The transmitter's performance characteristics, which we listed in this section, arc mostly a matter of regulatory compliance. Designs must comply with the requirements, but there is no compelling reason to increase costs by improving on them. When our transmitter designs fail to meet the required minimum performance we cannot expect type approval. The requirements are, however, not capricious and arbitrary barriers designed merely to increase the costs of radios offered into GSM service; they are designed to make the best use of the spectrum, and yield a balance between system capacity, costs, and the best quality of service the technology can deliver.

3.8.2.Receivers

The wide use of traditional radio technology and techniques in the RF functions contrasted with the use of integrated functions on silicon for the baseband functions that we examined in the transmitter is also evident in receivers. The receiver's job is to extract the signal from a distant transmitter and recover the modulation. Once the modulation is recovered, advanced techniques

(1) Equalize and decode the channel (2) separate the signaling from traffic information, (3) recreate the voice sounds that entered the transmitter, and (4) bring audio or user data out of the transceiver,

GSM's minimum requirement is to be able to recover a transmitter's symbol stream with less then only 1-bit error in 100 for signals for less then -102 dBm, which is only 6.3 x 10^{14} The noise present at the input of a GSM receiver will be at best, about -121 dBm. This bit error ratio (BER) requirement as criteria for receiver sensitivity is rather modest when compared to wire line system where one error in one million symbols is considered intolerable. Because channels in wire line systems are relatively stable and orderly compared to those in wireless systems, the protection coding in wire line systems would be too weak to be able 10 respond to the error rates typical in radio channels. Current GSM receivers perform far better than the minimum -102-dBm requirement. Typical implementations achieve the required RER performance at nominal levels of between -105 and -110 dBm. Outside system influences, including the presence of strong blocking signals, or co-channels and adjacent channels interferers, Doppler and deep fades, quickly reduce margins in the designs. To recover information with low BBR from a signal that is suffering interference from a co-channel signal (from a distant base station or mobile station using the same physical channel), the available signal bandwidth should be high. The opposite is the case for adjacent channel interference, which typically occurs at higher levels than co-channel interference. Here we want to have an as narrow as possible filter characteristics in the signal recovery path in order to separate the good signal from the bad ones.

Surface acoustic wave (SAW} filters are the device of choice for IF filters. They are piezoelectric transducers on which interfering mechanical waves are setup such that the devices become band pass filters with sharp cutoff characteristics. SAW devices have high insertion losses (about 20 to 30 dB). The most narrow bandwidth characteristic is selected such that it can still pass the received signal without distorting the symbols that eventually appear from the demodulator; too narrow a filter will hurt the BER.

In general, improved system performance in wireless networks is limited to optimizing receivers, not the transmitters, and all kinds of tricks and techniques are brought to receiver designs lo improve BER performance, The improvements lend to be confined to the baseband processes (DSPs or ASICs) after the demodulator or detector. All air interface standards go through a maturing process, such that we can see, as a rough rule, a steady improvement in the average BRR performance of all the receivers deployed in a particular kind of air interface technology of about 1 dB each year. Eventually, we see receiver improvements slow down, such that each 1-dB improvement takes much longer than a year to achieve, and each improvement quickly becomes more and more costly. This slowing in receiver performance improvements occurs as they slowing in receiver improvements, then we declare the host air interface technology to be "mature," and we can start to look for spurts of innovation elsewhere in the systems that wring additional capacity, performance and quality from the system by further lowering average BERs throughout the systems.

We also see increased interest new air interface proposals as system improvements slow and become more expensive. A receiver fulfills its duties by processing the small amounts of energy at the antenna such that the original information is recovered with minimum distortion and errors. The specifications that arc used to judge a receiver's particular abilities to do its work are discussed in the following paragraphs.

The sensitivity of a receiver refers to its ability to react properly to a weak signal. Digital receivers use the maximum BER at some low RF level as a measure of their performance. This is analogous to the SNR Technique used in analog receiver's . The SNR technique is not useful in digital receivers because they have some kind of decision circuit early in their signal recovery and analysis stages that make constant determinations on what the actual transmitted symbols probably were. If the BER= 1%, then the decision circuit makes one bad decision for each 100 symbols it is called on to judge

The selectivity is a receiver's ability to reject unwarned signals that are very close to the frequency of the desired signal. These interfering signals are usually spurious emissions from nearby transmitters. The duplexer and the IF stage's SAW filter are the devices of interest here. The average price of phones can he held down with strict adherence to outside system matters such as low transmitter adjacent channel power emissions, strict frequency planning, and proper system design. Intermodulation rejection is the receiver's ability to overcome its own natural tendency to generate an internal on-channel signal from off-channel signals present at the antenna. The receiver's spur-free dynamic range is a measure of its immunity to intermodulation. Which is primarily determined by the receiver's input stages. Figure illustrates the derivation of the IF3, or third-order intercept point, and the spur-free dynamic range of a receiver.

A receiver's non linearity can be measured by injecting two closely spaced signals (fl and f2) of increasing but equal amplitude into the RF input (antenna port) while observing the rise in third-order intermodulation products at both 2(f1)-f2 and 2{f2} –f1. Intermodulation products appear in the receiver when the input test signals exceed whatever level drives the receiver beyond its spur-free dynamic range. Figure shows that the level of the third-order intermodulation products (the dark curve with the greatest slope in the figure) rises a rate three times greater than the rise in the input signals (the dark curve to the left of the steep curve).

We confine our attention to the third-order products because all of the other intermodulation products are far outside the receiver's own pass band. As the amplitude of the input signals increases, the level of the receiver's output, both the fundamental signals and intermodulation products, increases until the input stages start to saturate (the curved parts of the dark traces in the figure 3.4 (f we extrapolate the straight parts of both curves in Figure beyond the saturation levels, then the point at which both lines intercept is called the third-order intercept (IP3},

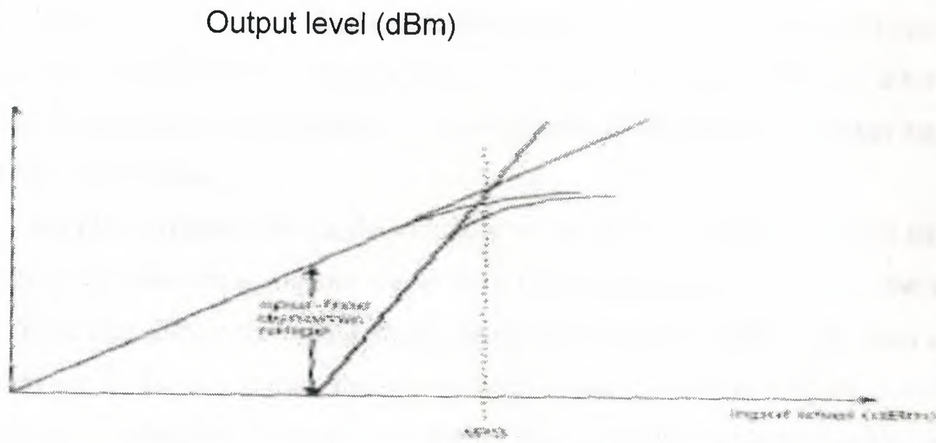

Figure 3.4 Receiver dynamic ranges,

The further to the right IP3 is in Figure 3.5 the better the receiver's intermodulation rejection performance. Receivers exhibiting poor intermodulation performance will loose sensitivity; we say the receivers suffer from desensitization, as they pass close to transmitters in a different network from that whose services the subscriber is using. Image rejection is a measure of the receiver's ability to attenuate signals that appear in the receiver's IF stages, or output, as a result of an off-channel signal with the same onset from the lo frequency as the desired input signal. Figure illustrates how images arc rejected from a receiver's output, The top part of the figure depicts a. spectrum analyzer display on which we see our desired signal (made fatter then the other signals for ease of identification) together with other that are not particularly interesting to us and need to he filtered out- The frequency of our LO is plotted in the center of the scale. The second part of the figure has the pass band of the RF preselection filter, our duplexer, plotted on the scale; signals outside the pass band are attenuated.

The third part of the figure plots the down converted outputs of the receiver 's IF where the LO frequency translates itself to 0 HZ. The subsequent IF filter's pass band is plotted on the right side of the "IF spectrum " scale together with another image channel the mixer translates to -IF. The bottom part of Figure 3.5 depicts the result out of the TF stages. Since the receiver's detector cannot distinguish between the desired signal and the image, it accepts the superposition of both signals. If the lower side of the preselector filter is less then 2 x IF away from the desired signal (the condition shown in die second part of the figure), then the image will be attenuated. Raising the IP will allow more freedom in preselector specifications but will demand IF fillers with a smaller fractional pass band specification.

With the exception of the demodulator or the detector, the devices and functions in the receiver have the analogous duties they had in the transmitter, but in the reverse order. The chief difference is that since the signal levels are millions of times smaller than they are in the transmitter, the required performance of the amplifiers, mixers, and oscillators (including the synthesizer) is higher. Just as we did in the transmitter, we will follow the signal from the antenna back toward the user.

1. A duplexer is a three-port filter. The antenna is attached to a common port, and the receiver is attached lo another port where we find low loss at the receiver's operating frequencies between the receiver port and the antenna. The transmitter is connected to the remaining part where another low-loss path. At the transmitter's operating frequencies, is found to the antenna. The filter is constructed in such a way as to isolate the transmitter from the receiver. Even though the TDMA/TDD timing in a GSM mobile station calls for a minimum idle time of two time slots between transmit and receive instances, the isolation the duplexer affords between the transmitter and the receiver is valuable.

2. A preselector filter limits the bandwidth of the receiver, which reduces intermodulation (IM) distortion. Additional preselector filtering is sometimes required to reduce the cost of the duplexer.

3. The RF amplifier also referred to as a low-noise amplifier (LNA)— determines the IP3 of the receiver, and isolates the preselector tiller from the image filter that follows the amplifier. As we discovered in Figure 3.4, the location of IP3 is a way we can judge a receiver's ability to limit any harmful effects off-channel signals may have on the receiver's performance. Intel-modulation distortion is generated in active components in

the receiver's circuits, which act as mixers for the input signal and another off-channel signal that yields an output within the receiver's pass band. The RF amplifier raises the level of the input signal together with the noise. Improvements in intermodulation performance of any RF amplifier are usually accomplished at the expense of additional current requirements for the amplifier, thus reducing battery capacity. The influence of the noise a device can contribute in a circuit also fights our efforts to optimize a receiver's treatment of weak signals. Space restrains us from including a treatment of these important influences.

4. An image filter can follow the RF amplifier to attenuate spurious signals and images that also attenuates harmonic distortion generated in the RF amplifier.
5. The LO is the mix-down reference for the input signal, and it is generated in the same frequency synthesizer that creates the transmitter's carrier, hut not on the same frequency. An injection filter between the LO and the mixer will attenuate noise from the LO.
6. The mixer accepts the LO's input together with the amplified and filtered input signal to generate a lower frequency copy of the input signal. The output is called the intermediate frequency (IF). There are many kinds of mixers, each with their own advantages and disadvantages.
7. The detector or demodulator is the last analog function in the receiver. It reverses the transmitter's modulation process in order to recover the original I- and Q-signals, which are send to the baseband processes as shown in the lower left corner of Figure 3.3.

Designing a receiver is an elaborate balancing act that starts with allocating gains, losses, and signal levels over all eight blocks in the receiver's
RF and IF paths. The process is reduced to an orderly one in which is highly recommended for the reader new to this kind of work. As the design proceeds, the designer will find herself clearing spurious signals generated in the last mixer stages, which starts the balancing act over again.

4. THE GSM AIR 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. 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.

4.1. Frequency allocation

Different frequency bands are used for GSM 900/1800 and GSM 1900 as shown in table 4.1.In some countries, an operator applies for the available frequencies. In other countries, e.g. United States, an operator purchases available frequency bands at auctions.

Network type	Frequency band UL/DL
GSM 900	890-915/935-960 MHz
GSM1800	1710 - 1785/1805 -1880 MHz
GSM1900	1850 - 1910/1930 -1990 MHz

Table 4.1 Frequency bands for the different GSM-based networks

4.2.Duplex Distance

The distance between the uplink and downlink frequencies is known as duplex distance. The duplex distance is different for the different frequency bands (table 4.2).

Standard	GSM 900	GSM1800	GSM 1900
Duplex distance	45MHz	95MHz	80MHz

Table 4.2 Duplex differences for different frequency bands

For the GSM-900 system two frequency bands have been made available:

- 890 - 915 MHz for the uplink (direction MS to BS)
- 935 - 960 MHz for the downlink (direction BS to MS).

- The band 890-915 Mhz has been allocated for the uplink direction (transmitting from the mobile station to the base station).
- The band 935-960 Mhz has been allocated for the downlink direction (transmitting from the base station to the mobile station).

The 25 MHz bands are then divided into 124 pairs of frequency duplex channels with 200 kHz carrier spacing using Frequency Division Multiple Access (FDMA). Since it is not possible for a same cell to use two adjacent channels, the channel spacing can be said to be 200 kHz interleaved. One or more carrier frequencies are assigned to individual Base Station (BS) and a technique known as Time Division Multiple Access (TDMA) is used to split this 200 kHz radio channel into 8 time slots (which creates 8 logical channels). A logical channel is therefore defined by its frequency and the TDMA frame time slot number. By employing eight time slots, each channel transmits the digitized speech in a series of short bursts: a GSM terminal is only ever transmitting for one eighth of the time.

8-slot TDMA together with the 248 physical half-duplex channels corresponds to a total of 1984 logical half-duplex channels. This corresponds to roughly 283 (1984 / 7) logical half-duplex channels per cell. This is because a cell can only use one seventh of the total number of frequencies, see Figure 4.3

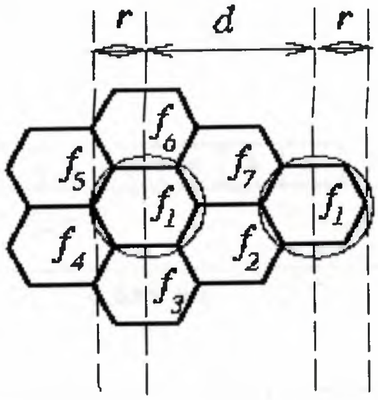

Figure 4.3 Typical cellular scheme

Seven sets of frequencies are sufficient to cover an arbitrarily large area, providing that the repeat-distance d is larger than twice the maximum radius r covered by each transmitter. Each of the frequency channels is segmented into 8 time slots of length 0.577 ms (15/26 ms). The 8 time slots makes up a TDMA frame of length 4.615 ms (120/26 ms). The recurrence of one particular time slot every 4.615 ms makes up one basic channel. The GSM system distinguishes between traffic channels (used for user data) and control channels (reserved for network management messages). In this overview, we consider only the Traffic Channel/Full-Rate Speech (TCH/FS) used to carry speech at 13 kbps.

TCHs for the uplink and downlink are separated in time by 3 burst periods, so that the mobile does not has to transmit and receive simultaneously. TCHs are defined using a 26-frame multiframe (i.e. a 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 / 26 frames / 8 burst periods per frame). Out of the 26 frames, 24 are used for traffic, one is used for the Slow Associated Control Channel (SACCH) and one is currently unused (see Figure 4.4)

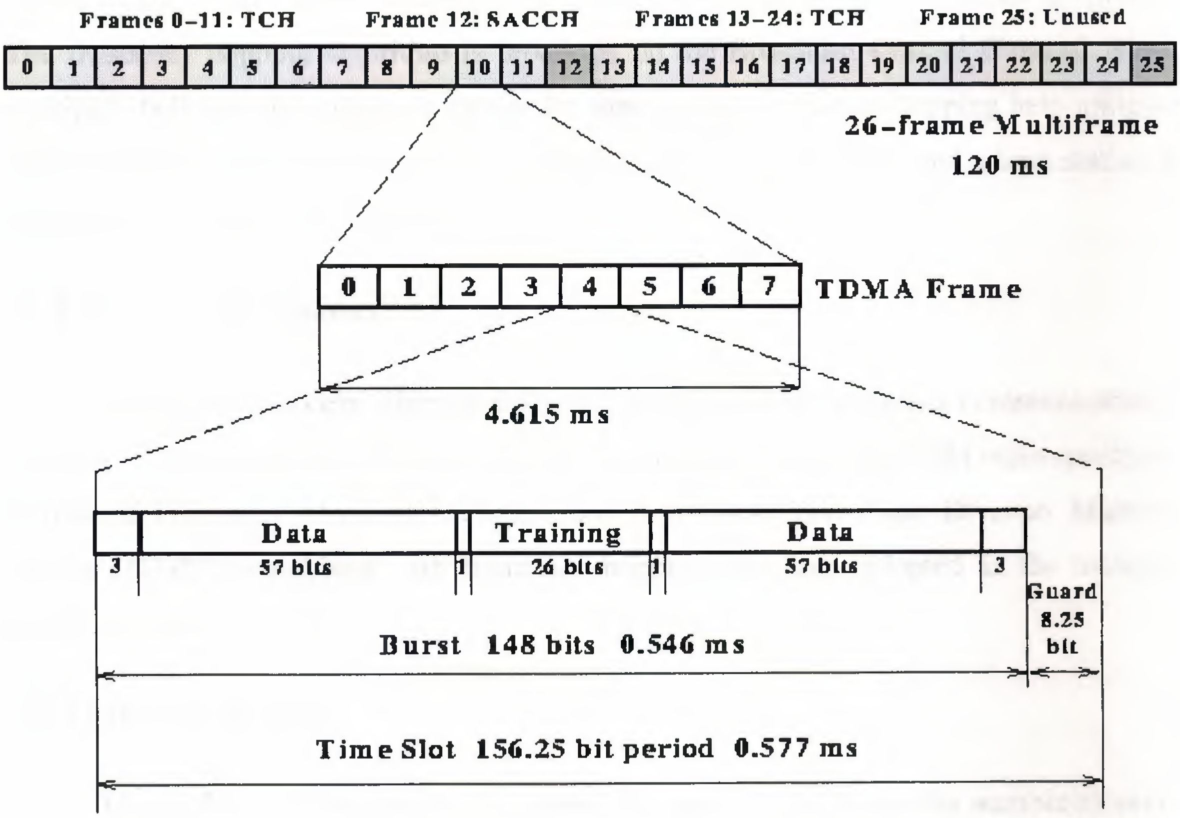

Figure 4.4 The TDMA frame structure

Data are transmitted in bursts which are placed within the time slots. The transmission bit rate is 271 kb/s (bit period 3.79 microseconds). To allow for time alignment errors, time dispersion etc, the data burst is slightly shorter than the time slot (148 out of the 156.25 bit periods available within a time slot). The burst is the transmission quantum of GSM. Its transmission takes place during a time window lasting (576 + 12/13) microseconds, i.e. (156 + 1/4) bit duration. A normal burst contains two packets of 58 bits (57 data bits + 1 stealing bit) surrounding a training sequence of 26 bits. The 26-bit training sequence is of a known pattern that is compared with the received pattern in order to reconstruct the rest of the original signal (multipath equalization). The actual implementation of the equalizer is not specified in the GSM specifications. Three ``tail'' bits are added on each side. GSM can use slow frequency hopping where the mobile station and the base station transmit each TDMA frame on a different carrier frequency. The frequency hopping algorithm is broadcast on the Broadcast Control Channel. Since multipath fading is dependent on carrier frequency, slow frequency hopping help mitigate the problem. Frequency hopping is an option for each individual cell and a base station is not required to support this feature.

4.3. Multiple access schemes

The multiple access schemes 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.

4.3.1.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 band 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.

4.4.Channel structure

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:

- The traffic channels used to transport speech and data information.
- The control channels used for network management messages and some channel maintenance tasks.

4.4.1.Traffic channels (TCH)

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:

- 24 frames are reserved to traffic.
- 1 frame is used for the Slow Associated Control Channel (SACCH).
- 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.

4.4.2. Control channels

According to their functions, four different classes of control channels are defined:

- Broadcast channels.
- Common control channels.
- Dedicated control channels.
- Associated control channels.

4.4.2.1.Broadcast channels (BCH)

The base station, to provide the mobile station with the sufficient information it needs to synchronize with the network, uses the BCH channels. Three different types of BCHs can be distinguished:

- The Broadcast Control Channel (BCCH), which gives to the mobile station the parameters needed in order to identify and access the network
- The Synchronization Channel (SCH), which gives to the mobile station the training sequence needed in order to demodulate the information transmitted by the base station
- 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

4.4.2.2.Common Control Channels (CCCH)

The CCCH channels help to establish the calls from the mobile station or the network. Three different types of CCCH can be defined:

- The Paging Channel (PCH). It is used to alert the mobile station of an incoming cal
- The Random Access Channel (RACH), which is used by the mobile station to request access to the network
- The Access Grant Channel (AGCH). The base station, to inform the mobile station about which channel it should use, uses it. This channel is the answer of a base station to a RACH from the mobile station

4.4.2.4. Dedicated Control Channels (DCCH)

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:

- The Standalone Dedicated Control Channel (SDCCH), which is used in order to exchange signaling information in the downlink and uplink directions.
- The Slow Associated Control Channel (SACCH). It is used for channel maintenance and channel control.

4.4.2.5. Associated Control Channels

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.

4.5.Burst structure

As it has been stated before, the burst is the unit in time of a TDMA system. Four different types of bursts can be distinguished in GSM:

- The frequency-correction burst is used on the FCCH. It has the same length as the normal burst but a different structure.
- The synchronization burst is used on the SCH. It has the same length as the normal burst but a different structure.
- The random access burst is used on the RACH and is shorter than 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.
- 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.

4.6. 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.

4.7. From source information to radio waves

The figure 4-5 presents the different operations that have to be performed in order to as from speech to radio waves and vice versa, If the source of information is data and not speech, the speech coding will not be performed.

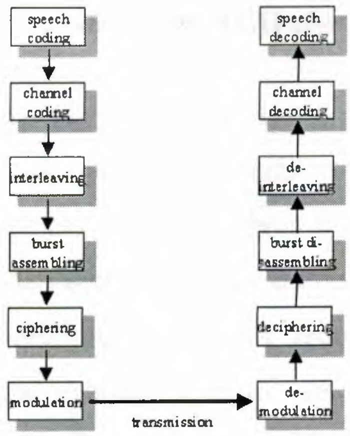

Figure 4.5 Different operations

4.7.1.Speech coding

The transmission of speech is, at the moment, the most important service of a mobile cellular system. The GSM speech codes, which will transform the analog signal (voice) into a digital representation, has to meet the following criteria:

- A good speech quality, at least as good as the one obtained with previous cellular systems.
- To reduce the redundancy in the sounds of the voice. This reduction is essential due to the limited capacity of transmission of a radio channel.
- The speech codes must not be very complex because complexity is equivalent to high costs.

The final choice for the GSM speech codes is a code named RPE-LTP (Regular Pulse Excitation Long-Term Prediction). This code uses the information from previous samples (this information does not change very quickly) in order to predict the current sample. The speech signal is divided into blocks of 20 ms. These blocks are then passed to the speech codes, which has a rate of 13 kbps, in order to obtain blocks of 260 bits.

4.7.2.Channel coding

Channel coding adds redundancy bits to the original information in order to detect and correct, if possible, errors occurred during the transmission.

4.7.2.1.Channel coding for the GSM data TCH channels

The channel coding is performed using two codes: a block code and a convolutional code. The block code corresponds to the block code defined in the GSM Recommendations 05.03. The block code receives an input block of 240 bits and adds four zero tail bits at the end of the input block. The output of the block code is consequently a block of 244 bits. A convolutional code adds redundancy bits in order to protect the information. A convolutional encoder contains memory. This property differentiates a convolutional code from a block code. A convolutional code can be defined by three variables: n, k and K. The value n corresponds to the number of bits at the output of the encoder, k to the number of bits at the input of the block and K to the memory of the encoder. The ratio, R, of the code is defined as follows: $R = k/n$.

Let's consider a convolutional code with the following values: k is equal to 1, n to 2 and K to 5. This convolutional code uses then a rate of $R = 1/2$ and a delay of $K = 5$, which means that it will add a redundant bit for each input bit. The convolutional code uses 5 consecutive bits in order to compute the redundancy bit. As the convolutional code is a 1/2 rate convolutional code, a block of 488 bits is generated. These 488 bits are punctured in order to produce a block of 456 bits. Thirty-two bits, obtained as follows, are not transmitted:

$C(11 + 15j)$ for $j = 0, 1... 31$

The block of 456 bits produced by the convolutional code is then passed to the interleave.

4.7.2.2.Channel coding for the GSM speech channels

Before applying the channel coding, the 260 bits of a GSM speech frame are divided in three different classes according to their function and importance. The most important class is the class 1a containing 50 bits. Next in importance is the class 1b, which contains 132 bits. The least important is the class II, which contains the remaining 78 bits. The different classes are coded differently. First of all, the class 1a bits are block-coded.

Three parity bits, used for error detection, are added to the 50 class 1a bits. The resultant 53 bits are added to the class 1b bits. Four zero bits are added to this block of 185 bits (50+3+132). A convolutional code, with r = 1/2 and K = 5, is then applied, obtaining an output block of 378 bits. The class II bits are added, without any protection, to the output block of the convolutional coder. An output block of 456 bits is finally obtained.

4.7.2.3.Channel coding for the GSM control channels

In GSM the signaling information is just contained in 184 bits. Forty parity bits, obtained using a fire code, and four zero bits are added to the 184 bits before applying the convolutional code (r = 1/2 and K = 5). The output of the convolutional code is then a block of 456 bits, which does not need to be punctured.

4.7.3. Interleaving

An interleaving rearranges 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.

4.7.3.1.Interleaving for the GSM control channels

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 interleave for control channels is called a block rectangular interleave.

4.7.3.2. 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 interleave for speech channels is called a block diagonal interleave.

4.7.3.3. 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:

- The first and the twenty-second bursts carry one block of 6 bits each
- The second and the twenty-first bursts carry one block of 12 bits each
- The third and the twentieth bursts carry one block of 18 bits each
- 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.

4.7.4. Burst assembling

The burst assembling procedure is in charge of grouping the bits into bursts.

4.7.5.Ciphering

Ciphering is used to protect signaling and user data. First of all, a ciphering key is computed using the algorithm A8 stored on the SIM card, the subscriber key and a random number delivered by the network (this random number is the same as the one used for the authentication procedure). Secondly, a 114-bit sequence is produced using the ciphering key, an algorithm called A5 and the burst numbers. This bit sequence is then XORed with the two 57 bit blocks of data included in a normal burst.

In order to decipher correctly, the receiver has to use the same algorithm A5 for the deciphering procedure.

4.7.6.Modulation

The modulation chosen for the GSM system is the Gauss Ian Modulation Shift Keying (GMSK). The aim of this section is not to describe precisely the GMSK modulation as it is too long and it implies the presentation of too many mathematical concepts. Therefore, only brief aspects of the GMSK modulation are presented in this section.

The GMSK modulation has been chosen as a compromise between spectrum efficiency, complexity and low spurious radiations (that reduce the possibilities of adjacent channel interference). The GMSK modulation has a rate of 270 5/6 k bauds and a BT product equal to 0.3. Figure 4.6 presents the principle of a GMSK modulator.

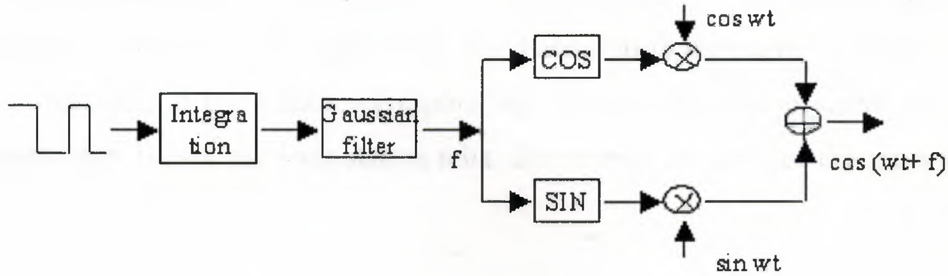

Figure 4.6: GMSK modulator

4.7.7.Discontinuous transmission (DTX)

This is another aspect of GSM that could have been included as one of the requirements of the GSM speech codes. The function of the DTX is to suspend the radio transmission during the silence periods. This can become quite interesting if we take into consideration the fact that a person speaks less than 40 or 50 percent during a conversation. The DTX helps then to reduce interference between different cells and to increase the capacity of the system. It also extends the life of a mobile's battery. The DTX function is performed thanks to two main features:

- The Voice Activity Detection (VAD), which has to determine whether the sound represents speech or noise, even if the background noise is very important. If the voice signal is considered as noise, the transmitter is turned off producing then, an unpleasant effect called clipping.
- The comfort noise. An inconvenient of the DTX function is that when the signal is considered as noise, the transmitter is turned off and therefore, a total silence is heard at the receiver. This can be very annoying to the user at the reception because it seems that the connection is dead. In order to overcome this problem, the receiver creates a minimum of background noise called comfort noise. The comfort noise eliminates the impression that the connection is dead.

4.7.8. Timing advance

The timing of the bursts transmissions is very important. Mobiles are at different distances from the base stations. Their delay depends, consequently, on their distance. The aim of the timing advance is that the signals coming from the different mobile stations arrive to the base station at the right time. The base station measures the timing delay of the mobile stations. If the bursts corresponding to a mobile station arrive too late and overlap with other bursts, the base station tells, this mobile, to advance the transmission of its bursts.

4.7.9. Power control

At the same time the base stations perform the timing measurements, they also perform measurements on the power level of the different mobile stations. These power levels are adjusted so that the power is nearly the same for each burst. A base station also controls its power level. The mobile station measures the strength and the quality of the signal between itself and the base station. If the mobile station does not receive correctly the signal, the base station changes its power level.

4.7.10.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.

4.7.11.Multipath and equalization

At the GSM frequency bands, radio waves reflect from buildings, cars, hills, etc. So not only the 'right' signal (the output signal of the emitter) is received by an antenna, but also many reflected signals, which corrupt the information, with different phases. An equalizer is in charge of extracting the 'right' signal from the received signal. It estimates the channel impulse response of the GSM system and then constructs an inverse filter. The receiver knows which training sequence it must wait for. The equalizers will then, comparing the received training sequence with the training sequence it was expecting, compute the coefficients of the channel impulse response. In order to extract the 'right' signal, the received signal is passed through the inverse filter.

CONCLUSION

The development of GSM is the first step towards a true personal communication system that will allow communication anywhere, anytime, and with anyone. The functional architecture of GSM, employing intelligent networking principles, and its ideology, which provides enough standardization to ensure compatibility, but still allows manufacturers and operators freedom, has been widely adopted in the development of future wireless systems, GSM is a very complex standard. It can be considered as the first serious attempt to fulfill the requirements for a universal personal communication system. GSM is then used as a basis for the development of the Universal Mobile Telecommunication System (UMTS).

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