NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

MOBILE PHONES AND BASE STATION

Graduation Project
EE- 400

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Nicosia-2002

ACKNOWLEDGEMENTS

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GSM (Global System for Mobile Communications) is a European digital communications standard which provides full duplex data traffic to any device fitted with GSM capability, such as a phone, fax, or pager, at a rate of 9600 bps using the TDMA communications scheme. Since GSM is purely digital, it can easily interface with other digital communications systems, such as ISDN, and digital devices, such as Group 3 facsimile machines.

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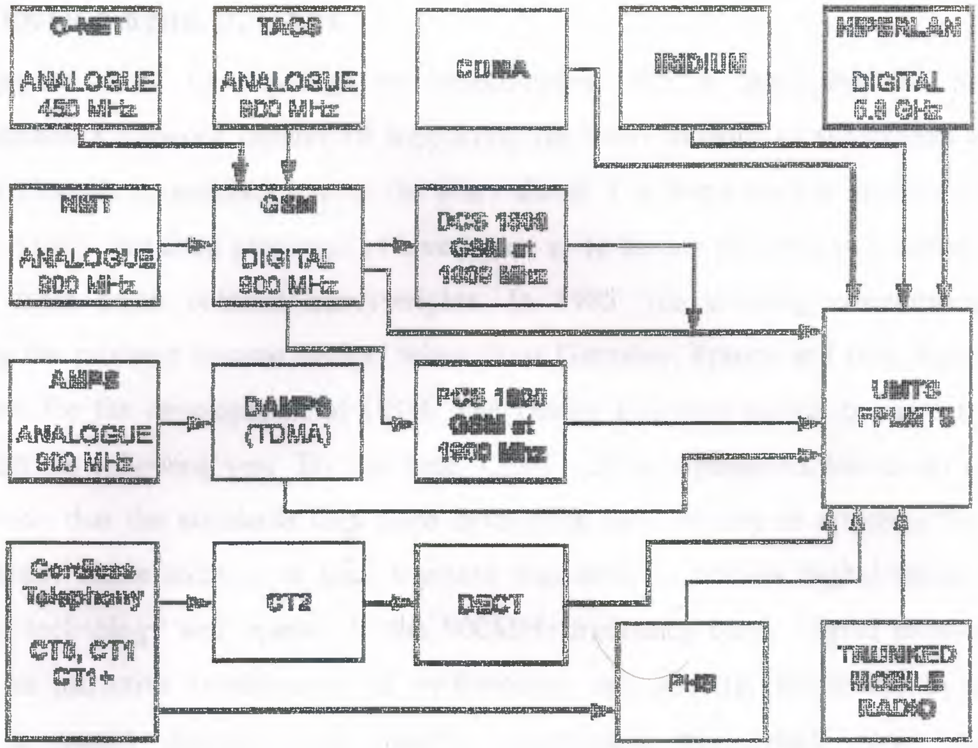

Figure 1.1 The Mobile Evolution

1.2 History Of GSM

During the early 1980s, analog cellular telephone systems were experiencing rapid growth in Europe, particularly in Scandinavia and the United Kingdom, but also in France and Germany. In the Nordic and Benelux countries the NMT 450 was developed, TACS in the UK and C-Netz in West Germany. The Radio com 2000 was in France and RTMI/RTMS in Italy. But each system was incompatible with everyone else's in equipment and operation and as business was becoming increasingly international, the cutting edge of the communications industry focused on exclusively local cellular solutions. These systems were fine if you wanted to call the office if you were in your own home, but not if you were with a client in another country. Also home market revenue simply wouldn't justify sustained programs of investment. As a solution in 1982 CEPT, the Conference des Administrations Europeans des Postes et Telecommunications comprised the telecom administrations of twenty-six European countries, established the Group Special Mobile (GSM).

1.2.1 Developments of GSM

Its objective was to develop the specification for a pan-European mobile communications network capable of supporting the many millions of subscribers likely to turn to mobile communications in the years ahead. The home market revenue simply wouldn't justify sustained programs of investment so to further progress they lobbied for support from some political heavyweights. In 1985, the growing commitment to resolving the problem became evident when West Germany, France and Italy signed an agreement for the development of GSM. The United Kingdom added its name to the agreement the following year. By this time, CEPTs Group Special Mobile could argue persuasively that the standards they were developing held the key to a technically and economically viable solution as their standard was likely to employ digital rather than analogue technology and operate in the 900MHz frequency band. Digital technology offered an attractive combination of performance and spectral efficiency. In other words, it would provide high quality transmission and enable more callers simultaneously to use the limited radio band available. In addition, such a system would allow the development of advanced features like speech security and data communications. Handsets could be cheaper and smaller. It would also make it possible to introduce the first hand-held terminals - even though in the early days in terms of size and weight these would be practically indistinguishable from a brick. Finally, the digital approach neatly complemented the Integrated Services Digital Network (ISDN), which was being developed by land-based telecommunications systems throughout the world. But the frequencies to be employed by the new standard were being snapped up by the analogue networks. Over-capacity crisis had started to sound alarm bells throughout the European Community. Demand was beginning to outstrip even the most optimistic projections. The Group Special Mobile's advocacy of digital cellular technology was on hand to offer light at the end of the tunnel. The Directive ensured that every Member State would reserve the 900MHz frequency blocks required for the rollout program. Although these were somewhat smaller than the amount advocated by the CEPT, the industry had finally achieved the political support it needed to advance its objectives. The logistical nightmare in the GSM, which followed soon left this achievement as a distant, dream so single, permanent organization at the helm.

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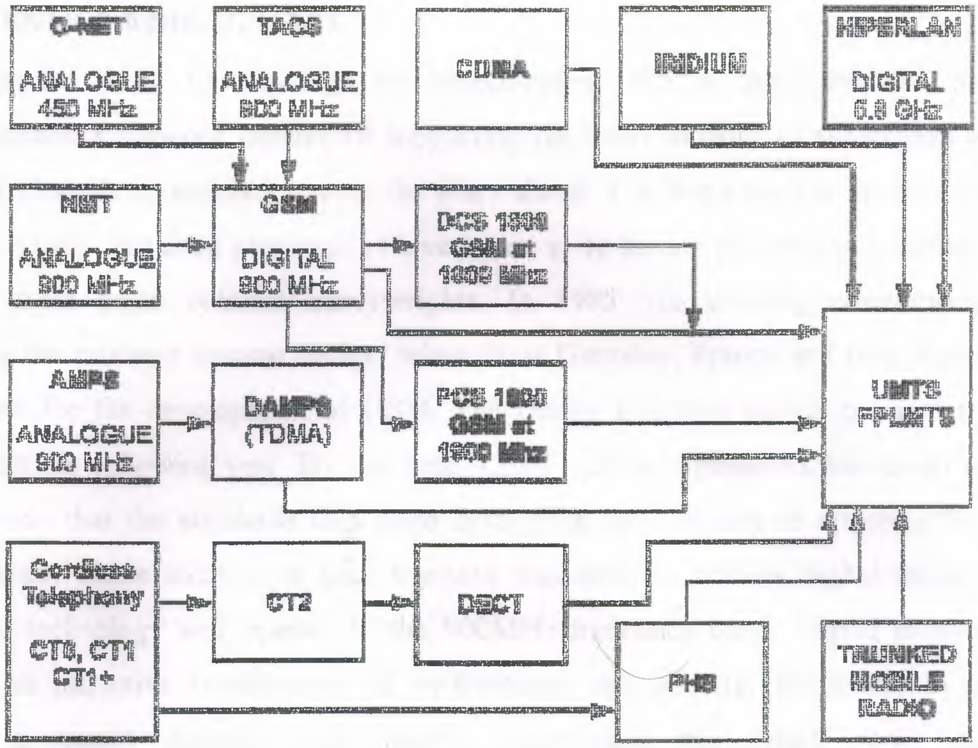

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In1986 the GSM Permanent Nucleus was formed and its head quarters established in Paris. It was all very well agreeing the technology and standards for this new product. But what about the creation of a market? It was essential to forge a commercial agreement between potential operators who would commit themselves to implementing the standard by a particular date. Without such an agreement there could be no network. Without the network there would be no terminals. Without network and terminals there would be no service. Stephen Temple of the UK's Department of Trade and Industry was charged with the task of drafting the first Memorandum of Understanding (MOU). In September 1987 network operators from thirteen countries signed a MOU in Copenhagen. One of the most important conclusions drawn from the early tests was that the new standard should employ Time Division Multiple Access (TDMA) technology. The strength of its technical performance ensured that narrowband TDMA had the support of major players like Nokia, Ericsson and Siemens. This promised the flexibility inherent in having access to a broad range of suppliers and the potential to get product faster into the marketplace. But as always as soon as one problem was solved other problems looming on the horizon.

In 1989, the UK Department of Trade and Industry published a discussion document called “Phones on the Move”. This advocated the introduction of mass-market mobile communications using new technology and operating in the 1800 MHz frequency band. The UK government licensed two operators to run what became known as Personal Communications Networks (PCN). Operating at the higher frequency gave the PCN operators virtually unlimited capacity, where as 900MHz was limited. The next hurdle to over come was that of the deadline. If the 1 July 1991 launch date was not met there was a real danger that confidence in GSM technology would be fatally undermined but moral received a boost when in 1989 the responsibility for specification development passed from the GSM Permanent Nucleus to the newly created European Telecommunications Standards Institute (ETSI). In addition, the UK’s PCN turned out to be more of an opportunity than a threat. The new operators decided to utilize the GSM specification - slightly modified because of the higher frequency - and the development of what became known as DCS 1800 was carried out by ETSI in parallel with GSM standardization. In fact, in 1997 DCS 1800 was renamed GSM 1800 to reflect the affinity between the two technologies. With so many manufacturers creating so many products in so many countries, it soon became apparent that it was critical that each type of terminal was subject to a rigorous approval regime. Rogue terminals could cause untold damage to the new networks. The solution was the introduction of Interim Type Approval (ITA). Essentially, this was a procedure in which only a subset of the approval parameters was tested to ensure that the terminal in question would not create any problems for the networks. In spite of considerable concern expressed by some operators, ITA terminals became widely available in the course of 1992. True hand held terminals hit the market at the end of that year and the GSM bandwagon had finally started to roll. From here the G.S.M became a success story. In 1987, the first of what was to become an annual event devoted to the worldwide promotion of GSM technology was staged by conference organizers IBC Technical Services. The Pan European Digital Cellular Conference. This year it celebrated its tenth anniversary in Cannes, attracting over 2,400 delegates. By the end of 1993, GSM had broken through the 1 million-subscriber barrier with the next million already on the horizon. By June 1995 Phase 2 of standardization came in to play and a demonstration of fax, video and data communication via GSM. When the GSM standard was being drawn up by the CEPT, six separate systems were all considered as the base. There were seven criteria deemed to be of importance when assessing which of the six would be used. Each country developed its own system, which was incompatible with everyone else's in equipment and operation. This was an undesirable situation, because not only was the mobile equipment limited to operation within national boundaries, which in a unified Europe were increasingly unimportant, but there was also a very limited market for each type of equipment, so economies of scale and the subsequent savings could not be realized. The Europeans realized this early on, and in 1982 the Conference of European Posts and Telegraphs (CEPT) formed a study group called the Group Special Mobile (GSM) to study and develop a pan-European public land mobile system. The proposed system had to meet certain criteria. In 1989, GSM responsibility was transferred to the European Telecommunication Standards Institute (ETSI), and phase-I of the GSM specifications were published in 1990. Commercial service was started in mid-1991, and by 1993 there were 36 GSM networks in 22 countries with 25 additional countries having already selected or considering GSM. This is not only a European standard - South Africa, Australia, and many Middle and Far East countries have chosen GSM. Although standardized in Europe, GSM is not only a European standard. Over 200

GSM networks (including DCS1800 and PCS1900) are operational in 110 countries around the world. In the beginning of 1994, there were 1.3 million subscribers worldwide, which had grown to more than 55 million by October 1997. With North America making a delayed entry into the GSM field with a derivative of GSM called PCS1900, GSM systems exist on every continent, and the acronym GSM now aptly stands for Global System for Mobile communications. The developers of GSM chose an unproven (at the time) digital system, as opposed to the then-standard analog cellular systems like AMPS in the United States and TACS in the United Kingdom. They had faith that advancements in compression algorithms and digital signal processors would allow the fulfillment of the original criteria and the continual improvement of the system in terms of quality and cost. The over 8000 pages of GSM recommendations try to allow flexibility and competitive innovation among suppliers, but provide enough standardization to guarantee proper inter-working between the components of the system. This is done by providing functional and interface descriptions for each of the functional entities defined in the system. The development of GSM started in 1982, when the Conference of European Posts and Telegraphs (CEPT) formed a study group called Group Special Mobile (the initial meaning of GSM). The group was to study and develop a pan-European public cellular system in the 900 MHz range, using spectrum that had been previously allocated. At that time, there were many incompatible analog cellular systems in various European countries. Some of the basic criteria for their proposed system were:

- Good subjective speech quality.
- Low terminal and service cost.
- Support for international roaming.
- Ability to support handheld terminals.
- Support for range of new services and facilities.
- Spectral efficiency.
- ISDN compatibility.

In 1989, the responsibility for GSM was transferred to the European Telecommunication Standards Institute (ETSI), and the Phase I recommendations were published in 1990. At that time, the United Kingdom requested a specification based on GSM but for higher user densities with low-power mobile stations, and operating at 1.8

GHz. The specifications for this system, called Digital Cellular System (DCS1800) were published 1991. Commercial operation of GSM networks started in mid-1991 in European countries. By the beginning of 1995, there were 60 countries with operational or planned GSM networks in Europe, the Middle East, the Far East, Australia, Africa, and South America, with a total of over 5.4 million subscribers. As it turned out, none of the six candidates was actually used! The information collected during the tests did enable the GSM (Group Special Mobile) to design the specifications of the current GSM network. The total change to a digital network was one of the fundamental factors of the success of GSM. Digital transmission is easier to decode than analogue due to the limited number of possible input values (0,1), and as ISDN was becoming de facto at the time, it was logical to avail of digital technology. This also ensured that GSM could evolve properly in an increasingly digital world, for example with the introduction of an 8kps speech coder. It is much easier to change channel characteristics digitally than analogously. Finally, the transmission method decided on for the network was TDMA, as opposed to FDMA and CDMA. In 1989, responsibility for the specification was passed from CEPT to the newly formed and now famous European Telecommunications Standards Institute (ETSI). By 1990, the specifications and explanatory notes on the system were documented extensively, producing 138 documents in total, some reaching sizes of several hundred pages in length services.

1.3 Technology

1.3.1 Services Provided by GSM

From the beginning, the planners of GSM wanted ISDN compatibility in terms of the services offered and the control signaling used. However, radio transmission limitations, in terms of bandwidth and cost, do not allow the standard ISDN B-channel bit rate of 64 kbps to be practically achieved. Using the ITU-T definitions, telecommunication services can be divided into bearer services, tele-services, and supplementary services. The digital nature of GSM allows data, both synchronous and asynchronous, to be transported as a bearer service to or from an ISDN terminal. Data can use either the transparent service, which has a fixed delay but no guarantee of data integrity, or a non-transparent service, which guarantees data integrity through an Automatic Repeat Request (ARQ) mechanism, but with a variable delay. The data rates supported by

GSM are 300 bps, 600 bps, 1200 bps, 2400 bps, and 9600 bps. The most basic tele-service supported by GSM is telephony. As with all other communications, speech is digitally encoded and transmitted through the GSM network as a digital stream. There is also an emergency service, where the nearest emergency-service provider is notified by dialing three digits (similar to 911). A variety of data services is offered. GSM users can send and receive data, at rates up to 9600 bps, to users on POTS (Plain Old Telephone Service), ISDN, Packet Switched Public Data Networks, and Circuit Switched Public Data Networks using a variety of access methods and protocols, such as X.25 or X.32. Since GSM is a digital network, a modem is not required between the user and GSM network, although an audio modem is required inside the GSM. Network to inter-work with POTS. Other data services include Group 3 facsimile, as described in ITU-T recommendation T.30, which is supported by use of an appropriate fax adaptor. A unique feature of GSM, not found in older analog systems, is the Short Message Service (SMS). SMS is a bi directional service for short alphanumeric (up to 160 bytes) messages. Messages are transported in a store-and-forward fashion. For point-to-point SMS, a message can be sent to another subscriber to the service, and an acknowledgement of receipt is provided to the sender. SMS can also be used in a cell-broadcast mode, for sending messages such as traffic updates or news updates. Messages can also be stored in the SIM card for later retrieval supplementary services are provided on top of tele-services or bearer services. In the current (Phase I) specifications, they include several forms of call forward (such as call forwarding when the mobile subscriber is unreachable by the network), and call barring of outgoing or incoming calls, for example when roaming in another country. Many additional supplementary services will be provided in the Phase 2 specifications, such as caller identification, call waiting, multi-party conversations. GSM was designed having interoperability with ISDN in mind, and the services provided by GSM are a subset of the standard ISDN services. Speech is the most basic, and most important, tele-service provided by GSM. In addition, various data services are supported, with user bit rates up to 9600 bps. Specially equipped GSM terminals can connect with PSTN, ISDN, Packet Switched and Circuit Switched Public Data Networks, through several possible methods, using synchronous or asynchronous transmission. Also supported are Group 3 facsimile service, video-tex, and teletex. Other GSM services include a cell broadcast service, where messages such as traffic reports, are broadcast to users in particular cells. A service unique to GSM, the Short Message Service, allows users to send and receive point-to-point alphanumeric messages up to a few tens of bytes. It is similar to paging services, but much more comprehensive, allowing bi-directional messages, store-and-forward delivery, and acknowledgement of successful delivery.

1.4 The Different GSM-Based Networks

Different frequency bands are used for GSM 900, GSM1800 and GSM 1900 (Table 1.3.). 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.

Table 1.3 Frequency Bands For The Different GSM-Based Networks

Network type	Frequency band UL / DL	Implementations
GSM 900	890-915 / 935-960 MHz	GSM 900
GSM1800	1710 – 1785 / 1805 -1880 MHz	GSM 1800
GSM1900	1850-1910 / 1930-1990 MHz	GSM1900

1.4.1 Where are GSM Frequencies Used?

GSM networks presently operate in three different frequency ranges. These are:

a) GSM 900

(Also called GSM) operates in the 900 MHz frequency range and is the most common in Europe and the world.

b) GSM 1800

(Also called PCN (Personal Communication Network), and DCS 1800) - operates in the 1800 MHz frequency range and is found in a rapidly-increasing number of countries including France, Germany, Switzerland, the UK, and Russia. A European Commission mandate requires European Union members to license at least one DCS 1800 operator before 1998.

c) GSM 1900

(Also called PCS (Personal Communication Services), PCS 1900, and DCS 1900) - the only frequency used in the United States and Canada for GSM. Note that the terms PCS is commonly used to refer to any digital cellular network operating in the1900 MHz frequency range, not just GSM.

2. GSM RADIO INTERFACE

2.1 Overview

The Radio interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces of the GSM system. One of the main objectives of GSM is roaming. Therefore, in order to obtain a complete compatibility between mobile stations and networks of different manufacturers and operators, the radio interface must be completely defined. The spectrum efficiency depends on the radio interface and the transmission, more particularly in aspects such as the capacity of the system and the techniques used in order to decrease the interference and to improve the frequency reuse scheme. The specification of the radio interface has then an important influence on the spectrum efficiency.

2.2 Frequency Allocation

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

- 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).

These bands were allocated by the ITU (International Telecom Union) who are responsible for allocating radio spectrum on an international basis. Although these bands were (and still are) used by analog systems in the early 1980's, the top 10 MHz were reserved for the already emerging GSM Network by the CEPT (European Conference of Posts and Telecommunications: translated from French). But not all the countries can use the whole GSM frequency bands. This is due principally to military reasons and to the existence of previous analog systems using part of the two 25 Mhz frequency bands.

2.3 Multiple Access Scheme

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

It is hoped that eventually the GSM network will use the entire bandwidth. It is apparent from this that the bandwidth you use on a day-to-day basis to operate your mobile phone is limited. It would seem that only a certain number of users can operate on the bandwidth simultaneously. However GSM has devised a method to maximize the bandwidth available. They use a combination of Time and Frequency Division Multiple Access (TDMA/FDMA).

a) **FDMA:** 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.

This is the division of the bandwidth in to 124 carrier frequencies each of 200 kHz. At least one of these is assigned to each base station. Figure 2.1 shows the FDMA System.

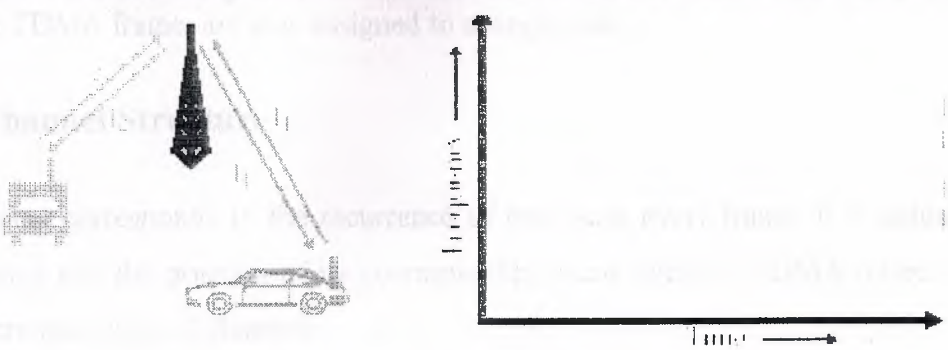


Figure2.1 Frequency Division Multiple Access

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

The carrier frequencies are then divided again into 8 time slots. This prevents mobiles from transmitting and receiving calls at the same time as they are allocated separate time slots. Figure 2.2 shows Time Division Multiple Access System.

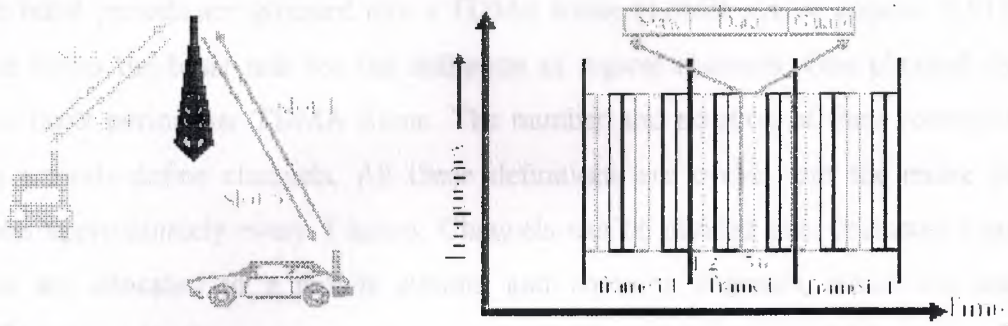


Figure 2.2 Time Division Multiple Access

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.

2.4 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.

Since radio spectrum is a limited resource shared by all users, a method must be devised to divide up the bandwidth among as many users as possible. The method chosen by GSM is a combination of Time- and Frequency-Division Multiple Access (TDMA/FDMA). The FDMA part involves the division by frequency of the (maximum) 25 MHz bandwidth into 124 carrier frequencies spaced 200 kHz apart. One or more carrier frequencies are assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme. The fundamental unit of time in this TDMA scheme is called a burst period and it lasts 15/26 ms (or approx. 0.577 ms). Eight burst periods are grouped into a TDMA frame (120/26 ms, or approx. 4.615 ms), which forms the basic unit for the definition of logical channels. One physical channel is one burst period per TDMA frame. The number and position of their corresponding burst periods define channels. All these definitions are cyclic, and the entire pattern repeats approximately every 3 hours. Channels can be divided into dedicated channels, which are allocated to a mobile station, and common channels, which are used by mobile stations in idle mode.

2.4.1 Traffic Channels

A traffic channel (TCH) is used to carry speech and data traffic. Traffic channels are defined using a 26-frame multi frame, or group of 26 TDMA frames. The length of a 26-frame multi frame is 120 ms, which is how the length of a burst period is defined (120 ms divided by 26 frames divided by 8 burst periods per frame). Out of the 26 frames, 24 are used for traffic, 1 is used for the Slow Associated Control Channel (SACCH) and 1 is currently unused (see Figure 3.1). TCHs for the uplink and downlink are separated in time by 3 burst periods, so that the mobile station does not have to transmit and receive simultaneously, thus simplifying the electronics. In addition to these full-rate TCHs, there are also half-rate TCHs defined, although they are not yet implemented.

Half-rate TCHs will effectively double the capacity of a system once half-rate speech coders are specified (i.e., speech coding at around 7 kbps, instead of 13 kbps). Eighth-rate TCHs are also specified, and are used for signaling. In the recommendations, they are called Stand-alone Dedicated Control Channels (SDCCH). 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.

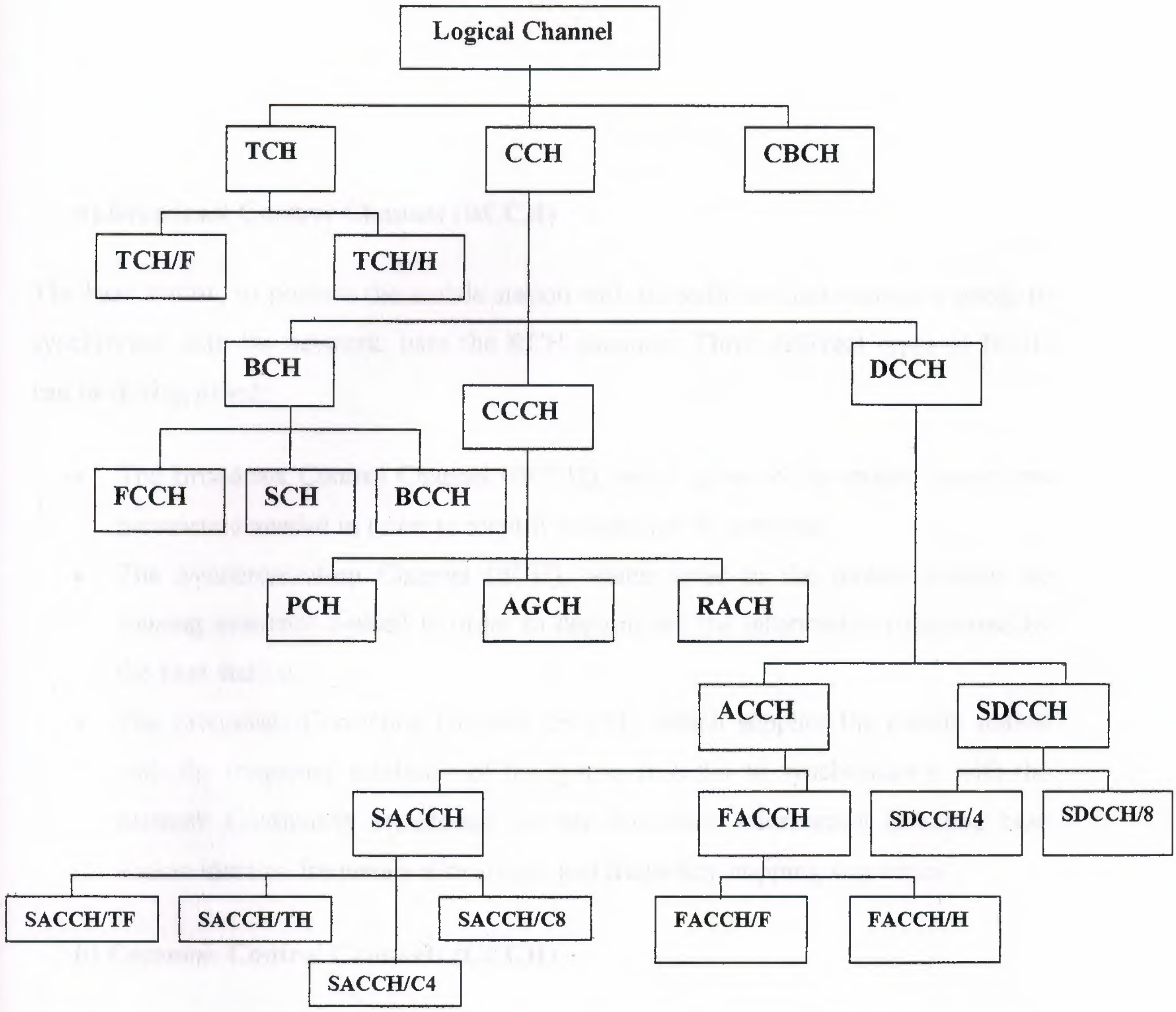
Half-rate traffic channels (TCH/H), which double the capacity of the system, are also grouped in a 26-Multiframe but the internal structure is different.

2.4.2 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.

Common channels can be accessed both by idle mode and dedicated mode mobiles. Idle mode mobiles to exchange the signalling information required to change to dedicated mode use the common channels. Mobiles already in dedicated mode monitor the surrounding base stations for handover and other information. The common channels are defined within a 51-frame multiframe, so that dedicated mobiles using the 26-frame multiframe TCH structure can still monitor control channels. Figure 2.3 shows the Logical channels. The common channels include:



TCH: Traffic Channel.
TCH/F: Traffic Channel/Full.
TCH/H: Traffic Channel/Half.
CCH: Control Channel.
BCH:Broadcast Channel.
CBCH:Cell Broadcast Channel.
CCCH: Common Control Channel.
ACCH: Associated Control Channel.
SACCH: Slow Associated Control Channel.
FACCH: Fast Associated Control Channel.
SDCCH: Stand-Alone Ded icated Control Channel.

FCCH: Freq. Correction Channel.
SCH: Synchronization Channel.
BCCH: Broadcast Control Channel.
PCH: Paging Channel.
AGCH: Access Grant Channel.
RACH: Random Access Channel.
DCCH: Dedicated Control Channel.

Figure 2.3 Structure of Logical Channels

a) Broadcast Control Channel (BCCH)

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 Continually broadcasts, on the downlink, information including base station identity, frequency allocations, and frequency-hopping sequences.

b) 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 call.
- 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.

c) Frequency Correction Channel (FCCH) and Synchronization Channel (SCH)

Used to synchronize the mobile to the time slot structure of a cell by defining the boundaries of burst periods, and the time slot numbering. Every cell in a GSM network broadcasts exactly one FCCH and one SCH, which are by definition on time slot number 0 (within a TDMA frame).

d) 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.

e) 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.

f) Random Access Channel (RACH)

Slotted Aloha channel used by the mobile to request access to the network.

g) Paging Channel (PCH)

Used to alert the mobile station of an incoming call.

h) Access Grant Channel (AGCH)

Used to allocate an SDCCH to a mobile for signaling (in order to obtain a dedicated channel), following a request on the RACH.

2.4.3 Burst Structure

There are four different types of bursts used for transmission in GSM. The normal burst is used to carry data and most signaling. It has a total length of 156.25 bits, made up of two 57 bit information bits, a 26 bit training sequence used for equalization, 1 stealing bit for each information block (used for FACCH), 3 tail bits at each end, and an 8.25 bit guard sequence, as shown in Figure2.4. The 156.25 bits are transmitted in 0.577 ms, giving a gross bit rate of 270.833 kbps. The F burst, used on the FCCH, and the S burst, used on the SCH, have the same length as a normal burst, but a different internal structure, which differentiates them from normal bursts (thus allowing synchronization). The access burst is shorter than the normal burst, and is used only on the RACH. 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.

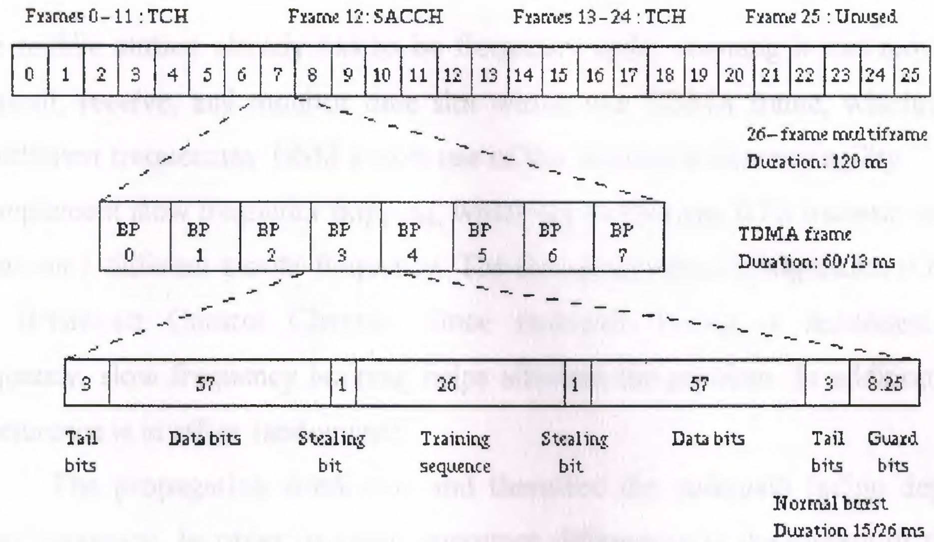


Figure 2.4 Structure of the 26-Multiframe, the TDMA frame and the normal burst

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.

2.4.4 Frequency Hopping

The mobile station already has to be frequency agile, meaning it can move between a transmit, receive, and monitor time slot within one TDMA frame, which normally are on different frequencies. GSM makes use of this inherent frequency agility to implement slow frequency hopping, where the mobile and BTS 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 helps alleviate the problem. In addition, co-channel interference is in effect randomized.

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.

2.5 From source information to radio waves

The figure 2.5 presents the different operations that have to be performed in order to pass from the speech source to radio waves and vice versa. If the source of information is data and not speech, the speech coding will not be performed.

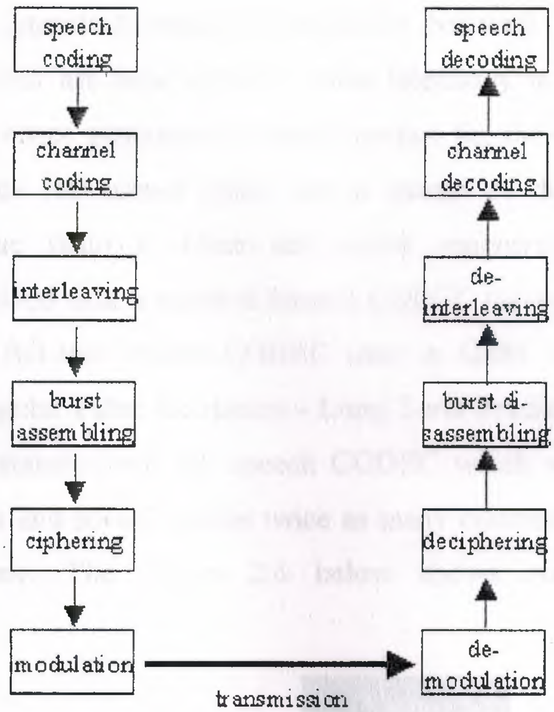


Figure 2.5 From Speech Source To Radio Waves

2.5.1 Speech Coding

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

- 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 coder must not be very complex because complexity is equivalent to high costs.

The final choice for the GSM speech coder is a coder named RPE-LTP (Regular Pulse Excitation Long-Term Prediction). This coder 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 coder, which has a rate of 13 kbps, in order to obtain blocks of 260 bits. Obviously the most important aspect of the GSM Network is speech transmission. Although other services are now offered, voice telephony is still the most popular service available and hence generates the most revenue for the various companies. The device that transforms the human voice into a stream of digital data, suitable for transmission over the radio interface and which regenerates an audible analog representation of received data is called a Speech CODEC (speech transcoder or speech coder/decoder). The full-rate speech CODEC used in GSM is known as RPE-LTP, which stands for "Regular Pulse Excitation - Long Term Prediction". It is hoped there will eventually be a standardized full speech CODEC which will half the amount of data to be transmitted and so will enable twice as many customers to use the same slot in the TDMA frame. The Figure 2.6 below shows audio signal processing

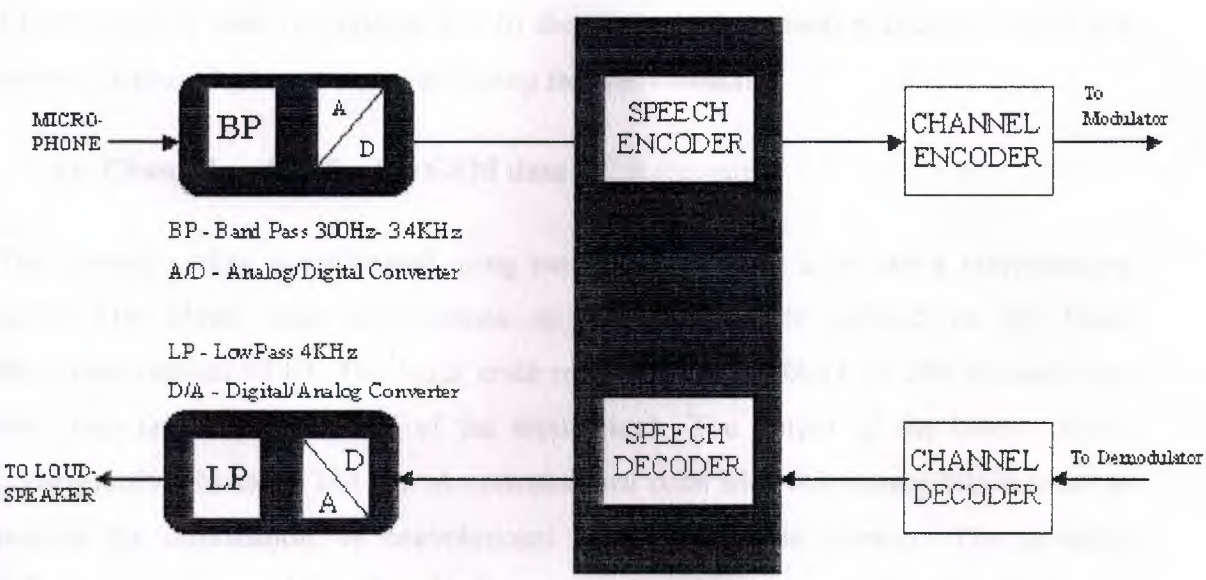


Figure 2.6 Audio Signal Processing

GSM is a digital system, so speech which is inherently analog, has to be digitized. The method employed by ISDN, and by current telephone systems for multiplexing voice lines over high-speed trunks and optical fiber lines, is Pulse Coded Modulation (PCM). The output stream from PCM is 64 kbps, too high a rate to be feasible over a radio link. The 64 kbps signal, although simple to implement, contains much redundancy. The GSM group studied several speech coding algorithms on the basis of subjective speech quality and complexity (which is related to cost, processing delay, and power consumption once implemented) before arriving at the choice of a Regular Pulse Excited Linear Predictive Coder (RPE-LPC) with a Long Term Predictor loop. Basically, information from previous samples, which does not change.

Very quickly, is used to predict the current sample. The coefficients of the linear combination of the previous samples, plus an encoded form of the residual, the difference between the predicted and actual sample, represent the signal. Speech is divided into 20 millisecond samples, each of which is encoded as 260 bits, giving a total bit rate of 13 kbps. This is the so-called Full-Rate speech coding. Recently, some North American GSM1900 operators have implemented an Enhanced Full-Rate (EFR) speech-coding algorithm. This is said to provide improved speech quality using the existing 13 kbps bit rate.

2.5.2 Channel coding

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

a) 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:

$$C\,(11 + 15\,j) \text{ for } j = 0, 1, \ldots, 31 \qquad (2.1)$$

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

b) 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 Ia containing 50 bits. Next in importance is the class Ib, 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 Ia bits are block-coded. Three parity bits, used for error detection, are added to the 50 class Ia bits. The resultant 53 bits are added to the class Ib 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.

c) 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. Electromagnetic interference can disrupt encoded speech and data transmitted over the GSM Network. Because of this this complicated encoding and block interleaving is used to protect the Network. Speech and data rates use different algorithms. Radio emissions too can cause interference if they occur outside of the allotted bandwidth and must be strictly controlled to allow for both GSM and older analog systems to co-exist. Because of natural and man-made electromagnetic interference, the encoded speech or data signal transmitted over the radio interface must be protected from errors. GSM uses convolutional encoding and block interleaving to achieve this protection. The exact algorithms used differ for speech and for different data rates. The method used for speech blocks will be described below. Recall that the speech coder produces a 260-bit block for every 20 ms speech sample. From subjective testing, it was found that some bits of this block were more important for perceived speech quality than others. The bits are thus divided into three classes:

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

Class Ia bits have a 3 bit Cyclic Redundancy Code added for error detection. If an error is detected, the frame is judged too damaged to be comprehensible and it is discarded. It is replaced by a slightly attenuated version of the previous correctly received frame. These 53 bits, together with the 132 Class Ib bits and a 4-bit tail sequence (a total of 189 bits), are input into a 1/2 rate convolutional encoder of constraint length 4. Each input bit is encoded as two output bits, based on a combination of the previous 4 input bits. The convolutional encoder thus outputs 378 bits, to which are added the 78 remaining Class II bits, which are unprotected. Thus every 20 ms speech sample is encoded as 456 bits, giving a bit rate of 22.8 kbps. To further protect against the burst errors common to the radio interface, each sample is interleaved. The 456 bits output by the convolutional encoder are divided into 8 blocks of 57 bits, and these blocks are transmitted in eight consecutive time-slot bursts. Since each time-slot burst can carry two 57-bit blocks, each burst carries traffic from two different speech samples. Recall that each time-slot burst is transmitted at a gross bit rate of 270.833 kbps. This digital signal is modulated onto the analog carrier frequency using Gaussian-filtered Minimum Shift Keying (GMSK). GMSK was selected over other modulation schemes as a compromise between spectral efficiency, complexity of the transmitter, and limited spurious emissions. The complexity of the transmitter is related to power consumption, which should be minimized for the mobile station. The spurious radio emissions, outside of the allotted bandwidth, must be strictly controlled so as to limit adjacent channel interference, and allow for the co-existence of GSM and the older analog systems (at least for the time being).

2.5.3 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.

a) 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 (4x114 = 456). The 456 bits are divided into eight blocks of 57 bits. The first block of 57 bits contains the bit numbers (0, 8, 16,....., 448), the second one the bit numbers (1, 9, 17, ...,449), etc. The last block of 57 bits will then contain the bit numbers (7, 15,....., 455). The first four blocks of 57 bits are placed in the even-numbered bits of four bursts. The other four blocks of 57 bits are placed in the odd-numbered bits of the same four bursts. Therefore the interleaving depth of the GSM interleaving for control channels is four and a new data block starts every four bursts. The interleaver for control channels is called a block rectangular interleaver.

b) Interleaving for the GSM speech Channels

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

c) 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.

2.5.4 Burst Assembling

The burst assembling procedure is in charge of grouping the bits into bursts. Section 2.4.3. presents the different bursts structures and describes in detail the structure of the normal burst.

2.5.5 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.

2.5.6 Modulation

The modulation chosen for the GSM system is the Gaussian Minimum 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 kbauds and a BT product equal to 0.3. Figure 2.7. presents the principle of a GMSK modulator.

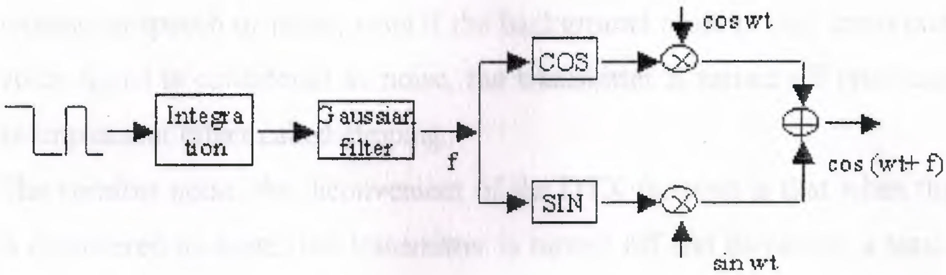


Figure 2.7 GMSK Modulator

2.6 Discontinuous Transmission (DTX)

Minimizing co-channel interference is a goal in any cellular system, since it allows better service for a given cell size, or the use of smaller cells, thus increasing the overall capacity of the system. Discontinuous Transmission (DTX) is a method that takes advantage of the fact that a person speaks less that 40 percent of the time in normal conversation, by turning the transmitter off during silence periods. An added benefit of DTX is that power is conserved at the mobile unit. The most important component of DTX is, of course, Voice Activity Detection (VAD). It must distinguish between voice and noise inputs, a task that is not as trivial as it appears, considering background noise. If a voice signal is misinterpreted as noise, the transmitter is turned off and a very annoying effect called clipping is heard at the receiving end. If, on the

other hand, noise is misinterpreted as a voice signal too often, the efficiency of DTX is dramatically
decreased. Another factor to consider is that when the transmitter is turned off, there is total silence heard at the receiving end, due to the digital nature of GSM. To assure the receiver that the connection is not dead, comfort noise is created at the receiving end by trying to match the characteristics of the transmitting end's background noise. This is another aspect of GSM that could have been included as one of the requirements of the GSM speech coder. 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.

2.7 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.

2.8 Power Control

There are five classes of mobile stations defined, according to their peak transmitter power, rated at 20, 8, 5, 2, and 0.8 watts. To minimize co-channel interference and to conserve power, both the mobiles and the Base Transceiver Stations operate at the lowest power level that will maintain an acceptable signal quality. Power levels can be stepped up or down in steps of 2 dB from the peak power for the class down to a minimum of 13 dBm (20 milli watts). The mobile station measures the signal strength or signal quality (based on the Bit Error Ratio), and passes the information to the Base Station Controller, which ultimately decides if and when the power level should be changed. Power control should be handled carefully, since there is the possibility of instability. This arises from having mobiles in co-channel cells alternatingly increase their power in response to increased co-channel interference caused by the other mobile increasing its power. This in unlikely to occur in practice but it is (or was as of 1991) under study. 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.

2.9 Discontinuous Reception

Another method used to conserve power at the mobile station is discontinuous reception. The paging channel, used by the base station to signal an incoming call, is structured into sub-channels. Each mobile station needs to listen only to its own sub-channel. In the time between successive paging sub-channels, the mobile can go into sleep mode, when almost no power is used. 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.

2.10 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 equalizer 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. At the 900 MHz range, radio waves bounce off everything - buildings, hills, cars, airplanes, etc. Thus many reflected signals, each with a different phase, can reach an antenna. Equalization is used to extract the desired signal from the unwanted reflections. It works by finding out how a known transmitted signal is modified by multipath fading, and constructing an inverse filter to extract the rest of the desired signal. This known signal is the 26-bit training sequence transmitted in the middle of every time-slot burst. The actual implementation of the equalizer is not specified in the GSM specifications.

3. MOBILE PHONES

3.1 Overview

Mobile phones may be thought of as cordless phones with elaborate portable and base units. High-power transmitters and elavated antennas that provide the radio carrier link over an area within 20 to 30 miles from the base station antenna, as well as the multiplexing, detecting, sorting and selecting features required to simultaneously service 60 subscribers per base station, are the major differences between cordless phones and mobile phones.

3.2 Base Unit

The base station can transmit and receive on several different frequencies simultaneously to provide several individual channels for use at the same time. The radio base station transmitter output power is typically 200-250 watts and the radiated power can be as high as 500 watts if the transmitting antenna gain is included. It covers a circular area of up to 30 milles in radius for clear reliable communications, but transmitters with the same frequence are not spacet closer than about 60 to 100 milles because of the noise interfrence levels.

The receiver contains filters, high-gain amplifiers, and demodulators to provide a usable voice signal to the phone line. The control terminal contains the necessary detector and timing and logic circuits to control the transmission link between the base unit and the mobile units. As a result, phone calls are coupled to and from the standard phone system just like calls that are carried completely over wired facilities. The control terminal has the necessary interface circuits so that a call initiated at a mobile unit is interconnected through the national or international phone system to the called party just as any other phone call.

The national and international phone system facilities are owned by the respective phone companies. The base units and mobile units may be owned by the phone company or by a separate company called a radio common carrier (RCC). When the mobile system is run by a RCC, the RCC is charged by the telephone company for the use of the standard phone system just like any other customer.

The cost is then included in the charge by the RCC to the eventual user of the mobile units.

To subscribe to mobile phone service, a user has only to apply, and be accepted by the RCC or the phone company operating the system. When the application is accepted, the user can lease or purchase the mobile equipment.

3.3 Mobile Unit

The mobile unit in the user's vehicle consists of a receiver containing amplifiers, a mixer and a demodulator; a transmitter containing a modulator, carrier oscillators and amplifiers; the necessary control logic; a control unit with microphone, speaker, keypad and switches; antennas and the interconnecting cables. The control unit performs all of the functions associated with normal phone use. A modern control head with automatic functions is illustrated.

The mobile phone user with automatic control places and receives calls in the same manner as with an oridinary phone. When the handset is lifted to place a call, the radio unit automatically selects an available channel. If no channel is available, the busy light comes on, If a channel is found, the user hears the normal dial tone from the phone system, and can then dial the number and proceed as if the phone were direct wired. An incoming call to the mobile unit is signaled by a ringing tone and is answered simply by lifting the handset and talking. Thus, the automatic mobile phone is as easily used as a phone. The mobile phone combines the mobility of the radio link and the world-wide switched network of the existing phone system to provide a communication link to any other phone in the world.

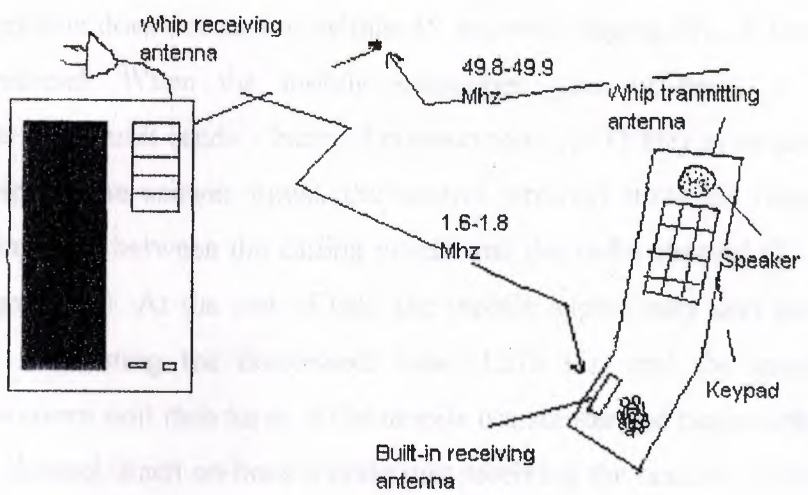


Figure 3.1 Mobile Unit

3.4 Detailed Operation

Different signalling techniques have to used in a mobile phone system in contrast with a wired facility. Since there are no wires connecting the telephone to the network, both speech and signalling must be transmitted via radio. For wireless operation, tones are used for those signaling functions, which are otherwise performed by voltage and current in hard-wired systems. This is accomplished by the use of special tones rather than applying a voltage level or detecting a current. The proper tone transmitted to the mobile unit will, for example, ring the mobile phone to indicate an incoming call just as with a standard phone different tone is used to indicate off-hook, busy, etc. The Improved Mobile Telephone System (IMTS) uses in band signalling tones from 1300 Hz to 2200 Hz. The older Mobile Phone System (MTS) had in band signalling tones in the 600 HZ to 1500 Hz range. Some systems use 2805 Hz as manual operation.

channel. When the mobile unit receives its correct seven-digit address, the mobile supervisory unit turns on the mobile transmitter and sends the acknovledgement signal Ack (5), using the 2150 Hz guard-tone, back to the control terminal. If this acknowledgement is not received by the control terminal within 3 seconds after out-pulsing the address, seize tone is removed and the call is abandoned. However, upon receipt of the mobile acknowledgement signal, the terminal sends standard repetitive ringing at a cycle of 2 seconds on, 4 seconds off, using idle and seize tones as before. If the mobile does not answer within 45 seconds, ringing (6), is discontinued and the call abandoned. When the mobile subscriber goes off-hook to answer, the mobile supervisory unit sends a burst of connect tone (1633 Hz) as an answer signal (8). Upon receipt of the answer signal, the control terminal stops the ringing and establishes a talking path between the calling circuit and the radio channel (7). When the subscriber hangs-up (8). At the end of call, the mobile supervisory unit sends disconnect signal (12. Alternating the disconnect tone (1336 Hz) and the guard tone. The mobile supervisory unit then turns of the mobile transmitter and begins searching for the market idle channel. Each on-hook mobile unit receiving the number transmission compares the received number to its unit number. Only the one mobile unit with a number match remains locked on that channel.

3.5 Outgoing Call

The sequence for a call originated by a mobile subscriber is illustrated. When the subscriber goes off-hook to place the call, the mobile unit must be locked on the marked-idle channel. If not, the hand set will be inoperative and the busy lamp on the control unit will light, indicating to the subscriber that no channel is available. If the mobile unit is locked on the marked idle channel, the mobile supervisory unit will turn on the mobile transmitter to initiate the acknowledgement or handshake sequence.

Then mobile unit transmits its own number so the control terminal can identify it as a subscriber and can charge the call to the number. The roaming functions, are similar to those.

When a call is originated from the field, the mobile unit finds a marked idle channel and broadcasts an acknowledgement to the base by sending its identification. The mobile unit than completes a call in the usual manner by receiving a dial tone, then dialling the number and waiting for the called party to answer.Figure 3.3 shows the Outgoing Call.

3.6 Mobile Station

A Mobile Station consists of two main elements: The Mobile Terminal (MT) and the Subscriber Identity Module (SIM). 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 handheld terminals have experienced the biggest success thanks to their 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.

3.7 Mobile Internal Call(MIC)

The MSI sends the call setup information dialed by the mobile subscriber (MSISDN) to the MSI(1). The MSC request information about the celling mobile subscriber MS2 from the VLR (2). The MSI uses the dialling information (MSISDN) to establish the HLR and sets up signalling connection to it (3). The HLR sends a request to the VLR in whose are the called mobile subscriber MS2 is currently roaming (4). The VLR sends the requested MSRN back to the HLR. The HLR forwards the MSRN to the MSC(5). Steps (6) to (9) are the same as steps (6) to (9) traditional silicon in photovoltaic cells in space because of its supetior efficiency yielding about one-third more power for comparable cell areas.

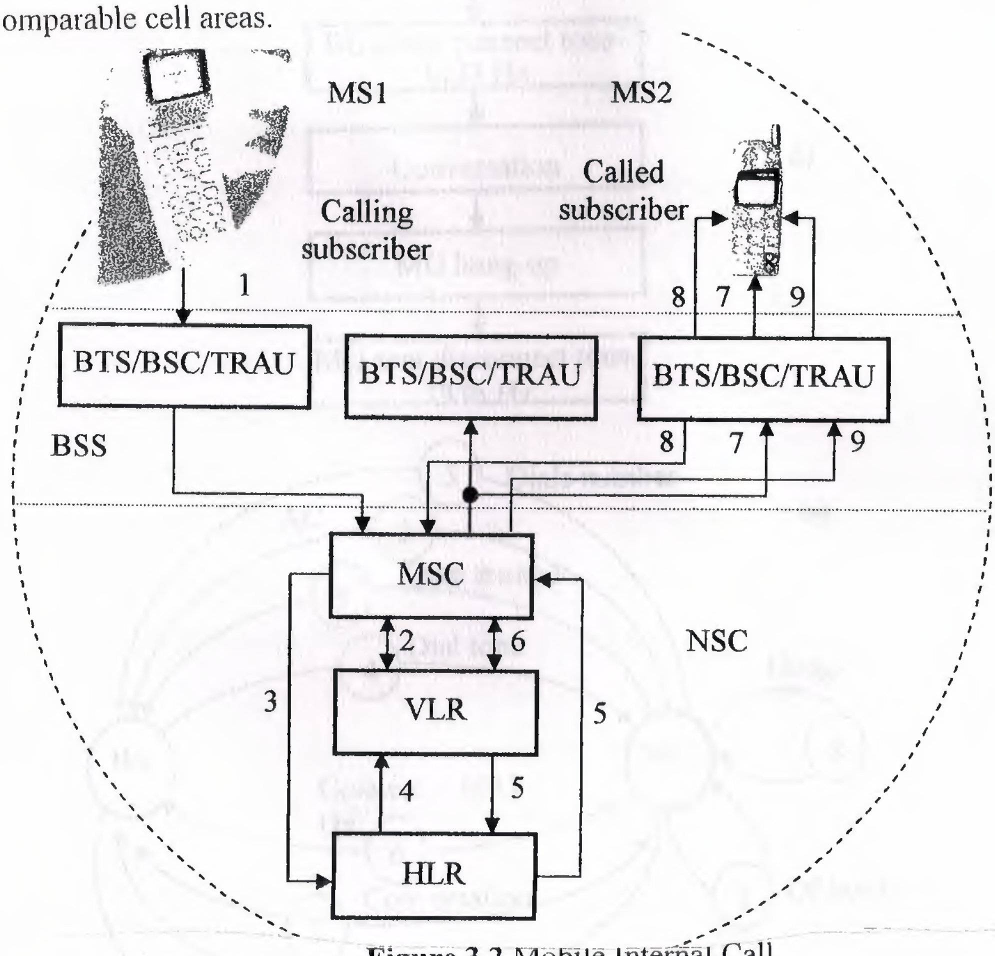


Figure 3.2 Mobile Internal Call

NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

MOBILE PHONES AND BASE STATION

Graduation Project
EE- 400

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INTRODUCTION

GSM (Global System for Mobile Communications) is a European digital communications standard which provides full duplex data traffic to any device fitted with GSM capability, it can easily interface with other digital communications systems, such as ISDN, and digital devices, such as Group 3 facsimile machines.

Unlike any other service, GSM products such as cellular phones require the use of a Subscriber Identity Module, or SIM card. These small electronic devices are approximately the size of a credit card and record all of the user information it. This includes data such as programmed telephone numbers and network security features, which identify the user. Without this module, the device will not function. This allows for greater security and also greater ease of use as this card may be transported from one phone to another, while maintaining the same information available to the user. GSM is also present outside of Europe but known by different names.

The only difference between these systems is the frequency at which operate. The number stands for the operating frequency in megahertz. While each system uses the GSM standard, they are not compatible with each other.

1. INTRODUCTION TO GSM

1.1 Overview

GSM (Global System for Mobile Communications) is a European digital communications standard which provides full duplex data traffic to any device fitted with GSM capability, such as a phone, fax, or pager, at a rate of 9600 bps using the TDMA communications scheme. Since GSM is purely digital, it can easily interface with other digital communications systems, such as ISDN, and digital devices, such as Group 3 facsimile machines.

Unlike any other service, GSM products such as cellular phones require the use of a Subscriber Identity Module, or SIM card. These small electronic devices are approximately the size of a credit card and record all of the user information it. This includes data such as programmed telephone numbers and network security features, which identify the user. Without this module, the device will not function. This allows for greater security and also greater ease of use as this card may be transported from one phone to another, while maintaining the same information available to the user. GSM is also present outside of Europe but known by different names.

In North America it is known as PCS 1900 and elsewhere are DCS 1800 (also known as PCS). The only difference between these systems is the frequency at which operate. The number stands for the operating frequency in megahertz. While each system uses the GSM standard, they are not compatible with each other. Figure 1.1 shows the evolution of the Mobile.

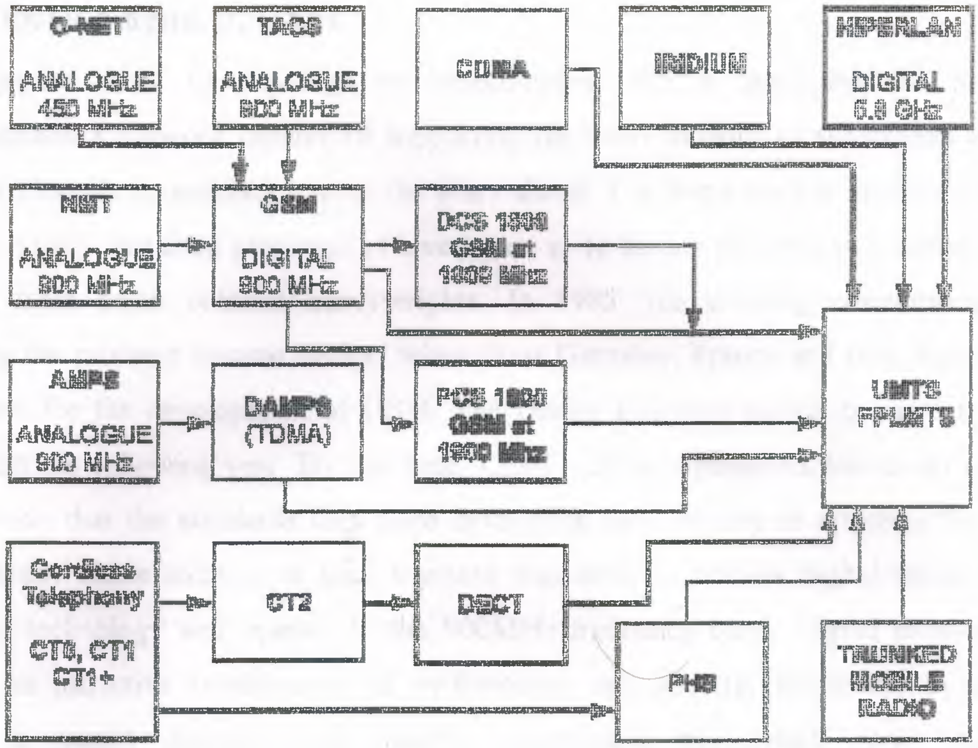


Figure 1.1 The Mobile Evolution

1.2 History Of GSM

During the early 1980s, analog cellular telephone systems were experiencing rapid growth in Europe, particularly in Scandinavia and the United Kingdom, but also in France and Germany. In the Nordic and Benelux countries the NMT 450 was developed, TACS in the UK and C-Netz in West Germany. The Radio com 2000 was in France and RTMI/RTMS in Italy. But each system was incompatible with everyone else's in equipment and operation and as business was becoming increasingly international, the cutting edge of the communications industry focused on exclusively local cellular solutions. These systems were fine if you wanted to call the office if you were in your own home, but not if you were with a client in another country. Also home market revenue simply wouldn't justify sustained programs of investment. As a solution in 1982 CEPT, the Conference des Administrations Europeans des Postes et Telecommunications comprised the telecom administrations of twenty-six European countries, established the Group Special Mobile (GSM).

1.2.1 Developments of GSM

Its objective was to develop the specification for a pan-European mobile communications network capable of supporting the many millions of subscribers likely to turn to mobile communications in the years ahead. The home market revenue simply wouldn't justify sustained programs of investment so to further progress they lobbied for support from some political heavyweights. In 1985, the growing commitment to resolving the problem became evident when West Germany, France and Italy signed an agreement for the development of GSM. The United Kingdom added its name to the agreement the following year. By this time, CEPTs Group Special Mobile could argue persuasively that the standards they were developing held the key to a technically and economically viable solution as their standard was likely to employ digital rather than analogue technology and operate in the 900MHz frequency band. Digital technology offered an attractive combination of performance and spectral efficiency. In other words, it would provide high quality transmission and enable more callers simultaneously to use the limited radio band available. In addition, such a system would allow the development of advanced features like speech security and data communications. Handsets could be cheaper and smaller. It would also make it possible to introduce the first hand-held terminals - even though in the early days in terms of size and weight these would be practically indistinguishable from a brick. Finally, the digital approach neatly complemented the Integrated Services Digital Network (ISDN), which was being developed by land-based telecommunications systems throughout the world. But the frequencies to be employed by the new standard were being snapped up by the analogue networks. Over-capacity crisis had started to sound alarm bells throughout the European Community. Demand was beginning to outstrip even the most optimistic projections. The Group Special Mobile's advocacy of digital cellular technology was on hand to offer light at the end of the tunnel. The Directive ensured that every Member State would reserve the 900MHz frequency blocks required for the rollout program. Although these were somewhat smaller than the amount advocated by the CEPT, the industry had finally achieved the political support it needed to advance its objectives. The logistical nightmare in the GSM, which followed soon left this achievement as a distant, dream so single, permanent organization at the helm.

In1986 the GSM Permanent Nucleus was formed and its head quarters established in Paris. It was all very well agreeing the technology and standards for this new product. But what about the creation of a market? It was essential to forge a commercial agreement between potential operators who would commit themselves to implementing the standard by a particular date. Without such an agreement there could be no network. Without the network there would be no terminals. Without network and terminals there would be no service. Stephen Temple of the UK's Department of Trade and Industry was charged with the task of drafting the first Memorandum of Understanding (MOU). In September 1987 network operators from thirteen countries signed a MOU in Copenhagen. One of the most important conclusions drawn from the early tests was that the new standard should employ Time Division Multiple Access (TDMA) technology. The strength of its technical performance ensured that narrowband TDMA had the support of major players like Nokia, Ericsson and Siemens. This promised the flexibility inherent in having access to a broad range of suppliers and the potential to get product faster into the marketplace. But as always as soon as one problem was solved other problems looming on the horizon.

In 1989, the UK Department of Trade and Industry published a discussion document called "Phones on the Move". This advocated the introduction of mass-market mobile communications using new technology and operating in the 1800 MHz frequency band. The UK government licensed two operators to run what became known as Personal Communications Networks (PCN). Operating at the higher frequency gave the PCN operators virtually unlimited capacity, where as 900MHz was limited. The next hurdle to over come was that of the deadline. If the 1 July 1991 launch date was not met there was a real danger that confidence in GSM technology would be fatally undermined but moral received a boost when in 1989 the responsibility for specification development passed from the GSM Permanent Nucleus to the newly created European Telecommunications Standards Institute (ETSI). In addition, the UK's PCN turned out to be more of an opportunity than a threat. The new operators decided to utilize the GSM specification - slightly modified because of the higher frequency - and the development of what became known as DCS 1800 was carried out by ETSI in parallel with GSM standardization. In fact, in 1997 DCS 1800 was renamed GSM 1800 to reflect the affinity between the two technologies. With so many manufacturers creating so many products in so many countries, it soon became apparent that it was critical that each type of terminal was subject to a rigorous approval regime. Rogue terminals could cause untold damage to the new networks. The solution was the introduction of Interim Type Approval (ITA). Essentially, this was a procedure in which only a subset of the approval parameters was tested to ensure that the terminal in question would not create any problems for the networks. In spite of considerable concern expressed by some operators, ITA terminals became widely available in the course of 1992. True hand held terminals hit the market at the end of that year and the GSM bandwagon had finally started to roll. From here the G.S.M became a success story. In 1987, the first of what was to become an annual event devoted to the worldwide promotion of GSM technology was staged by conference organizers IBC Technical Services. The Pan European Digital Cellular Conference. This year it celebrated its tenth anniversary in Cannes, attracting over 2,400 delegates. By the end of 1993, GSM had broken through the 1 million-subscriber barrier with the next million already on the horizon. By June 1995 Phase 2 of standardization came in to play and a demonstration of fax, video and data communication via GSM. When the GSM standard was being drawn up by the CEPT, six separate systems were all considered as the base. There were seven criteria deemed to be of importance when assessing which of the six would be used. Each country developed its own system, which was incompatible with everyone else's in equipment and operation. This was an undesirable situation, because not only was the mobile equipment limited to operation within national boundaries, which in a unified Europe were increasingly unimportant, but there was also a very limited market for each type of equipment, so economies of scale and the subsequent savings could not be realized. The Europeans realized this early on, and in 1982 the Conference of European Posts and Telegraphs (CEPT) formed a study group called the Group Special Mobile (GSM) to study and develop a pan-European public land mobile system. The proposed system had to meet certain criteria. In 1989, GSM responsibility was transferred to the European Telecommunication Standards Institute (ETSI), and phase-I of the GSM specifications were published in 1990. Commercial service was started in mid-1991, and by 1993 there were 36 GSM networks in 22 countries with 25 additional countries having already selected or considering GSM. This is not only a European standard - South Africa, Australia, and many Middle and Far East countries have chosen GSM. Although standardized in Europe, GSM is not only a European standard. Over 200

GSM networks (including DCS1800 and PCS1900) are operational in 110 countries around the world. In the beginning of 1994, there were 1.3 million subscribers worldwide, which had grown to more than 55 million by October 1997. With North America making a delayed entry into the GSM field with a derivative of GSM called PCS1900, GSM systems exist on every continent, and the acronym GSM now aptly stands for Global System for Mobile communications. The developers of GSM chose an unproven (at the time) digital system, as opposed to the then-standard analog cellular systems like AMPS in the United States and TACS in the United Kingdom. They had faith that advancements in compression algorithms and digital signal processors would allow the fulfillment of the original criteria and the continual improvement of the system in terms of quality and cost. The over 8000 pages of GSM recommendations try to allow flexibility and competitive innovation among suppliers, but provide enough standardization to guarantee proper inter-working between the components of the system. This is done by providing functional and interface descriptions for each of the functional entities defined in the system. The development of GSM started in 1982, when the Conference of European Posts and Telegraphs (CEPT) formed a study group called Group Special Mobile (the initial meaning of GSM). The group was to study and develop a pan-European public cellular system in the 900 MHz range, using spectrum that had been previously allocated. At that time, there were many incompatible analog cellular systems in various European countries. Some of the basic criteria for their proposed system were:

- Good subjective speech quality.
- Low terminal and service cost.
- Support for international roaming.
- Ability to support handheld terminals.
- Support for range of new services and facilities.
- Spectral efficiency.
- ISDN compatibility.

In 1989, the responsibility for GSM was transferred to the European Telecommunication Standards Institute (ETSI), and the Phase I recommendations were published in 1990. At that time, the United Kingdom requested a specification based on GSM but for higher user densities with low-power mobile stations, and operating at 1.8

GHz. The specifications for this system, called Digital Cellular System (DCS1800) were published 1991. Commercial operation of GSM networks started in mid-1991 in European countries. By the beginning of 1995, there were 60 countries with operational or planned GSM networks in Europe, the Middle East, the Far East, Australia, Africa, and South America, with a total of over 5.4 million subscribers. As it turned out, none of the six candidates was actually used! The information collected during the tests did enable the GSM (Group Special Mobile) to design the specifications of the current GSM network. The total change to a digital network was one of the fundamental factors of the success of GSM. Digital transmission is easier to decode than analogue due to the limited number of possible input values (0,1), and as ISDN was becoming de facto at the time, it was logical to avail of digital technology. This also ensured that GSM could evolve properly in an increasingly digital world, for example with the introduction of an 8kps speech coder. It is much easier to change channel characteristics digitally than analogously. Finally, the transmission method decided on for the network was TDMA, as opposed to FDMA and CDMA. In 1989, responsibility for the specification was passed from CEPT to the newly formed and now famous European Telecommunications Standards Institute (ETSI). By 1990, the specifications and explanatory notes on the system were documented extensively, producing 138 documents in total, some reaching sizes of several hundred pages in length services.

1.3 Technology

1.3.1 Services Provided by GSM

From the beginning, the planners of GSM wanted ISDN compatibility in terms of the services offered and the control signaling used. However, radio transmission limitations, in terms of bandwidth and cost, do not allow the standard ISDN B-channel bit rate of 64 kbps to be practically achieved. Using the ITU-T definitions, telecommunication services can be divided into bearer services, tele-services, and supplementary services. The digital nature of GSM allows data, both synchronous and asynchronous, to be transported as a bearer service to or from an ISDN terminal. Data can use either the transparent service, which has a fixed delay but no guarantee of data integrity, or a non-transparent service, which guarantees data integrity through an Automatic Repeat Request (ARQ) mechanism, but with a variable delay. The data rates supported by

GSM are 300 bps, 600 bps, 1200 bps, 2400 bps, and 9600 bps. The most basic tele-service supported by GSM is telephony. As with all other communications, speech is digitally encoded and transmitted through the GSM network as a digital stream. There is also an emergency service, where the nearest emergency-service provider is notified by dialing three digits (similar to 911). A variety of data services is offered. GSM users can send and receive data, at rates up to 9600 bps, to users on POTS (Plain Old Telephone Service), ISDN, Packet Switched Public Data Networks, and Circuit Switched Public Data Networks using a variety of access methods and protocols, such as X.25 or X.32. Since GSM is a digital network, a modem is not required between the user and GSM network, although an audio modem is required inside the GSM. Network to inter-work with POTS. Other data services include Group 3 facsimile, as described in ITU-T recommendation T.30, which is supported by use of an appropriate fax adaptor. A unique feature of GSM, not found in older analog systems, is the Short Message Service (SMS). SMS is a bi directional service for short alphanumeric (up to 160 bytes) messages. Messages are transported in a store-and-forward fashion. For point-to-point SMS, a message can be sent to another subscriber to the service, and an acknowledgement of receipt is provided to the sender. SMS can also be used in a cell-broadcast mode, for sending messages such as traffic updates or news updates. Messages can also be stored in the SIM card for later retrieval supplementary services are provided on top of tele-services or bearer services. In the current (Phase I) specifications, they include several forms of call forward (such as call forwarding when the mobile subscriber is unreachable by the network), and call barring of outgoing or incoming calls, for example when roaming in another country. Many additional supplementary services will be provided in the Phase 2 specifications, such as caller identification, call waiting, multi-party conversations. GSM was designed having interoperability with ISDN in mind, and the services provided by GSM are a subset of the standard ISDN services. Speech is the most basic, and most important, tele-service provided by GSM. In addition, various data services are supported, with user bit rates up to 9600 bps. Specially equipped GSM terminals can connect with PSTN, ISDN, Packet Switched and Circuit Switched Public Data Networks, through several possible methods, using synchronous or asynchronous transmission. Also supported are Group 3 facsimile service, video-tex, and teletex. Other GSM services include a cell broadcast service, where messages such as traffic reports, are broadcast to users in particular cells. A service unique to GSM, the Short Message Service, allows users to send and receive point-to-point alphanumeric messages up to a few tens of bytes. It is similar to paging services, but much more comprehensive, allowing bi-directional messages, store-and-forward delivery, and acknowledgement of successful delivery.

1.4 The Different GSM-Based Networks

Different frequency bands are used for GSM 900, GSM1800 and GSM 1900 (Table 1.3.). 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.

Table 1.3 Frequency Bands For The Different GSM-Based Networks

Network type	Frequency band UL / DL	Implementations
GSM 900	890-915 / 935-960 MHz	GSM 900
GSM1800	1710 – 1785 / 1805 -1880 MHz	GSM 1800
GSM1900	1850-1910 / 1930-1990 MHz	GSM1900

1.4.1 Where are GSM Frequencies Used?

GSM networks presently operate in three different frequency ranges. These are:

a) GSM 900

(Also called GSM) operates in the 900 MHz frequency range and is the most common in Europe and the world.

b) GSM 1800

(Also called PCN (Personal Communication Network), and DCS 1800) - operates in the 1800 MHz frequency range and is found in a rapidly-increasing number of countries including France, Germany, Switzerland, the UK, and Russia. A European Commission mandate requires European Union members to license at least one DCS 1800 operator before 1998.

c) GSM 1900

(Also called PCS (Personal Communication Services), PCS 1900, and DCS 1900) - the only frequency used in the United States and Canada for GSM. Note that the terms PCS is commonly used to refer to any digital cellular network operating in the1900 MHz frequency range, not just GSM.

2. GSM RADIO INTERFACE

2.1 Overview

The Radio interface is the interface between the mobile stations and the fixed infrastructure. It is one of the most important interfaces of the GSM system. One of the main objectives of GSM is roaming. Therefore, in order to obtain a complete compatibility between mobile stations and networks of different manufacturers and operators, the radio interface must be completely defined. The spectrum efficiency depends on the radio interface and the transmission, more particularly in aspects such as the capacity of the system and the techniques used in order to decrease the interference and to improve the frequency reuse scheme. The specification of the radio interface has then an important influence on the spectrum efficiency.

2.2 Frequency Allocation

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

- 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).

These bands were allocated by the ITU (International Telecom Union) who are responsible for allocating radio spectrum on an international basis. Although these bands were (and still are) used by analog systems in the early 1980's, the top 10 MHz were reserved for the already emerging GSM Network by the CEPT (European Conference of Posts and Telecommunications: translated from French). But not all the countries can use the whole GSM frequency bands. This is due principally to military reasons and to the existence of previous analog systems using part of the two 25 Mhz frequency bands.

2.3 Multiple Access Scheme

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

It is hoped that eventually the GSM network will use the entire bandwidth. It is apparent from this that the bandwidth you use on a day-to-day basis to operate your mobile phone is limited. It would seem that only a certain number of users can operate on the bandwidth simultaneously. However GSM has devised a method to maximize the bandwidth available. They use a combination of Time and Frequency Division Multiple Access (TDMA/FDMA).

a) **FDMA:** 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.

This is the division of the bandwidth in to 124 carrier frequencies each of 200 kHz. At least one of these is assigned to each base station. Figure 2.1 shows the FDMA System.

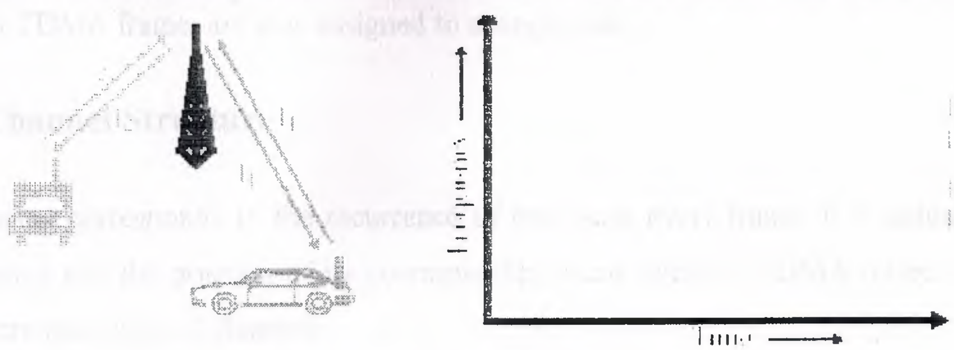


Figure2.1 Frequency Division Multiple Access

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

The carrier frequencies are then divided again into 8 time slots. This prevents mobiles from transmitting and receiving calls at the same time as they are allocated separate time slots. Figure 2.2 shows Time Division Multiple Access System.

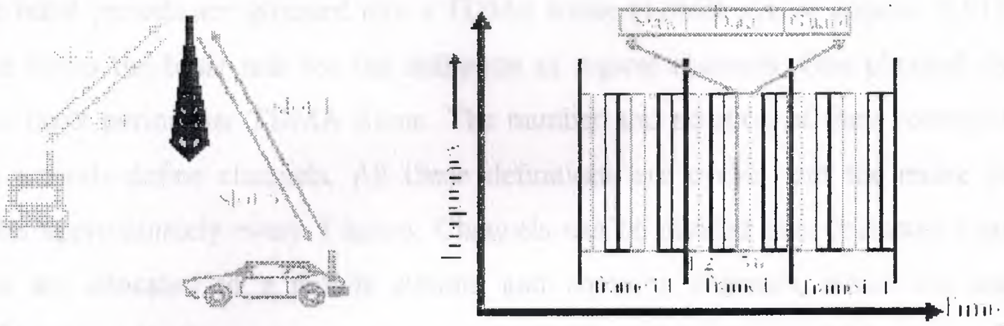


Figure 2.2 Time Division Multiple Access

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.

2.4 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.

Since radio spectrum is a limited resource shared by all users, a method must be devised to divide up the bandwidth among as many users as possible. The method chosen by GSM is a combination of Time- and Frequency-Division Multiple Access (TDMA/FDMA). The FDMA part involves the division by frequency of the (maximum) 25 MHz bandwidth into 124 carrier frequencies spaced 200 kHz apart. One or more carrier frequencies are assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme. The fundamental unit of time in this TDMA scheme is called a burst period and it lasts 15/26 ms (or approx. 0.577 ms). Eight burst periods are grouped into a TDMA frame (120/26 ms, or approx. 4.615 ms), which forms the basic unit for the definition of logical channels. One physical channel is one burst period per TDMA frame. The number and position of their corresponding burst periods define channels. All these definitions are cyclic, and the entire pattern repeats approximately every 3 hours. Channels can be divided into dedicated channels, which are allocated to a mobile station, and common channels, which are used by mobile stations in idle mode.

2.4.1 Traffic Channels

A traffic channel (TCH) is used to carry speech and data traffic. Traffic channels are defined using a 26-frame multi frame, or group of 26 TDMA frames. The length of a 26-frame multi frame is 120 ms, which is how the length of a burst period is defined (120 ms divided by 26 frames divided by 8 burst periods per frame). Out of the 26 frames, 24 are used for traffic, 1 is used for the Slow Associated Control Channel (SACCH) and 1 is currently unused (see Figure 3.1). TCHs for the uplink and downlink are separated in time by 3 burst periods, so that the mobile station does not have to transmit and receive simultaneously, thus simplifying the electronics. In addition to these full-rate TCHs, there are also half-rate TCHs defined, although they are not yet implemented.

Half-rate TCHs will effectively double the capacity of a system once half-rate speech coders are specified (i.e., speech coding at around 7 kbps, instead of 13 kbps). Eighth-rate TCHs are also specified, and are used for signaling. In the recommendations, they are called Stand-alone Dedicated Control Channels (SDCCH). 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.

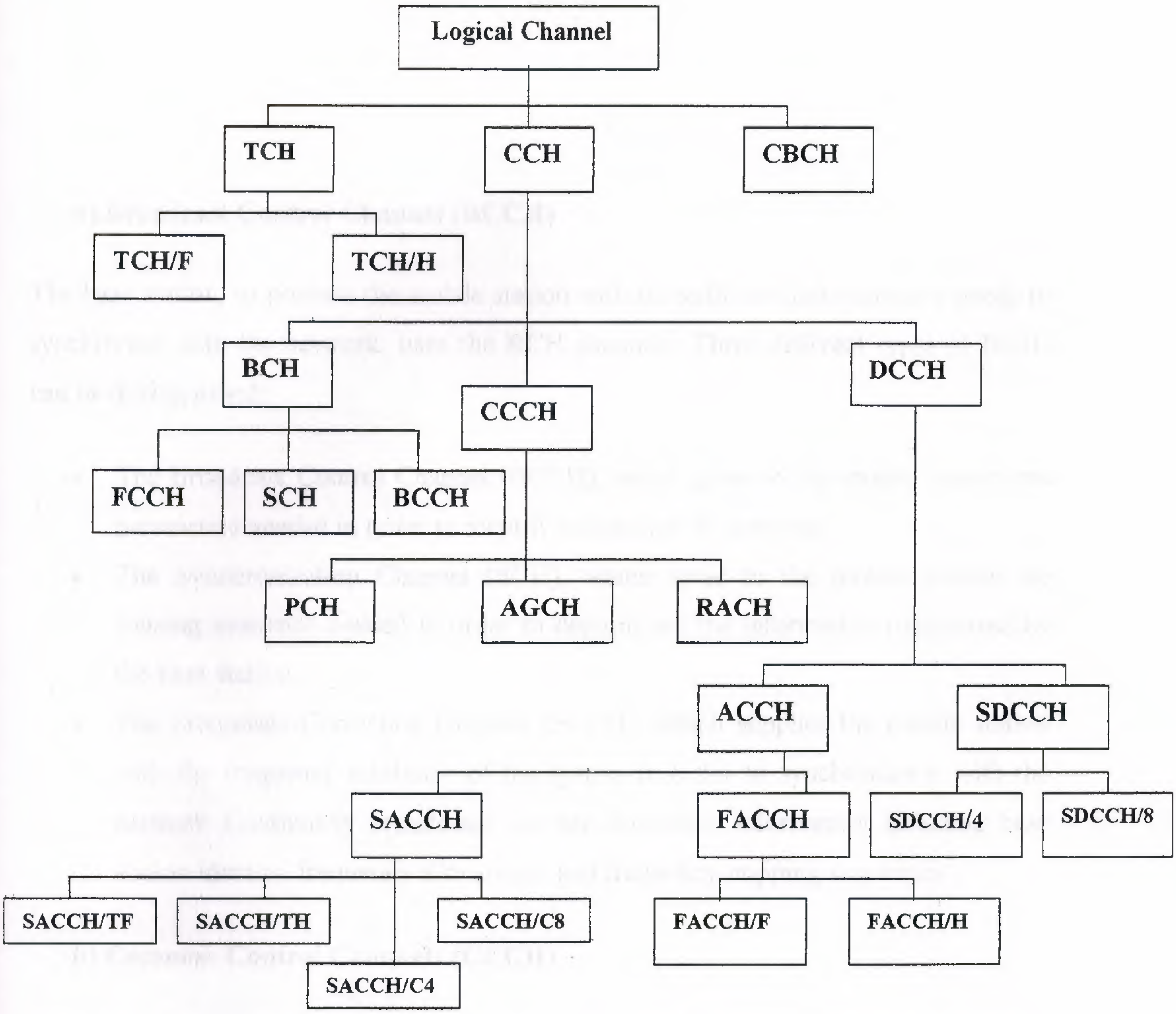
Half-rate traffic channels (TCH/H), which double the capacity of the system, are also grouped in a 26-Multiframe but the internal structure is different.

2.4.2 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.

Common channels can be accessed both by idle mode and dedicated mode mobiles. Idle mode mobiles to exchange the signalling information required to change to dedicated mode use the common channels. Mobiles already in dedicated mode monitor the surrounding base stations for handover and other information. The common channels are defined within a 51-frame multiframe, so that dedicated mobiles using the 26-frame multiframe TCH structure can still monitor control channels. Figure 2.3 shows the Logical channels. The common channels include:



TCH: Traffic Channel.
TCH/F: Traffic Channel/Full.
TCH/H: Traffic Channel/Half.
CCH: Control Channel.
BCH:Broadcast Channel.
CBCH:Cell Broadcast Channel.
CCCH: Common Control Channel.
ACCH: Associated Control Channel.
SACCH: Slow Associated Control Channel.
FACCH: Fast Associated Control Channel.
SDCCH: Stand-Alone Ded icated Control Channel.

FCCH: Freq. Correction Channel.
SCH: Synchronization Channel.
BCCH: Broadcast Control Channel.
PCH: Paging Channel.
AGCH: Access Grant Channel.
RACH: Random Access Channel.
DCCH: Dedicated Control Channel.

Figure 2.3 Structure of Logical Channels

a) Broadcast Control Channel (BCCH)

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 Continually broadcasts, on the downlink, information including base station identity, frequency allocations, and frequency-hopping sequences.

b) 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 call.
- 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.

c) Frequency Correction Channel (FCCH) and Synchronization Channel (SCH)

Used to synchronize the mobile to the time slot structure of a cell by defining the boundaries of burst periods, and the time slot numbering. Every cell in a GSM network broadcasts exactly one FCCH and one SCH, which are by definition on time slot number 0 (within a TDMA frame).

d) 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.

e) 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.

f) Random Access Channel (RACH)

Slotted Aloha channel used by the mobile to request access to the network.

g) Paging Channel (PCH)

Used to alert the mobile station of an incoming call.

h) Access Grant Channel (AGCH)

Used to allocate an SDCCH to a mobile for signaling (in order to obtain a dedicated channel), following a request on the RACH.

2.4.3 Burst Structure

There are four different types of bursts used for transmission in GSM. The normal burst is used to carry data and most signaling. It has a total length of 156.25 bits, made up of two 57 bit information bits, a 26 bit training sequence used for equalization, 1 stealing bit for each information block (used for FACCH), 3 tail bits at each end, and an 8.25 bit guard sequence, as shown in Figure2.4. The 156.25 bits are transmitted in 0.577 ms, giving a gross bit rate of 270.833 kbps. The F burst, used on the FCCH, and the S burst, used on the SCH, have the same length as a normal burst, but a different internal structure, which differentiates them from normal bursts (thus allowing synchronization). The access burst is shorter than the normal burst, and is used only on the RACH. 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.

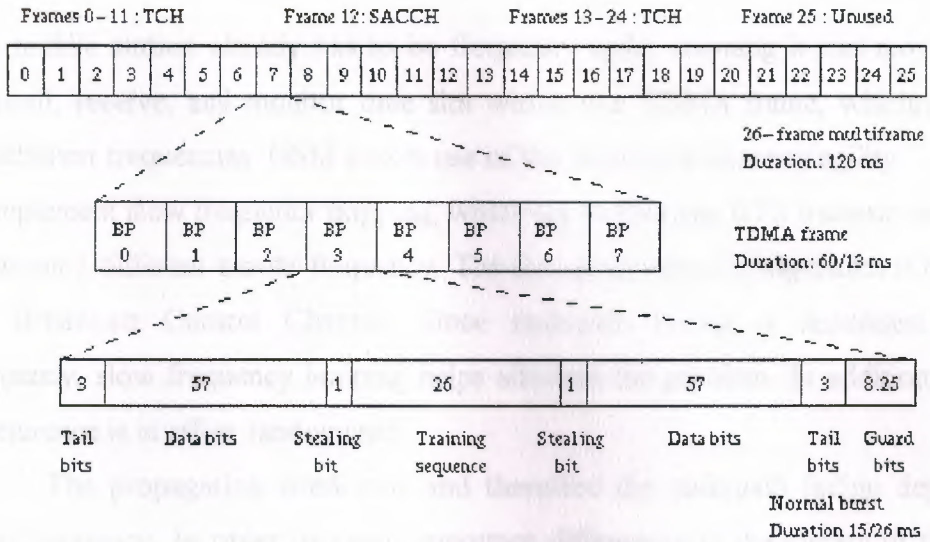


Figure 2.4 Structure of the 26-Multiframe, the TDMA frame and the normal burst

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.

2.4.4 Frequency Hopping

The mobile station already has to be frequency agile, meaning it can move between a transmit, receive, and monitor time slot within one TDMA frame, which normally are on different frequencies. GSM makes use of this inherent frequency agility to implement slow frequency hopping, where the mobile and BTS 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 helps alleviate the problem. In addition, co-channel interference is in effect randomized.

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.

2.5 From source information to radio waves

The figure 2.5 presents the different operations that have to be performed in order to pass from the speech source to radio waves and vice versa. If the source of information is data and not speech, the speech coding will not be performed.

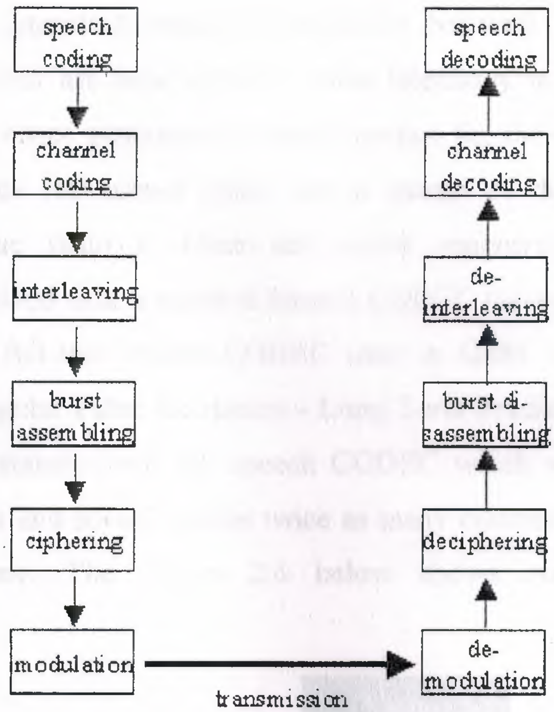


Figure 2.5 From Speech Source To Radio Waves

2.5.1 Speech Coding

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

- 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 coder must not be very complex because complexity is equivalent to high costs.

The final choice for the GSM speech coder is a coder named RPE-LTP (Regular Pulse Excitation Long-Term Prediction). This coder 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 coder, which has a rate of 13 kbps, in order to obtain blocks of 260 bits. Obviously the most important aspect of the GSM Network is speech transmission. Although other services are now offered, voice telephony is still the most popular service available and hence generates the most revenue for the various companies. The device that transforms the human voice into a stream of digital data, suitable for transmission over the radio interface and which regenerates an audible analog representation of received data is called a Speech CODEC (speech transcoder or speech coder/decoder). The full-rate speech CODEC used in GSM is known as RPE-LTP, which stands for "Regular Pulse Excitation - Long Term Prediction". It is hoped there will eventually be a standardized full speech CODEC which will half the amount of data to be transmitted and so will enable twice as many customers to use the same slot in the TDMA frame. The Figure 2.6 below shows audio signal processing

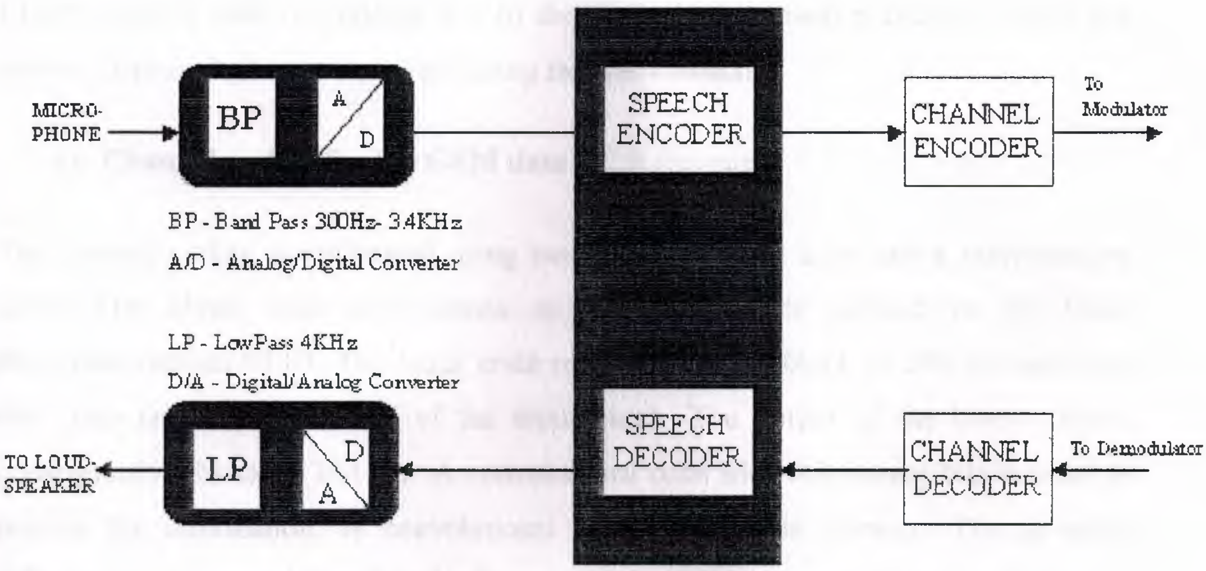


Figure 2.6 Audio Signal Processing

GSM is a digital system, so speech which is inherently analog, has to be digitized. The method employed by ISDN, and by current telephone systems for multiplexing voice lines over high-speed trunks and optical fiber lines, is Pulse Coded Modulation (PCM). The output stream from PCM is 64 kbps, too high a rate to be feasible over a radio link. The 64 kbps signal, although simple to implement, contains much redundancy. The GSM group studied several speech coding algorithms on the basis of subjective speech quality and complexity (which is related to cost, processing delay, and power consumption once implemented) before arriving at the choice of a Regular Pulse Excited Linear Predictive Coder (RPE-LPC) with a Long Term Predictor loop. Basically, information from previous samples, which does not change.

Very quickly, is used to predict the current sample. The coefficients of the linear combination of the previous samples, plus an encoded form of the residual, the difference between the predicted and actual sample, represent the signal. Speech is divided into 20 millisecond samples, each of which is encoded as 260 bits, giving a total bit rate of 13 kbps. This is the so-called Full-Rate speech coding. Recently, some North American GSM1900 operators have implemented an Enhanced Full-Rate (EFR) speech-coding algorithm. This is said to provide improved speech quality using the existing 13 kbps bit rate.

2.5.2 Channel coding

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

a) 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:

$$C\,(11 + 15\,j) \text{ for } j = 0, 1, \ldots, 31 \qquad (2.1)$$

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

b) 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 Ia containing 50 bits. Next in importance is the class Ib, 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 Ia bits are block-coded. Three parity bits, used for error detection, are added to the 50 class Ia bits. The resultant 53 bits are added to the class Ib 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.

c) 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. Electromagnetic interference can disrupt encoded speech and data transmitted over the GSM Network. Because of this this complicated encoding and block interleaving is used to protect the Network. Speech and data rates use different algorithms. Radio emissions too can cause interference if they occur outside of the allotted bandwidth and must be strictly controlled to allow for both GSM and older analog systems to co-exist. Because of natural and man-made electromagnetic interference, the encoded speech or data signal transmitted over the radio interface must be protected from errors. GSM uses convolutional encoding and block interleaving to achieve this protection. The exact algorithms used differ for speech and for different data rates. The method used for speech blocks will be described below. Recall that the speech coder produces a 260-bit block for every 20 ms speech sample. From subjective testing, it was found that some bits of this block were more important for perceived speech quality than others. The bits are thus divided into three classes:

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

Class Ia bits have a 3 bit Cyclic Redundancy Code added for error detection. If an error is detected, the frame is judged too damaged to be comprehensible and it is discarded. It is replaced by a slightly attenuated version of the previous correctly received frame. These 53 bits, together with the 132 Class Ib bits and a 4-bit tail sequence (a total of 189 bits), are input into a 1/2 rate convolutional encoder of constraint length 4. Each input bit is encoded as two output bits, based on a combination of the previous 4 input bits. The convolutional encoder thus outputs 378 bits, to which are added the 78 remaining Class II bits, which are unprotected. Thus every 20 ms speech sample is encoded as 456 bits, giving a bit rate of 22.8 kbps. To further protect against the burst errors common to the radio interface, each sample is interleaved. The 456 bits output by the convolutional encoder are divided into 8 blocks of 57 bits, and these blocks are transmitted in eight consecutive time-slot bursts. Since each time-slot burst can carry two 57-bit blocks, each burst carries traffic from two different speech samples. Recall that each time-slot burst is transmitted at a gross bit rate of 270.833 kbps. This digital signal is modulated onto the analog carrier frequency using Gaussian-filtered Minimum Shift Keying (GMSK). GMSK was selected over other modulation schemes as a compromise between spectral efficiency, complexity of the transmitter, and limited spurious emissions. The complexity of the transmitter is related to power consumption, which should be minimized for the mobile station. The spurious radio emissions, outside of the allotted bandwidth, must be strictly controlled so as to limit adjacent channel interference, and allow for the co-existence of GSM and the older analog systems (at least for the time being).

2.5.3 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.

a) 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 (4x114 = 456). The 456 bits are divided into eight blocks of 57 bits. The first block of 57 bits contains the bit numbers (0, 8, 16,....., 448), the second one the bit numbers (1, 9, 17, ...,449), etc. The last block of 57 bits will then contain the bit numbers (7, 15,....., 455). The first four blocks of 57 bits are placed in the even-numbered bits of four bursts. The other four blocks of 57 bits are placed in the odd-numbered bits of the same four bursts. Therefore the interleaving depth of the GSM interleaving for control channels is four and a new data block starts every four bursts. The interleaver for control channels is called a block rectangular interleaver.

b) Interleaving for the GSM speech Channels

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

c) 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.

2.5.4 Burst Assembling

The burst assembling procedure is in charge of grouping the bits into bursts. Section 2.4.3. presents the different bursts structures and describes in detail the structure of the normal burst.

2.5.5 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.

2.5.6 Modulation

The modulation chosen for the GSM system is the Gaussian Minimum 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 as 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 kbauds and a BT product equal to 0.3. Figure 2.7. presents the principle of a GMSK modulator.

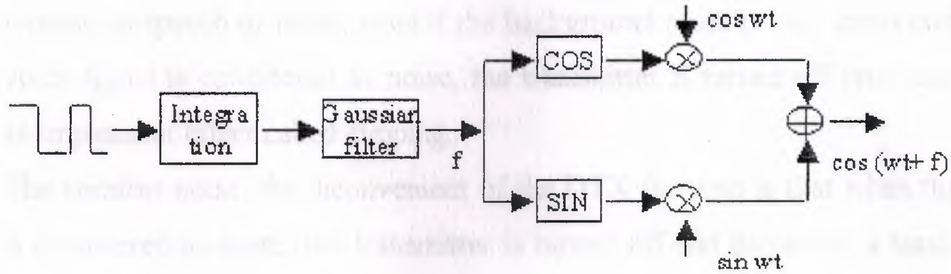


Figure 2.7 GMSK Modulator

2.6 Discontinuous Transmission (DTX)

Minimizing co-channel interference is a goal in any cellular system, since it allows better service for a given cell size, or the use of smaller cells, thus increasing the overall capacity of the system. Discontinuous Transmission (DTX) is a method that takes advantage of the fact that a person speaks less that 40 percent of the time in normal conversation, by turning the transmitter off during silence periods. An added benefit of DTX is that power is conserved at the mobile unit. The most important component of DTX is, of course, Voice Activity Detection (VAD). It must distinguish between voice and noise inputs, a task that is not as trivial as it appears, considering background noise. If a voice signal is misinterpreted as noise, the transmitter is turned off and a very annoying effect called clipping is heard at the receiving end. If, on the

other hand, noise is misinterpreted as a voice signal too often, the efficiency of DTX is dramatically
decreased. Another factor to consider is that when the transmitter is turned off, there is total silence heard at the receiving end, due to the digital nature of GSM. To assure the receiver that the connection is not dead, comfort noise is created at the receiving end by trying to match the characteristics of the transmitting end's background noise. This is another aspect of GSM that could have been included as one of the requirements of the GSM speech coder. 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.

2.7 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.

2.8 Power Control

There are five classes of mobile stations defined, according to their peak transmitter power, rated at 20, 8, 5, 2, and 0.8 watts. To minimize co-channel interference and to conserve power, both the mobiles and the Base Transceiver Stations operate at the lowest power level that will maintain an acceptable signal quality. Power levels can be stepped up or down in steps of 2 dB from the peak power for the class down to a minimum of 13 dBm (20 milli watts). The mobile station measures the signal strength or signal quality (based on the Bit Error Ratio), and passes the information to the Base Station Controller, which ultimately decides if and when the power level should be changed. Power control should be handled carefully, since there is the possibility of instability. This arises from having mobiles in co-channel cells alternatingly increase their power in response to increased co-channel interference caused by the other mobile increasing its power. This in unlikely to occur in practice but it is (or was as of 1991) under study. 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.

2.9 Discontinuous Reception

Another method used to conserve power at the mobile station is discontinuous reception. The paging channel, used by the base station to signal an incoming call, is structured into sub-channels. Each mobile station needs to listen only to its own sub-channel. In the time between successive paging sub-channels, the mobile can go into sleep mode, when almost no power is used. 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.

2.10 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 equalizer 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. At the 900 MHz range, radio waves bounce off everything - buildings, hills, cars, airplanes, etc. Thus many reflected signals, each with a different phase, can reach an antenna. Equalization is used to extract the desired signal from the unwanted reflections. It works by finding out how a known transmitted signal is modified by multipath fading, and constructing an inverse filter to extract the rest of the desired signal. This known signal is the 26-bit training sequence transmitted in the middle of every time-slot burst. The actual implementation of the equalizer is not specified in the GSM specifications.

3. MOBILE PHONES

3.1 Overview

Mobile phones may be thought of as cordless phones with elaborate portable and base units. High-power transmitters and elavated antennas that provide the radio carrier link over an area within 20 to 30 miles from the base station antenna, as well as the multiplexing, detecting, sorting and selecting features required to simultaneously service 60 subscribers per base station, are the major differences between cordless phones and mobile phones.

3.2 Base Unit

The base station can transmit and receive on several different frequencies simultaneously to provide several individual channels for use at the same time. The radio base station transmitter output power is typically 200-250 watts and the radiated power can be as high as 500 watts if the transmitting antenna gain is included. It covers a circular area of up to 30 milles in radius for clear reliable communications, but transmitters with the same frequence are not spacet closer than about 60 to 100 milles because of the noise interfrence levels.

The receiver contains filters, high-gain amplifiers, and demodulators to provide a usable voice signal to the phone line. The control terminal contains the necessary detector and timing and logic circuits to control the transmission link between the base unit and the mobile units. As a result, phone calls are coupled to and from the standard phone system just like calls that are carried completely over wired facilities. The control terminal has the necessary interface circuits so that a call initiated at a mobile unit is interconnected through the national or international phone system to the called party just as any other phone call.

The national and international phone system facilities are owned by the respective phone companies. The base units and mobile units may be owned by the phone company or by a separate company called a radio common carrier (RCC). When the mobile system is run by a RCC, the RCC is charged by the telephone company for the use of the standard phone system just like any other customer.

The cost is then included in the charge by the RCC to the eventual user of the mobile units.

To subscribe to mobile phone service, a user has only to apply, and be accepted by the RCC or the phone company operating the system. When the application is accepted, the user can lease or purchase the mobile equipment.

3.3 Mobile Unit

The mobile unit in the user's vehicle consists of a receiver containing amplifiers, a mixer and a demodulator; a transmitter containing a modulator, carrier oscillators and amplifiers; the necessary control logic; a control unit with microphone, speaker, keypad and switches; antennas and the interconnecting cables. The control unit performs all of the functions associated with normal phone use. A modern control head with automatic functions is illustrated.

The mobile phone user with automatic control places and receives calls in the same manner as with an oridinary phone. When the handset is lifted to place a call, the radio unit automatically selects an available channel. If no channel is available, the busy light comes on, If a channel is found, the user hears the normal dial tone from the phone system, and can then dial the number and proceed as if the phone were direct wired. An incoming call to the mobile unit is signaled by a ringing tone and is answered simply by lifting the handset and talking. Thus, the automatic mobile phone is as easily used as a phone. The mobile phone combines the mobility of the radio link and the world-wide switched network of the existing phone system to provide a communication link to any other phone in the world.

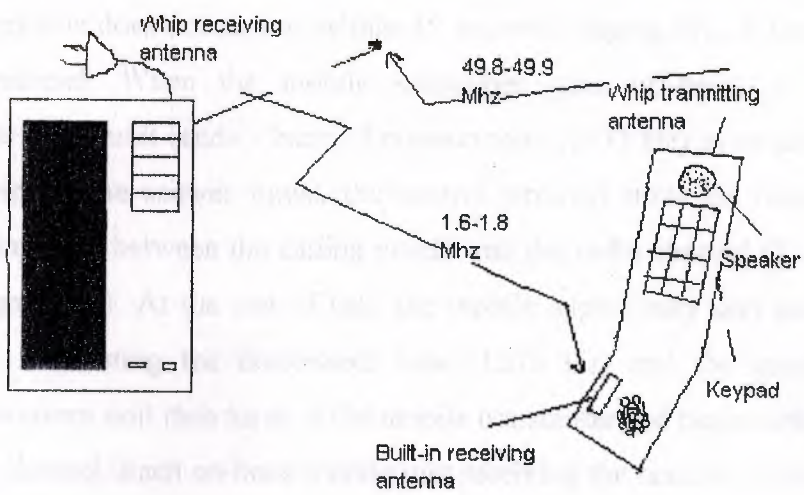


Figure 3.1 Mobile Unit

3.4 Detailed Operation

Different signalling techniques have to used in a mobile phone system in contrast with a wired facility. Since there are no wires connecting the telephone to the network, both speech and signalling must be transmitted via radio. For wireless operation, tones are used for those signaling functions, which are otherwise performed by voltage and current in hard-wired systems. This is accomplished by the use of special tones rather than applying a voltage level or detecting a current. The proper tone transmitted to the mobile unit will, for example, ring the mobile phone to indicate an incoming call just as with a standard phone different tone is used to indicate off-hook, busy, etc. The Improved Mobile Telephone System (IMTS) uses in band signalling tones from 1300 Hz to 2200 Hz. The older Mobile Phone System (MTS) had in band signalling tones in the 600 HZ to 1500 Hz range. Some systems use 2805 Hz as manual operation.

channel. When the mobile unit receives its correct seven-digit address, the mobile supervisory unit turns on the mobile transmitter and sends the acknovledgement signal Ack (5), using the 2150 Hz guard-tone, back to the control terminal. If this acknowledgement is not received by the control terminal within 3 seconds after out-pulsing the address, seize tone is removed and the call is abandoned. However, upon receipt of the mobile acknowledgement signal, the terminal sends standard repetitive ringing at a cycle of 2 seconds on, 4 seconds off, using idle and seize tones as before. If the mobile does not answer within 45 seconds, ringing (6), is discontinued and the call abandoned. When the mobile subscriber goes off-hook to answer, the mobile supervisory unit sends a burst of connect tone (1633 Hz) as an answer signal (8). Upon receipt of the answer signal, the control terminal stops the ringing and establishes a talking path between the calling circuit and the radio channel (7). When the subscriber hangs-up (8). At the end of call, the mobile supervisory unit sends disconnect signal (12. Alternating the disconnect tone (1336 Hz) and the guard tone. The mobile supervisory unit then turns of the mobile transmitter and begins searching for the market idle channel. Each on-hook mobile unit receiving the number transmission compares the received number to its unit number. Only the one mobile unit with a number match remains locked on that channel.

3.5 Outgoing Call

The sequence for a call originated by a mobile subscriber is illustrated. When the subscriber goes off-hook to place the call, the mobile unit must be locked on the marked-idle channel. If not, the hand set will be inoperative and the busy lamp on the control unit will light, indicating to the subscriber that no channel is available. If the mobile unit is locked on the marked idle channel, the mobile supervisory unit will turn on the mobile transmitter to initiate the acknowledgement or handshake sequence.

Then mobile unit transmits its own number so the control terminal can identify it as a subscriber and can charge the call to the number. The roaming functions, are similar to those.

When a call is originated from the field, the mobile unit finds a marked idle channel and broadcasts an acknowledgement to the base by sending its identification. The mobile unit than completes a call in the usual manner by receiving a dial tone, then dialling the number and waiting for the called party to answer.Figure 3.3 shows the Outgoing Call.

3.6 Mobile Station

A Mobile Station consists of two main elements: The Mobile Terminal (MT) and the Subscriber Identity Module (SIM). 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 handheld terminals have experienced the biggest success thanks to their 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.

3.7 Mobile Internal Call(MIC)

The MSI sends the call setup information dialed by the mobile subscriber (MSISDN) to the MSI(1). The MSC request information about the celling mobile subscriber MS2 from the VLR (2). The MSI uses the dialling information (MSISDN) to establish the HLR and sets up signalling connection to it (3). The HLR sends a request to the VLR in whose are the called mobile subscriber MS2 is currently roaming (4). The VLR sends the requested MSRN back to the HLR. The HLR forwards the MSRN to the MSC(5). Steps (6) to (9) are the same as steps (6) to (9) traditional silicon in photovoltaic cells in space because of its supetior efficiency yielding about one-third more power for comparable cell areas.

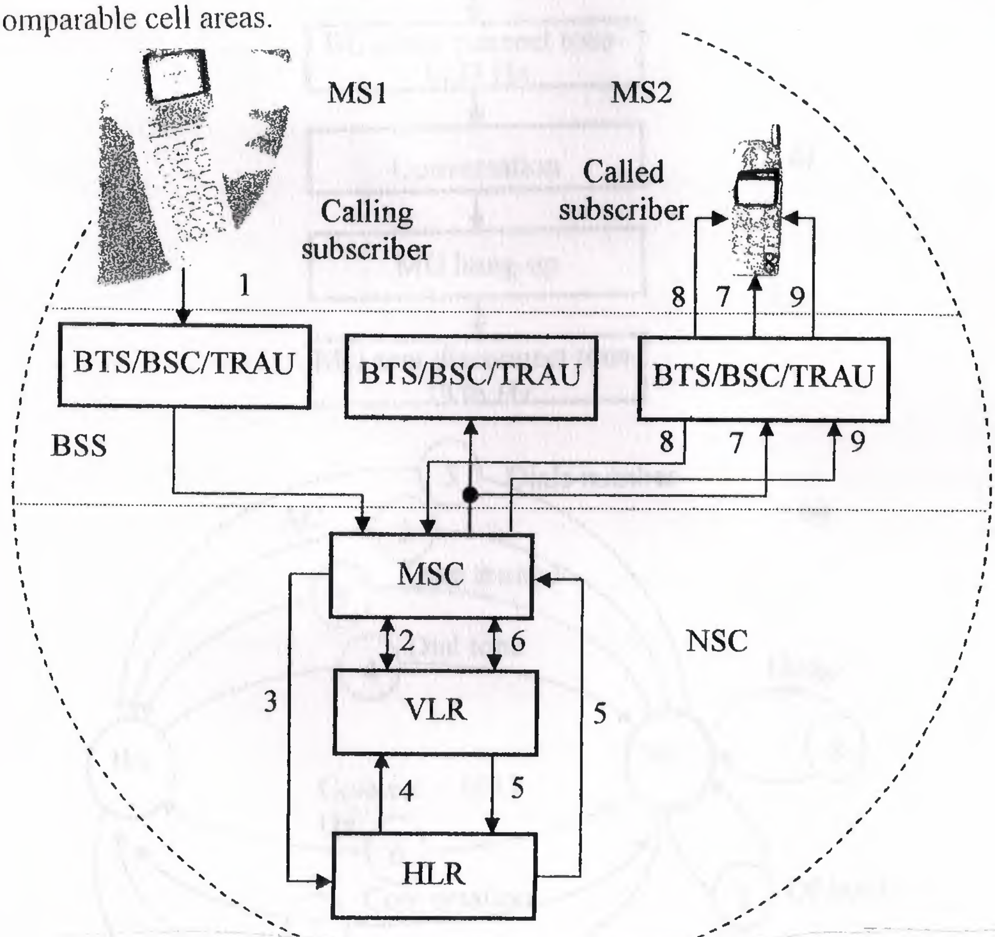
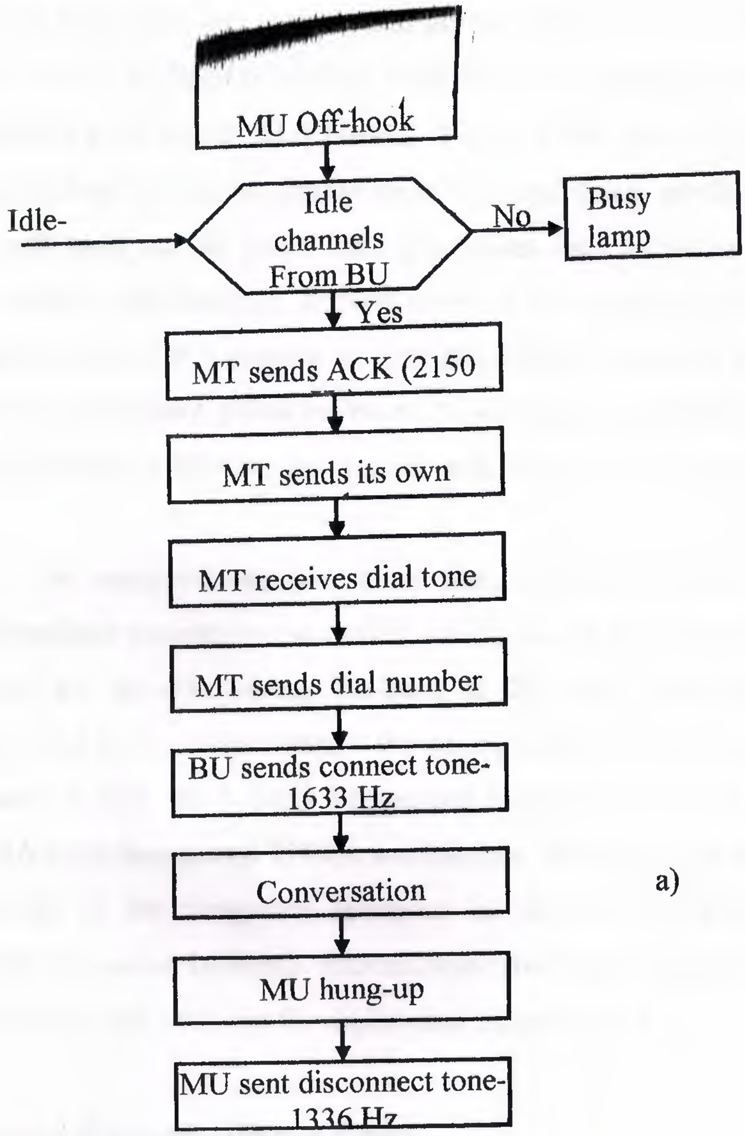
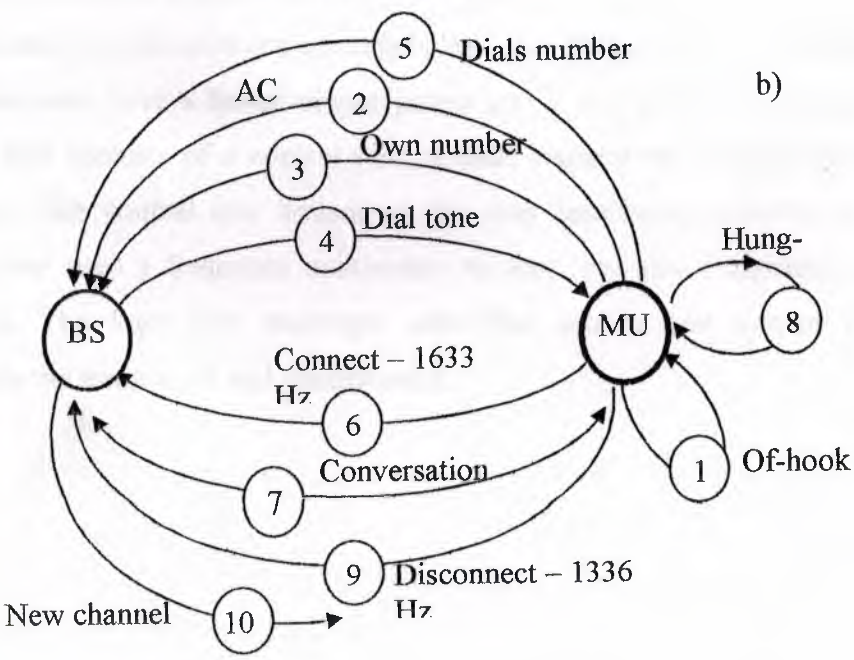


Figure 3.2 Mobile Internal Call



a)



b)

Figure 3.3 Outgoing Call

A trio of phased-away antennas extends and points earthward to establish direct links over the 1.610-1.625-GHz band to Iridium subscribers. The Iridium constellation, with a company-projected price tag of $3.4 billion, is one of the most costly concepts ever devised for providing mobile communication services. Each satellite in the Iridium constellation will send out 48 pencil-thin spot-beams each of which can handle 230 simultaneous duplex conversations. Iridium satellites are distributed among six evenly spaced, near-polar orbits (86.4 degrees inclination) 780 km above the earth, sixty of the satellites provide overlapping global coverage, Polar regions included. The other six are in-orbit spares. Iridium subscriber equipment offer voice, data, paging, and facsimile services.

used instead. The satellite-to-satellite cross links, the satellite-to-Iridium gateway stations and downlinks connecting the iridium satellites with their ground-based system control stations are provided using Ka-band at 20 GHz. The transmission links connecting the hand-held communicators, the paging units, and the remote area phones will all be handled with the L-band frequencies between 1.5 and 1.6 GHz. Iridium employs CDMA modulations and TDMA architecture. This approach will require that a dedicated portion of the frequency spectrum be allocated to Iridium to provide interference-free operation. Iridium's transmission rates have been set at 4800 bps for voice, and both 4800 and 2400 bps for digital data transmissions.

3.8 Mobile And Portable Phone Units

Mobile and portable units are essentially the same things. The only difference is that the portable units have a lower output power and a less efficient antenna. Each mobile phone unit consists of a control unit, a radio transceiver, a logic unit, and a mobile antenna. The control unit houses all the user interfaces, including a handset. The transceiver uses a frequency synthesiser to tune into any designated cellular system channel. The logic unit interrupts subscriber actions and system commands and manages the transceiver and control units.

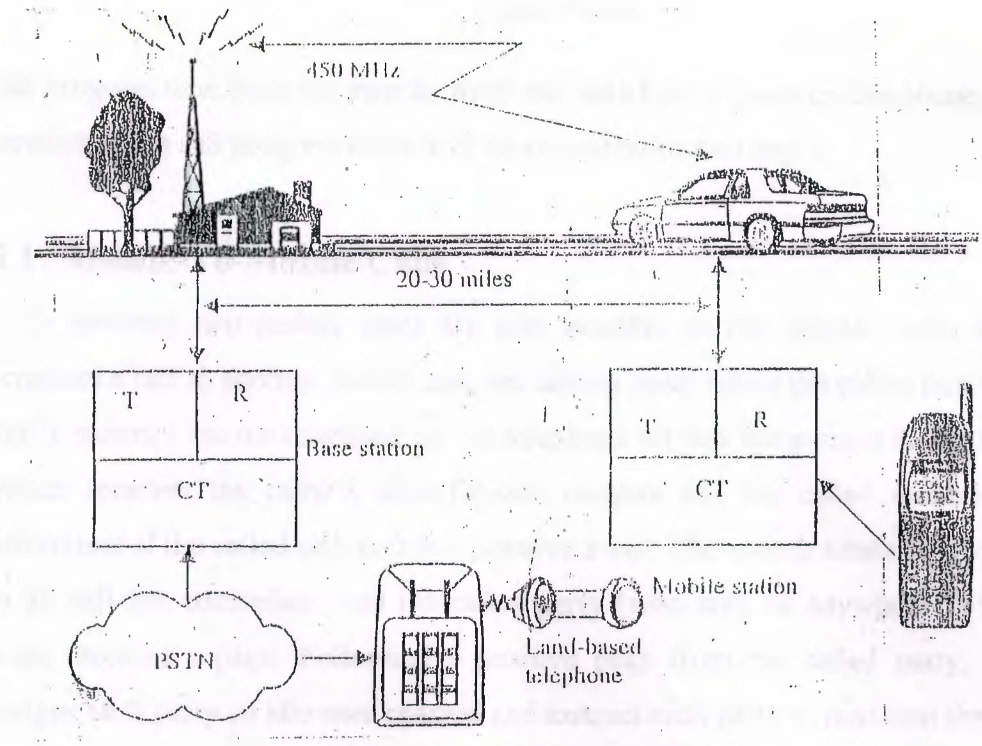


Figure 3.4 Portable Unit Of Mobile Phones

3.9 Wireline-To-Mobile Calls

The cellular system's switching centre receives a from a wireline party through a dedicated interconnect line from the public switched phone network. The switch translates the received dialling digits and determines whether the mobile unit to which the call is destined is on or off hook (busy). If the mobile unit is available, the switch pages the mobile subscriber. Following a page response from the mobile unit, the switch assigns an idle channel and instructs the mobile unit to time into that channel. The mobile unit sends a verification of channel tuning the controller in the cell site and then sends an audible call progress tone to the subscriber's mobile phone. Causing it to ring. The switch terminates the call progress tones when it receives positive indication that the subscriber has answered the phone the conversation between the two parties has begun.

3.10 Mobile-To-Wireline Calls

A mobile subscriber who desires to call a wireline party first enters the called number into the unit's memory using Touch-Tone buttons or a dial on the phone unit. The subscriber then presses a send key, which transmits the called number as well as the mobile subscriber's identification number to the switch. If the identification number is valid, the switch routes the call over a leased wireline interconnection to the public call progress tone from the switch. After the called party picks up the phone, the switch terminates the call progress tones and the conversation can begin.

3.11 Mobile-To-Mobile Calls

Calls between two mobile units are also possible in the cellular radio system. To originate a call to another mobile unit, the calling party enters the called number into the unit's memory via the touchpad on the telephone set and the presses the send key. The switch receives the caller's identification number and the called number and then determines if the called unit is free to receive a call. The switch sends a page command to all cell-site controllers, and the called party (who may be anywhere in the service area) receives a page. Following a positive page from the called party, the switch assigns each party an idle user channel and instruct each party to tune into the respective user channel. Then the called party's phone rings. When the system receives notice that the called party has answered the phone, the switch terminates the call progress tone, and the conversation may begin between the two mobile units. If a mobile subscriber wishes to initiate a call and all user channels are busy, the switch sends a directed retry command instructing the subscriber to reattempt the call through a neighbouring cell. If the system cannot allocates a user channel through the neighbouring cell, the switch transmits a intercept message to the calling mobile unit over the control channel. Whenever the called party is off look, the calling party receives busy signal. Also, if the called is invalid, the system either sends a record message via the control channel or provides and an announcement that the call cannot be processed.

3.12 Advanced Mobile Phone Service

Lyn the AMPS system, the interface between the land phone network and the radio paths to the mobiles occurs at the cell sites. In addition to performing unction needed for trunk termination and for radio transmission and reception, the cell site handles many semiautonomous functions under the general direction of the Mobile Phone Switching Office (MTSO).

Cell sites have facilities to:

Provide RF radiation, reception, and distribution.

Provide data communications with the MTSO and mobiles.

Locate mobiles.

Perform remotely ordered equipment testing.

Perform equipment control and reconfiguration functions.

Perform voice-processing functions.

Perform, call setup, call supervision, and call termination.

Handoff or receive from another cell site any mobile which has moved out of the normal service area of the cell site carrying the call. Programmable controllers control cell-site operations partially by wired logic and partially. Control functions are redundant and can be a configured as needed to overcome a localised failure. A battery plant assures maintenance of service in case of commercial power outage.

Facilities dependent upon traffic requirements in each cell coverage area are modular so those additional units may be installed as needed to match busy-hour traffic levels.

This will ensure that plant investment can grow sensibly as a function of anticipated revenues 48 voice channels. The precise number if a frame at each site is a function of the voice channels requirements for that site. There are four frame codes, and the smallest size cell site requires one of each code. Each radio frame has a maximum capacity of 16 radius. When the number of voice radius grows beyond 16, another radio frame must be added. Each line supervision frame (LSF) can handle 48 voice channels and, when this number is exceeded, another LSF is added. A single data frame (DF) and a single maintenance test frame (MTF) are necessary regardless if the number of voice radius in the cell site. The maximum size of a cell site is 144 voice radius, which would require a total of 14 frames; nine radio frames, three line supervision frames, one data frame, and one maintenance test frame.

Cell-Site Hardware

The hardware facilities of the AMPS cell-site connect the mobile radio customer to the land phone network and perform actions necessary for RF radiation, caption, and distribution; voice and data communications and processing; equipment easting, control, and reconfigration; and call set-up, supervision, and termination. Cell-site operational control is achieved partially through wired logic and partially through programmable controllers. This part describes the cell-site functional groups, their physical characteristics and designed, and the ways they inter/ace with the rest of the AMPS system.

3.13 Data Frame

The data frame contains the equipment for major cell-site control functions, which include communication with the MTSO, control of voice and data communication with mobiles, and communication with the controller in the maintenance test frame communication between controllers is necessary for requesting performance of specific tests and for receiving results. The DF contains both hardware logic and programmable controllers. Only one set of hardware logic and one controller is needed per cell site regardless of the number if voice radius. Because of the critical functions performed in the DF, redundancy if all subassemblies is provided to assure continuation of service in the presence of a failure. The OF can reconfigure itself under the direction of the MTSO, which maintains service by permitting any malfunctioning subassembly to be replaced with an off-line redunant unit. The data frame contains five major subsystems.

3.14 Central Control And Monitoring Site

Illustrates the system design of the CCM, which is based on an pH 2100 microprocessor data- acquisition system. Through software, it emulates radio plan control functions performed by an ess, supervises data gathering and recording and automatically calibrates and monitors the performance of all the Cellular Test Bed's land-based radio components. The CCM interrogates and instructs the mobile unit via telemetry link and the cell sites via specially, conditioned landlines. Operator intervention, if needed is also available. The cell site control message formats, as in the AMPS design, include seven parity bits to ensure high reliability ofdata transmission. The CCM software requests data retransmissions whenever errors occur. The CCM also contains the calibrated audio facilities necessary to conduct voice quality tests.

3.15 The Telemetry Site

The telemetry site (TM) incorporates the radio transceiver facilities, which permit the CCM to reliably instruct and interrogate the MCL. Anywhere within the CTB test probe area. To meet the transmit/receive path reliability requirements of this important radio link, the TM site is centrally located within the probe area and uses a high-gain transmits and diversity-receive antenna system elevated 230 feet above the local street surface. The TM site also incnipcirates voice communication facilities to administer test operations.

3.16 Mobile Communications Laboratory

The interior of the MCL. Contains radio, logic. Miniprocessor, and data-recording facilities. The RF/analog subsystem which consists of five measurement channels driven by two electronically selectable RF preamplifiers fed from two receive antennas appropriately paced for diversity reception, is illustrated. The same antennas and preamplifiers also feed the AMPS mobile radio used to evaluate the performance of the voice and signalling subsystems.

The main measurement receiver uses a computer-controlled agile local oscillator, which mixes the RF signals down to three intermediate frequencies. Each of these frequencies feeds into two highly selective channels that use logaithmic detectors. Two channels (one high-gain, one low-gain) service each IF signal to achieve an instantaneous dynamic range that is linear from –150 to –30 dBm. The two channels are adjusted to maintain a 20-dB overlap centred at –90 dBm. The measurements for calculating real-time average values are selected using either the high-gain measurement or by, accepting the low-gain result it exceeds a threshold approximately in the middle of the Overlap region.

Environmental noise is monitored on one antenna by a single logia-rhythmic detector with a linear range from –150 to –10 dBm. The output of the diversity switch in the mobile radio is measured by an eighth logarithmic detector having a linear range from –120 to –40 dBm, with the useful range extending nearly 10 dB more at each end.

Instantaneous data sampled from these receivers are processed to obtain a true incident power by a stored program reference tabulation, this processing translates the output from a 10-bit analog-to-digital converter to a number proportional to the corresponding instantaneous input signal power.The instantaneous signal power samples are summed over one-half second of real-time to calculate average values.

The MCL is also wquipped with a gyroscopic-bearing and distance-tracking system so that all system status and measurement information recorded each one-half second are tagged with true vehicle position.

3.17 CTB Calibration And Performance Monitoring

The calibration and performance-monitoring equipment in the CTB's hardware and software designs and the subsequent off-line statistical processing of the measurement data can precisely control and qualify the field experiments to obtain results comparable in resolution and reliability to those achieved in the laboratory. Examples of the call-ration and performance monitoring subsystems incorporated within the central and interferer cell sites the MCL uses a similar calibration and monitoring system.

The cell sites transmit calibration and monitoring subsystem monitors, via precision coupler and temperature-compensated detection circuits the RF power incident to and reflected from each antenna/cable assembly. The detected voltages, sampled and processed by the PROCON, are sent to the CCM, where they are monitored and recorded (on-line) to insure the integrity of the cell site transmit function.

The type of calibration and monitoring subsystem used in the cell site receivers is illustrated. In practice, the test generator is set, under CCM control, to a reference power level. The CCM then (via land lines and the PROCON) automatically steps a programmable, precision attenuator to supply the input reference signals necessary to calibrate the cell site instrument level, 1000 samples are taken and averaged to generate stored program reference tabulations which, during real-time data acquisition, are used to determine the true instantaneous signal strength incident at the terminals of the receive antenna. The test generator also furnishes a reference signal to each antenna and cable subsystem. The instrumentation receiver monitors the forward and reflected power to ensure that antenna system returns requirements are met. As shown, the test generator subsystem also furnishes the reference signal necessary to establish the FM quieting performance of the AMPS radius. Calibration of the CTB's transmit-and-receive subsystems is maintained within $\pm \frac{1}{2}$ dB during each field evaluation sequence. The calibrations are performed at least before and after each test sequence and are hardcopies as part of the data package.

3.18 Control/Recording Architecture

This section describes the system control and data-recording structures of the CTB that perform the AMPS emulation and data-acquisition functions. As noted previously, an extensive data of transmission parameters is established at the CCM every data frame. The algorithmic software module accesses the appropriate cell site transmission data are communicated to cell site and implemented by the operating system. The following paragraphs discuss the communication, control, measurement, and data-recording aspects of CTB operation.

3.19 CTB Data Communication

As described earlier, the CTB, which is linked with cell, sites by data lines and to the MCL by a full-duplex telemetry channel. These interconnections, together with powerful processing capability at each remote site, form a comprehensive data communications structure.

Basically, three types of message are used for are used for data communications within this field configuration: First, control messages, such as signalling requests to cell sites, permit the execution of system-level operations. Second special data acquisition requests and data messages to and from cell sites and the MCL permit the acquisition of data at the CCM.

Third, CTB operational-control messages permit the automatic calibration ofcell sites, synchronise the data-acquisition frame at each cell site status information onthe proper performance of the system. The last category of messages allows direct CCM instructions to the mobile logic unit via telemetry link and also permits the MCL and CCM operators to request test pauses.

The land-line messages are transmitted at a rate of 2400b/s, while the MCL data transfer rate is 1800b/s. All messages are formatted into 32-bit blocks with seven bits devoted to error control. The data are encoded in a shortened (127,120) Bose-Chaudhuri-Hocquenghem(BCH) code, which is used in an error-detection mode with retranmission.

a) System Control

The CTB configuration must be properly initialised to start data acquisition. First, the interferer transmitters and the main cell site instrumentation receivers must be tuned to the serving channel. Then the test can start by synchronising the data-acquisition frames at each cell site and the MCL with the CCM system clock. From that point on microscopic data measurement at the MCL and cell sites depend on their local clocks. The CCM data-collection subsystem initiates each frame with "request-for-data" messages to the cell sites and the MCL. The data received are checked and formatted by a CCM software module and placed in a buffer to the system-control algorithmic module. This module is coded so that it can access data variable to the AMPS control algorithms only at the proper time interval. The output of the module by requires a system reconfiguration, which is accomplished by the CCM with appropriate data-link messages. All system decisions, requests for action, and actions, are recorded with the underlying data for later analysis.

3.20 Measurement Of RF Transmissions Parameters

Radio transmissions parameters are measured at each of the cell sites and at the MCL. Each cell site instrumentation receiver switches sequentially to each of eight RF channels for sampling the mobile carrier level as received on each of two omnidirectional and three pairs of directional antennas. The data-sampling rate is 512 Hz enabling the acquisition system to make 64 measurement per channels each data frame. The samples are processed through a calibration stored-program reference tabulation to generate quantities proportional to the RS signal as receiver at the antenna terminals.

The cell site programmable controller than forms eight averages from these samples every data frame. If we assume an underlying Rayleigh distribution, these averages estimate the local means within a 95% confidence interval of approximately 1 dB. They are eight averages together with the final eight instantaneous samples from the RF parameter list, which is transmitted to the CCM, every data frame and recorded on digital tape in the formatted.

b) MCL activities

The MCL is a highly sophisticated data acquisition facility. Its five basic measurement channels are alternately switched to two diversity-receiving antennas. Further, measurement are made on both the high-and-low-gain if channels with the MCL computer selecting the proper value in real-time. Measurements are made on setup, voice, interferer and noise channels. In addition, the AMPS diversity signal and peak-noise distributions are measured.

4. BASE STATION

4.1 The Base Station Subsystem

The BSS connects the Mobile Station MS and the MSC. 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 and The Base Station Controller (BSC).

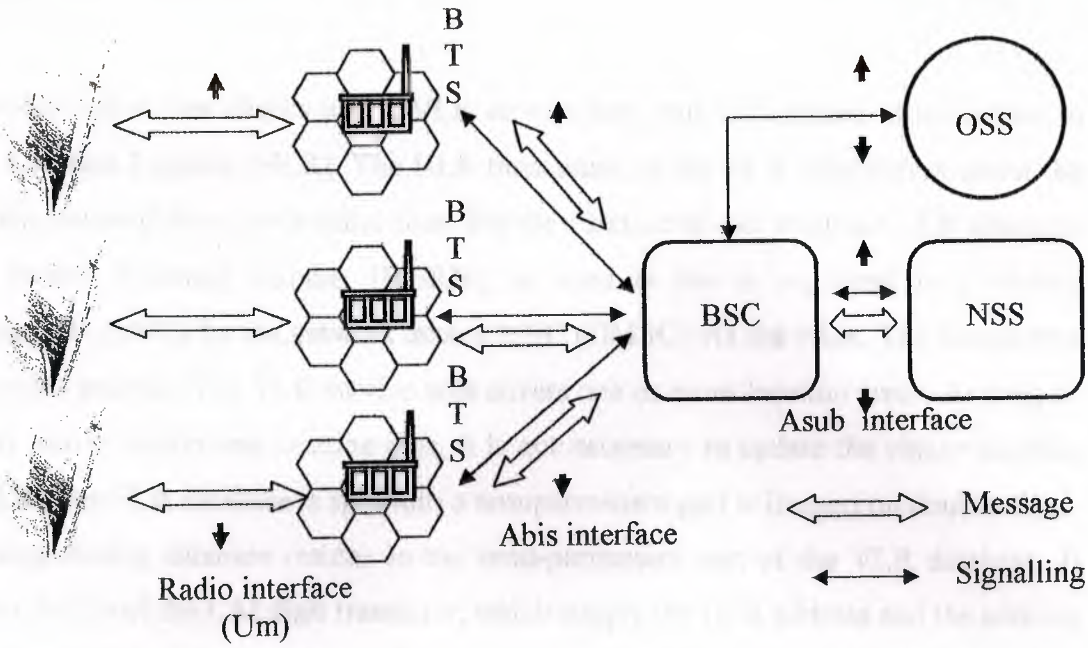
4.1.1 The Base Transceiver Station

The BTS corresponds to the transceivers and antennas used 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.

4.1.2 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. BSC can act as a concentrator for the links between the Abis and Asub interfaces.

The BSC involves a separate Transcoding and Rate Adaptation Unit (TRAU) for speech coding and data rate adaptation. The BSS components and interfaces are shown in Figure.



4.2 Network Switching Subsystem (NSS)

Mobile switching centre (MSC) is a stored-program controlled digital switching centre. The MSC is the switching centre in the PLMN, which acts as a gateway to other networks is linked to other MSCs in the PLMN, connects the network elements of the NSS with the network elements of the BSS in the coverage area of the PLMN. The MSC has functions that are familiar from the switching centres of the fixed networks as well as special functions that are not necessary in the switching centres of the fixed networks.

4.3 Visitor Location Register (VLR)

The VLR is essentially a database that holds all information on those mobile subscribers currently roaming in the VLR area it controls. The VLR can recognize a mobile subscriber by the following identifiers:

- the international mobile subscriber identification (IMSI);
- the local mobile subscriber identification (LMSI);
- mobile station roaming number (MSRN)

- the temporary mobile station identify (TMSI) together with the local area identity (LAI)

When a mobile subscriber checks into a VLR service area, this information is forwarded to his Home Location Register (HLR). The HLR then sends to the VLR information about the authorization status of this mobile subscriber. For the duration of call setup the VLR allocates a Mobile Station Roaming Number (MSRN); as soon as this is requested in a Mobile Terminating Call (MTC) by the network access MSC (GMSC) via the HLR. The connection is setup via this number. The VLR service area covers one or more location areas. As long as an MS only moves within one location area, it is not necessary to update the visitor location register VLR. The VLR database is split into a semipermanent part is imaged on double disks. The signaling-routing database resides in the semi-permanent part of the VLR database. It contains the IMSI and the LAI digit translator, which supply the HLR address and the address of the previous VLR. The national roaming database stores the data of the VLR areas in which the mobile subscriber is permitted to set up a connection. The mobile subscriber database resides in the transient part of the VLR database. It contains the call processing data of the mobile subscribers currently roaming in this area. Its memory is allocated dynamically and separately for each mobile subscriber. The data are distributed in several pools, e.g.:

- In the common data pool with IMSI, ISDN; TMSI, LAI and the registered services;
- In the basic telecommunications data pool with the registered and activated supplementary services;
- Closed User Group (CUG) data pool (e.g. CUG index)

Another transient database contains the temporary mobile subscriber identities (TMSI). With these an individual mobile subscriber is addressed and identified. The VLR database contains the current ciphering key (KC) and the ciphering key sequence number sent to the MS during authentication.

4.4 Home Location Register (HLR)

The HLR contains the main database of the mobile subscribers. The database entries may be generated, deleted and read by the PLMN operator, remotely by an OSS or by a Personalization Centre (PCS), for SIM) via the OMS or on the local OMT. By Subscriber

Controlled Input (SCI), the mobile subscriber can also remotely input specific subscriber data (for supplementary services). At call setup, the HLR can identify a mobile subscriber with the aid of the International Mobile Subscriber Identifier (IMSI) and By International Mobile Subscriber Identifier (MSISDN). The HLR participates in setting up a mobile terminating call (MTC). On setup of an MTC the HLR is requested by the network access MSC (GMSC), to retrieve the mobile subscriber roaming number (MSRN) of the mobile subscriber from the current VLR. The HLR does this and sends the MSRN to the GMSC. During a location update the HLR supports the current VLR of the mobile subscriber by supplying the necessary data, and the VLR in turn supplies its VLR address. The HLR database contains both semipermanent and transient data. The semipermanent data include: HLR mobile subscriber data and signaling data (network data of the HLR). The transient data include: HLR mobile subscriber data and traffic measurement data. The semipermanent HLR mobile subscriber data are split into the following data modules and tables:

- Common data module;
- Basic and supplementary services data modules;
- MSISDN and bearer capability data module;
- CUG data module;
- GSM bearer capability information element (BCIE);
- VLR roaming table (regional VLR roaming);
- IMSI exchange table.

The transient HLR mobile subscriber data are split into the following data modules:

- Mobility data module (e.g. authentication data, MSRN, relation to VLR address and Local Mobile Subscriber Identifier (LMSI), detach from the IMSI;
- Short message waiting data module.

4.5 Authentication Centre (AC)

The AC is equipped with several security boxes, in which the authentication keys and algorithms required for generation of the authentication parameters of a mobile subscriber are stored. In the AC for each mobile subscriber a number of authentication parameters RAND (random number), authentication response (SRES, signed response) and Kc (cipher key) are generated, before the mobile subscriber obtains access to the network. The authentication parameters are used by the VLR for authentication tests, i.e. to determine whether a mobile subscribes authorized for access to the network and call setup.

4.6 The Equipment Identity Register (EIR)

The EIR is also used for security purposes. The EIR is a database with information about the equipment types and equipment identities of all mobile radio equipment authorized in its service area. More particularly, it contains a list of all valid terminals. The functions belonging to the EIR perform the equipment identification. A terminal is identified by its International Mobile Equipment Identity (IMEI). The EIR allows then to forbid calls from stolen or unauthorized terminals.

4.6.1 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 altered.

4.7 The Operation and Support Subsystem.

The OSS is connected to the different components of the NSC 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. 1 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 f transfer decreases considerably the costs of the maintenance system.

4.8 The GSM functions

In GSM are defined five main functions: Transmission, Radio Resource Management, (RRM), Mobility Management (MM), Operation, Administration and Maintenance (OAM). The transmission functions include two sub-functions: the transmission of user information and transmission of signaling information. Not all component of the GSM network are strongly related with the transmission functions. The MS, BTS and BSC, are deeply concerned with transmission. But other components, such as the register HLR; VLR or EIR are concerned for signaling need.

The role of RRM is to establish, maintain, and release communication links between MS and BSS. RR function is also in charge of maintaining a connection even if user moves from one cell to another. The RR is also responsible for the management of the frequency spectrum and the reaction of the network to changing radio environment conditions.

The most important responsibility .of RR is handover described below.

4.8.1 Handover

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

a) Handover of channel in the same cell,

b) Handover of cells controlled by the same BSC;

c) Handover of cells belonging to the same MSC but controlled by different types BSCs;

d) Handover of cells controlled by different MSCs;

Handover is mainly controlled by the MSC. However to avoid unnecessary signaling information, the first two types of handovers are managed by the BSC.

4.8.2 Mobility Management

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

4.8.3. 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. When a 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 MSCNLR, which gives the location information to the subscriber's HLR. If the mobile station is authorized in the new MSCNLR, the subscriber's HLR cancels the registration of the mobile station with the old MSCNLR. 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.

4.9 Authentication and Security

The SIM card and the Authentication Centre are used for the authentication procedure. A secret key, stored in the SIM card and the AC, and a ciphering algorithm is used in order to verify the authenticity of the user. The mobile station and the AC compute a SRES (Signed Results) using the secret key, the algorithm A3 and a random number generated by the AC. 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.

4.10 Satellite Communications

The unique feature of communications satellites is their ability to link simultaneously all the users on the earth's surface, thereby providing distance insensitive point-to-multipoint communications. This capability applies to fixed terminals on earth and to mobile terminals on land, in the air and at sea. These features make satellite communications systems unique in design. In 1945 Arthur Clarke wrote that a satellite with a circular equatorial orbit at a correct altitude of 35, 796 km would make one revolution every 24 hour; that is, it would rotate at the same angular velocity as the earth. An observer looking at such a geostationary satellite would see it hanging at a fixed spot in the sky. Clarke showed that three geostationary satellites powered by solar energy could provide worldwide communications for all possible types of services.

4.10.1 Frequency Allocations for Satellite Services

Allocating frequencies to satellite services is a complicated process, which requires international coordination and planning. This is carried out under the auspices of the International Telecommunication Union ITU. To simplify frequency planning, the world is divided into three regions: *Region* 1: Europe, Africa, NIC (New Independent Contries), and Mongolia; *Region* 2: North and South America and Greenland; *Region* 3: Asia (excluding region 1 areas), Australia, and the Southwest Pacific. Within these regions, frequency bands are allocated to various satellite services, although a given service may be allocated different frequency bands in different regions. Some of the services provided by satellites are: Fixed satellite service (FSS), Broadcasting satellite service (BSS), Mobile satellite service (MSS); Navigational satellite service and Meteorological satellite service Table 4.1 lists the frequency band designations in common use for satellite services. The frequencies, used for satellite communications are allocated in SHF and EHF bands. The Ku band signifies the band under the K band, and the Ka band is the band above the K band. The Ku band is the one used at present for direct broadcast satellites and it is also used for certain fixed satellite services.

Frequency range, GHz	Band designation
0.1- 0.3	VHF
0.3 -1.0	UHF
1.0 - 2.0	L
2.0 - 4.0	S
4.0 - 8.0	C
8.0 - 12.0	X
12.0 -18.0	Ku
18.0 - 27.0	K
27.0 - 40.0	Ka
40.0 -100.0	Millimetre

Table 4.1 Frequency Band Designations

The C band is used for FSS, and no direct broadcast services are allowed in this band. The VHF band is used for certain mobile and navigational services and for data transfer from weather satellites. The L band is used for MSS and navigation system. For the FSS in the C band, the most widely used subrange is approximately 4 to 6 GHz. The higher frequency is nearly always used for the uplink to the satellite, for reasons which will be explained later, and common practice is to denote the C band by 6/4 GHz, giving the uplink frequency first. For the direct broadcast service in the Ku band the most widely used range is approximately 12 to 14 GHz, which is denoted by 14/12 GHz. Although frequency assignments are made much more precisely, they may lie some what outside the values quoted here (an example of assigned frequencies in the Ku band is 14,030 and 11,730). The approximate values stated above are quite satisfactory for use in calculations involving frequency, as wil1 be shown later in the text.

4.10.2 Satellite Systems

A satellite system consists basically of a satellite in space, which links many earth stations on the ground, as shown schematically in Figure 4.1. The user generates the baseband signal, which is routed to the earth station through the Terrestrial Network (TN). The terrestrial network can be a telephone switch or a dedicated link to the earth station (ES). At the earth station the baseband signal is processed and transmitted by a modulated radio frequency (RF) carrier to the satellite. The satellite can be thought of as a large repeater in space. It receives the modulated RF carriers in its uplink (earth-to- space) frequency spectrum from all the earth stations in the network, amplifies these carriers, and retransmits them back to the earth in downlink (space-to-earth) frequency spectrum which is different from the uplink frequency spectrum in order to avoid interference. The receiving earth station processes the modulated RF carrier down to the baseband signal, which is sent through the terrestrial network to the user. Most commercial communications satellites today utilize a 500-MHz bandwidth on the uplink and a 500-MHz bandwidth on the downlink. The most widely used frequency spectrum is

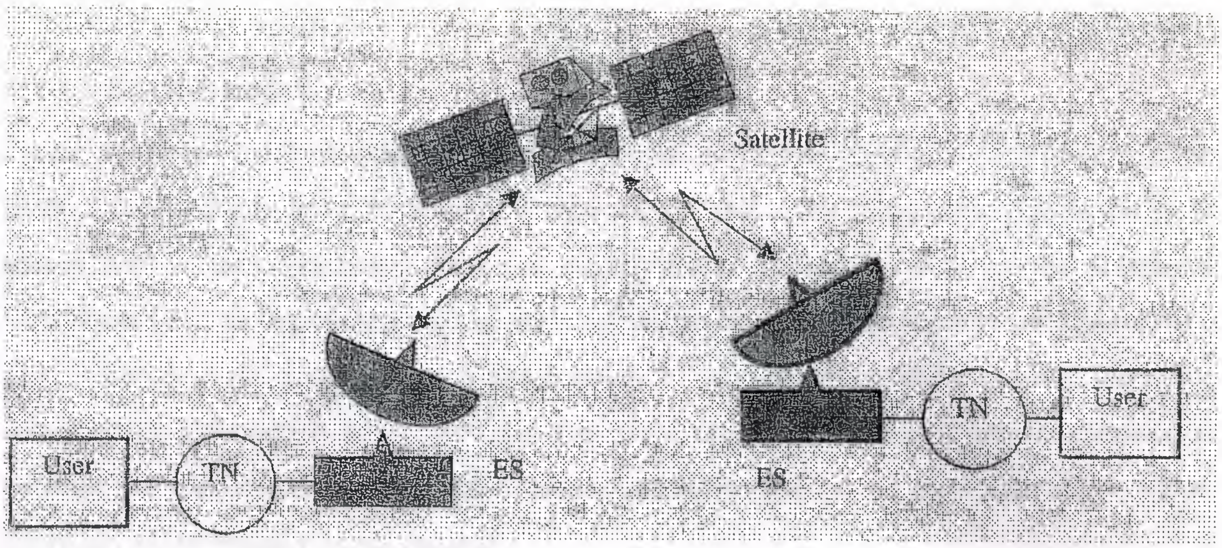


Figure 4.2 Satellite System

the 6/4-GHz band, with an uplink of 5.725 to 7.075 GHz and a downlink of 3.4 to 4.8 GHz. The 6/4-GHz band for geostationary satellites is becoming overcrowded because it is used by common carriers for terrestrial microwave link. New systems are now being operated in the 14/12-GHz band using an uplink of 12.75 to 14.8 GHz and a downlink of either 10.7 to 12.3 GHz or 12.5 to 12.7 GHz. The 14/l2-GHz band will be used extensively in the future and it has not been yet congested, but one problem exists-rain, which attenuates 14/l2-GHz signals much more than it does those at 6/4 GHz. The frequency spectrum in the 30/20-GHz band has also been set for commercial satellite communications; with a downlink of 18.1 to 21.2 GHz and an uplink of 27.5 to 31 GHz. Equipment for the 30/20-GH 1 band is still in the experimental stage and is expensive. The typical 500-MHz satellite bandwidth at the 6/4 and 14/l2-GHz bands can be segmented into many satellite transponder bandwidths. For example, eight transponders can be provided, each with a nominal bandwidth of 54 MHz and a center-to-center frequency spacing of 61 MHz. Modem communications satellites also employ frequency reuse to increase the number of transponders. E Frequency reuse can be accomplished through orthogonal polarization's where one transponder operates in one polarization (e.g., vertical polarization) and a cross-polarized transponder operates in the orthogonal polarization (e.g., horizontal polarization). With orthogonal polarization's a satellite can double the number of transponders in the available 500-Mhz, bandwidth, hence double its capacity.

4.10.3 Earth station

Figure 4.2 shows the functional elements of a digital earth station. Digital information in the form of binary digits from the terrestrial network TN enters the transmitter side of the earth station and is then processed (buffered, multiplexed, formatted, etc.) by the Baseband Equipment (BE) so that these forms of information can be sent to the appropriate destinations. The presence of noise and the non ideal nature of any communication channel introduce errors in the information being sent and thus limit the rate at which it can be transmitted between the source and the destination. Users generally establish an error rate above which the received information is not useable.

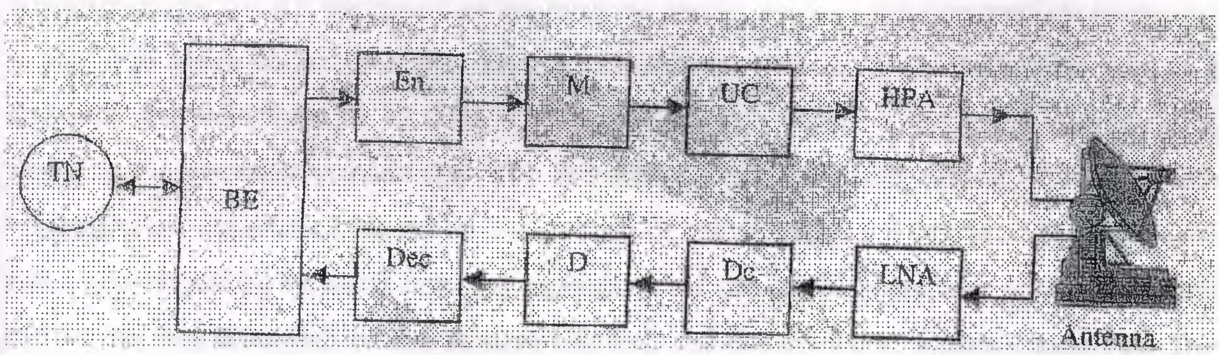


Figure 4.3 Earth Station

If the received information does not meet the error rate requirement, error-correction coding performed by the-encoder (En) can often be used to reduce the error rate to the acceptable level. In order to transmit the digital information over a satellite channel that is a bandpass channel, it is necessary to transfer the digital. Information to a carrier wave at the appropriate bandpass channel frequency .This procedure is performed by modulation. The function of the modulator (M) is to accept the symbol stream from the encoder (En) and use it to modulate an intermediate frequency (IF) carrier. in satellite communications, the IF carrier frequency is chosen at 70 MHz for a communication channel using a 36-MHz transponder bandwidth and at 140 MHz for a channel using a transponder bandwidth of 54 or 72 MHz. A carrier wave at an intermediate frequency rather than at the satellite RF uplink frequency is chosen because it is difficult to design a modulator M that works at the uplink frequency spectrum (6 or 14 GHz, as discussed previously). For binary modulation schemes, each output digit from the encoder is used to select one of two possible waveforms. For M-ary modulation schemes, the output of the encoder is segmented into sets of k digits, where M = 2k and each k-digit set or symbol is used to select one of the M waveforms. The modulated IF carrier from the modulator is fed to the Upconverter (Uc), where its intermediate frequency is translated to the uplink RF frequency. The high- Power Amplifier (HPA) then amplifies this modulated RF carrier by the antenna. The earth station antenna provides the transmitting modulated RF carrier too the satellite within the uplink frequency spectrum and receiving the RF carrier from the satellite within the downlink frequency spectrum. On the receiver side the earth station antenna receives the Low-Level modulated RF carrier in the downlink frequency spectrum of the satellite. A Low-Noise Amplifier (LNA) is used to amplify this low-level RF carrier to keep the carrier-to-noise ratio at a level necessary to meet the error rate requirement. The Downconverter (Dc) accepts the amplified RF carrier from the output of the low- noise amplifier and translates the downlink frequency to the intermediate frequency. The reason for downconverting the RF frequency of the received carrier wave to the intermediate frequency is that it is much easier to design the demodulator to work at 70 or 140 MHz than at a downlink frequency of 4 or 12 GHz. The demodulated IF carrier is fed to the demodulator, where the information is extracted. The demodulator (D) estimates, which of the possible symbols were transmitted, based on observation of the received I F carrier. The probability that a symbol will be correctly detected depends on the carrier- to-noise ratio of the modulated carrier, the characteristics of the satellite channel, and the detection scheme employed. The decoder performs a function opposite that of the encoder. Because the sequence of the symbols recovered by the demodulator may contain errors, the decoder must use the uniqueness of the redundant digits introduced by the encoder to correct the errors and recover information-bearing digits. The information stream is fed to the baseband equipment for processing for delivery to the terrestrial network. In the United States the Federal Communications Commission (FCC) assigns orbital positions for all communications satellites to avoid interference between adjacent satellite systems. Before 1983 the spacing was established at 4° of the equatorial arc, and the smallest earth station antenna for a simultaneous transmit-receive operation allowed by the FCC is 5 m in diameter. In 1983, the FCC ruled that fixed service communications satellites in the geostationary orbit should be spaced every 2° along the equatorial arc instead of 4°. This closer spacing allows twice as many satellites to occupy the same orbital arc.

4.10.4 Satellite orbits

As shown in Figure 4.3, there are three basic types of satellite orbits: Geostationary Earth Orbits c (GEO), Medium Earth Orbits (MEO), and Low Earth Orbits (LEO).

a) Geostationary Earth Orbit (GEO)

Most of the current satellites in operation fall into this category. This orbit is approximately 35, 786 km above the earth. The terms geosynchronous and geostationary are often used interchangeably, but there is an important difference between them.

Geosynchronous satellites have an orbit whose period is one sidereal day or 24 hours. However, due to the earth's revolution around the sun, the actual period is slightly shorter: 23 hour, 56 min, and 4.1s. Obviously, the satellite must also be in a direct orbit, that is, the satellite must move in the same direction as the rotation of the earth. The inclination of a geosynchronous satellite's orbit may be at any angle with respect to the earth's equatorial plane. A truly geostationary satellite also has the same period and the same direction of rotation as the earth. However, it must have an orbit that is close to the equatorial plane of the earth, that is, it must have a zero inclination. An observer looking at such a geostationary

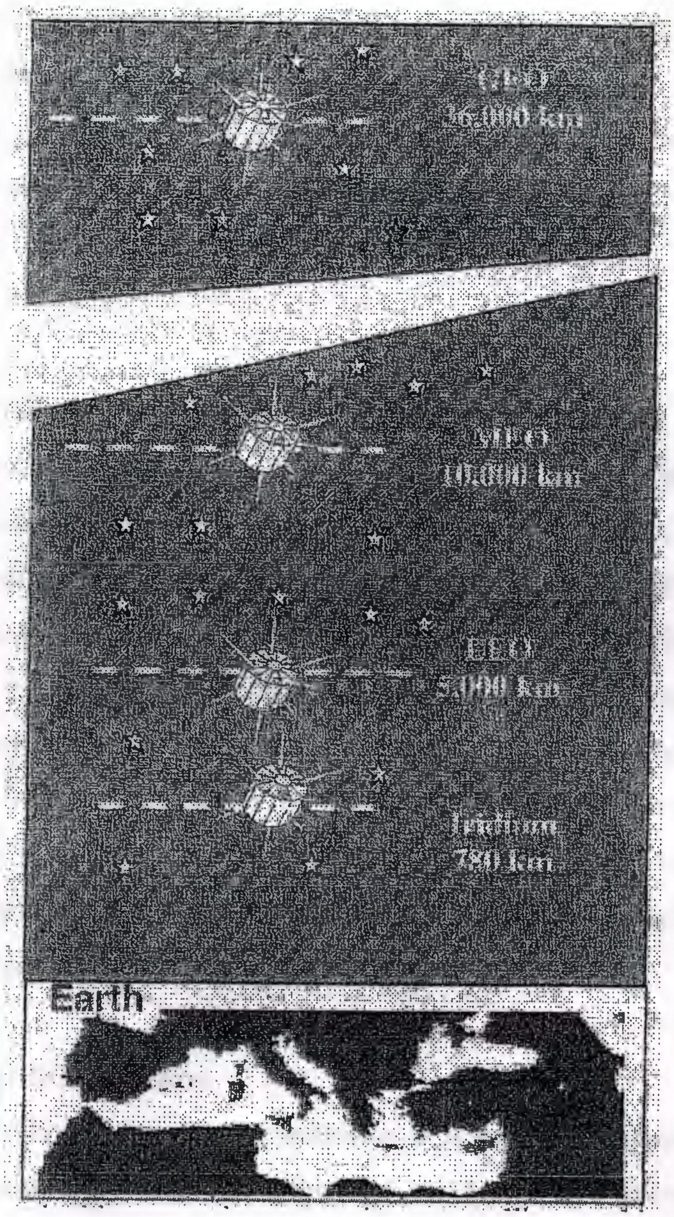


Figure 4.4 GEO

satellite would see it hanging a perfectly fixed spot in the sky. But this is all relative. An observer in space sees a geostationary satellite orbiting the earth at a speed of 11.068.8 km/h. The round-trip propagation delay for GEO link is about 260 ms. The geostationary orbit is now employed for most commercial satellites because of the following advantages:

- The satellite remains stationary with respectect to one point on earth; therefore the earth station antenna is not required to track the satellite periodically. This reduces the station's cost considerably;
- With a 5 minimum elevation angle of the earth station antenna, the geostationary satellite can cover almost 38% of the surface of the earth. Three geostationary

satellites (1200 apart) can cover the entire surface (except for the polar regions above latitudes 76° N and 76° *S).*

- The Doppler shift caused by a satellite drifting in orbit (because of the gravitational attraction of the moon and the sun) is small for all the earth stations within the geostationary satellite coverage. This is desirable for many synchronous digital systems.

b) Medium Earth Orbit (MEO) Satellites

The medium earth orbits are approximately at about 10,000 km above the earth. Their space-earth transmission loss is much less than for geostationary satellites, and the round-trip transmission is reduced to 100-150 ms. Circular MEO orbits have periods in the range 8-12 h. As a result of the lower orbit, they do not travel at the same speed relative to the earth. This introduces the need for several MEO satellites in order to provide continuous coverage. Although geostationary satellites seem likely to dominate satellite communications with high-speed link between fixed points, lower transmission loss of MEO satellites make them particularly attractive for mobile-satellite systems because hand-held terminals with much lower power and simple omnidirectional antennas can be used. An example of a MEO system is the proposed ICO system of ICO Global Communications.

c) Low Earth Orbit (LEO) Satellites

A low earth orbit would provide a further reduction in space-earth transmission loss relative to the geostationary orbit and transmission times of 20-25 ms. This allows the use even low power, handheld terminals in MSS.

Altitudes between 780 and 1400 km are favored, corresponding with orbital periods between 100 and 113 min. Thus, LEO systems require slightly more satellites than MEO systems to provide continuous coverage. For example, 66 satellites are used in the Iridium system. Examples of LEO systems are Iridium Inc.'s "Iridium", Loral-Oualcomm's "Globalstar".

d) Frequency Bands

Spectrum in several bands has been allocated internationally for Fixed Satellite Service (FSS) and MSS systems:

- Below 1 GHz for "Little LEO" MSS systems;
- Around 1.6 GHz and 2.4 GHz for "Big LEO" MSS systems;
- Around 2 GHz for future personal communications services (PCS) MSS systems;
- Around 4 and 18 GHz for FSS systems.

4.10.5 The Iridium- 66 Constellation

In June 1990, experts at Motorola Inc. unveiled their blue-sky vision for a futuristic new constellation of mobile communication satellites. Their initial architectural approach called for seven slender rings of satellites marching in single file up over the North and South Poles with 11 satellites in each circular ring (see Figure 4.4 (a)).

Motorola engineers decided to name their mobile communication system "Iridium" because their 77-satellite constellation was in direct analogy with the 77 electrons circling around Iridium's nucleus. (Iridium is one of the platinum metals, a precious silver-white substance harder than iron, nearly as brittle as flint glass, and denser than copper or brass. Iridium alloys are sold as jewellery). Unfortunately, when they later decided to increase the number of spot-beams and the transmitter power of each satellite, they also decided to reduce the number of satellites to only 66.

With a full load of fuel, an Iridium satellite weighs about 690 kg.

In space, a pair of wings with gallium arsenide photovoltaic solar cells deploys, and altitude is stabilized by a three-axis momentum-wheel control system. Gallium arsenide has been replacing traditional silicon in photovoltaic cells in space because of its superior efficiency

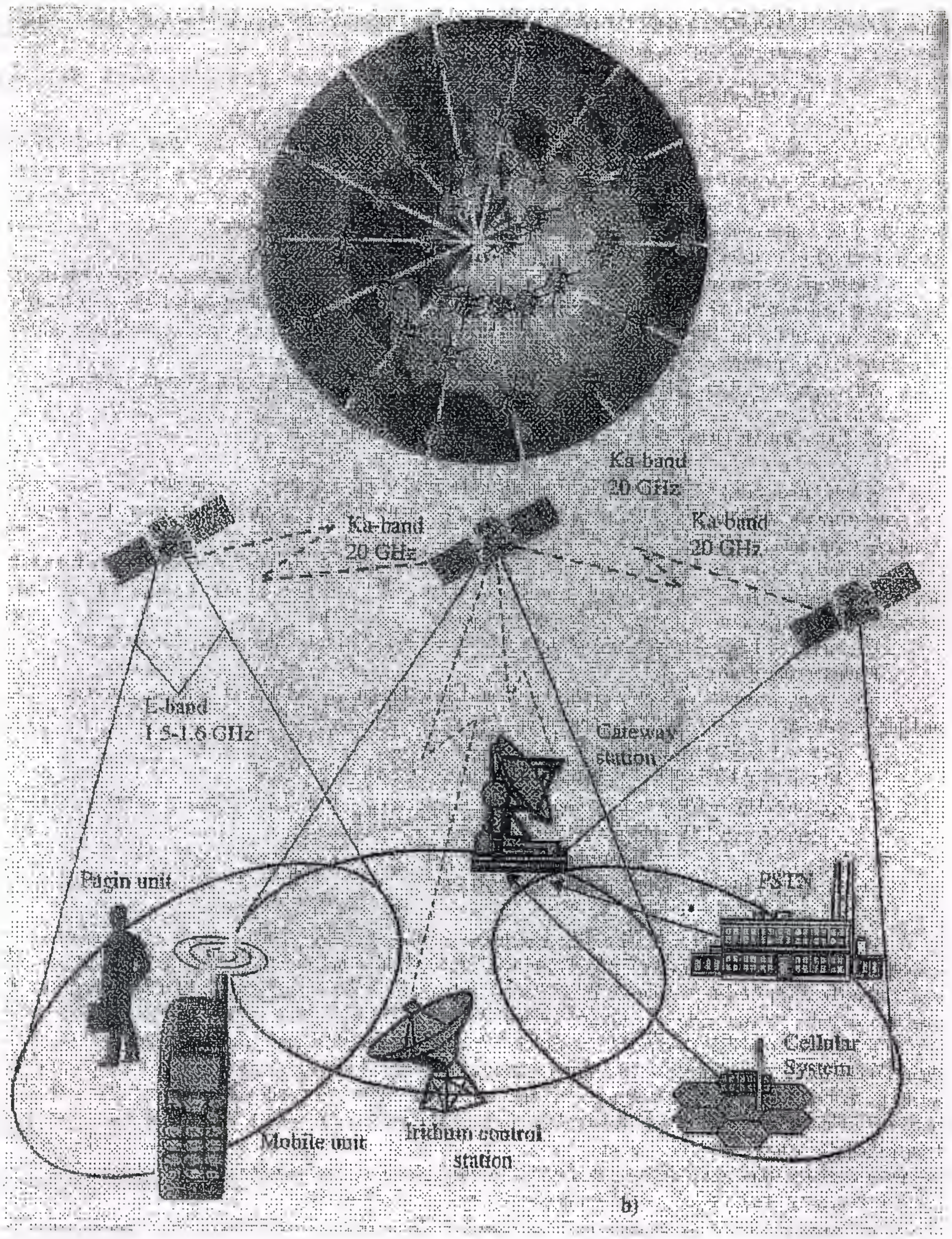


Figure 4.5.a The Iridium-66 Constellation

yielding about one-third more power for comparable cell areas.

A trio of phased-array antennas extends and points earthward to establish direct links over the 1.610-1.625-GHz band to Iridium subscribers. The Iridium constellation, with a company-projected price tag of $3.4 billion, is one of the most costly concepts ever devised for providing mobile communication services. Each satellite in the Iridium constellation will send out 48 pencil-thin spot-beams each of which can handle 230 simultaneous duplex conversations. Iridium satellites are distributed among six evenly spaced, near-polar orbits

(86.4 degrees inclination) 780 km above the earth, sixty of the satellites provide overlapping global coverage, Polar Regions included. The other six are in-orbit spares. Iridium subscriber equipment offer voice, data, paging, and facsimile services see Figure 4.5 (b)

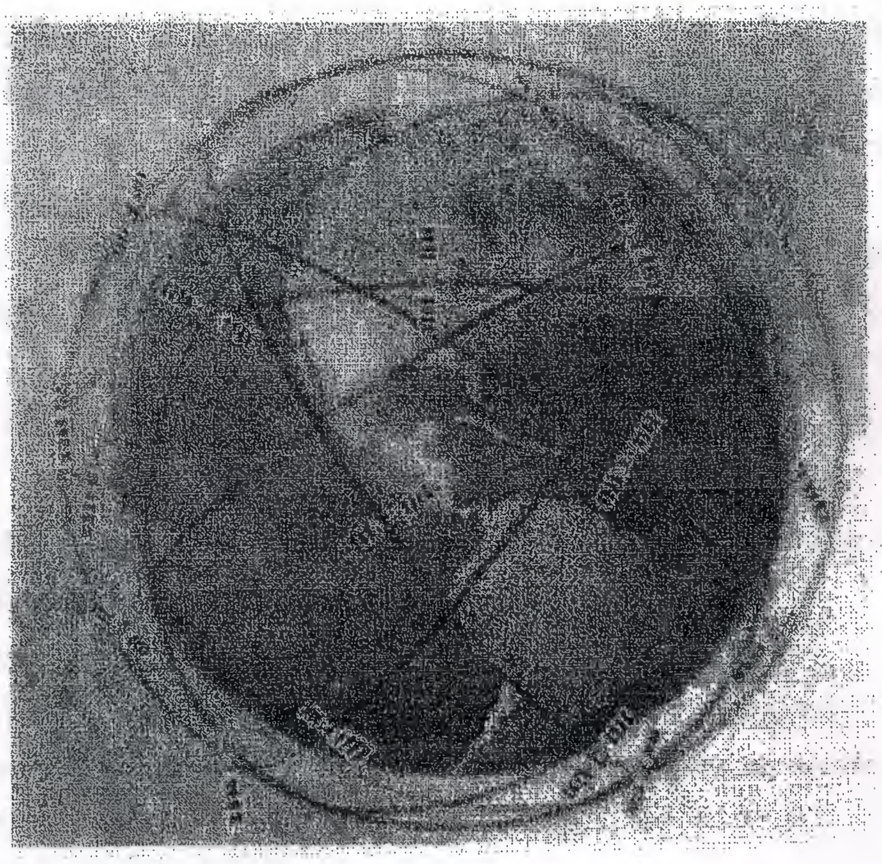


Figure.4.5.b The Globalstar-48 Constellation

A call placed by an Iridium subscriber to another subscriber is transmitted directly by satellite to its destination worldwide; it is the only worldwide system to do this. If the call is to a party with a conventional fixed or mobile phone, it will be upconverted and transmitted by a feeder link from the satellite to a gateway. From there it is routed through the public switched telephone network to its destination. When an Iridium communicator is activated, the nearest satellite (working in concert with the ground-based iridium network) ascertains the validity of that subscriber's account, and then determines the location of the user. The system automatically checks to see if an inexpensive terrestrial link is available to handle the call. If not, the call is relayed through the nearest satellite and, if necessary, from satellite to satellite to its destination. If an Iridium subscriber is at a remote location, the call will be transmitted directly to the intended recipient. If the subscriber is in the vicinity of a land-based telecommunication system, conventional terrestrial communication channels will be used instead.

The satellite-to-satellite cross links, the satellite-to-iridium gateway stations and downlinks connecting the Iridium satellites with their ground-based system control stations are provided using Ka-band at 20 GHz. The transmission links connecting the hand-held communicators, the paging units, and the remote area telephones will all be handled with the L-band frequencies between 1.5 and 1.6 GHz. Iridium employs CDMA modulations and TDMA architecture. This approach will require that a dedicated portion of the frequency spectrum be allocated to Iridium to provide interference-free operation.

Iridium's transmission rates have been set at 4800 bps for voice, and both 4800 and 2400 bps for digital data transmissions.

Characteristics of Iridium

1. Eleven orbital planes of six satellites each with al, 414 km circular orbit inclined at 86.4°.
2. Each satellite mass, approximately 690 kg;
3. 0rbital period -100 minute;
4. Electrical power; 2 sun collected solar arrays with sun tracking solar panel;
5. Antennas: satellite antenna provides 48 spot beams each can handle 230 duplex conversations
6. Frequency bands

Ka-band (20 GHz) -satellite-to-satellite; satellite-to-gateway and satellite-to-control stations;

L-band (1.5-1.6) GHz -direct, with subscribers;

The Iridium constellation, when fully implemented, will provide more than 1,100 simultaneous voices and data channels each with a 4800 bps data rate.

4.6 The Globalstar- 48 Constellation

Communication engineers at Loral Cellular Systems purposely avoided the use of cross linking between the Globalstar satellites and on-board switching techniques because they are convinced that these approaches are needless technological frills driven by engineering considerations rather than any real user needs.

as many functions as possible, including call processing and switching operations are located on the ground where they are accessible for maintenance and future upgrades.

Thus Globalstar is a potential competitor for Iridium, but it uses a much simpler constellation architecture with gateway stations tied into the existing ground-based infrastructure to be used for message switching.

The Global star constellation is being planned with 48 satellites (8 orbits, 6 satellites in each) boosted into eight orbit planes inclined 52 degrees with respect to the equator. The nominal altitude of the satellites is 1414 km. Each 450 kg Global star satellites will employ six spot-beams with CDMA modulations for highly effective use of the available frequency spectrum.

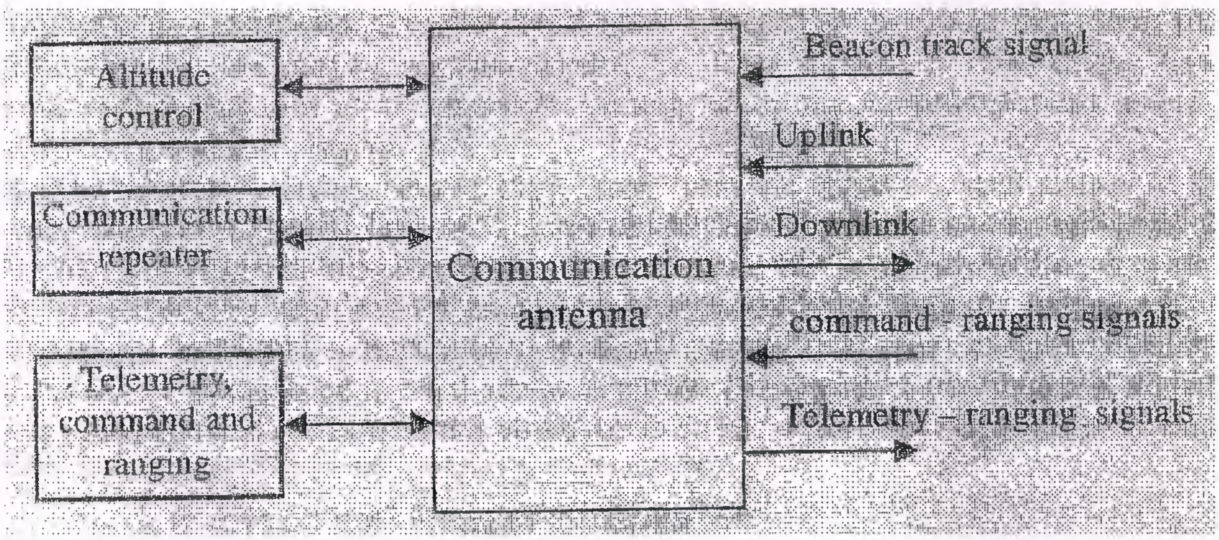


Figure 4.6 Communication Subsystems

The Globalstar constellation, when fully implemented, will provide 28,000 simultaneous voices and data channels each with a 4800 bps data rate. The antenna patterns will be formed on board each satellite by flat arrays constructed from both active and passive antenna elements.

Each satellite is powered by two deployable solar arrays, generating 1.1 kW. There they are processed and routed through the terrestrial infrastructure. But if the called party is another Globalstar subscriber the call will be uplinked from the same or another gateway to a satellite for transrnission to the destination

Globalstar has set up franchises with more than a hundred local service providers covering about 88 percent of the world's population. By the close of 1997 it had secured approvals for operations in 19 countries, among them being the United States, Russia, China, and Brazil.

To overcome limits on the frequencies available to users, Globalstar reuses the 16 MHz of bandwidth in each beam.

Aboard the satellite is a well-established repeater design that acts as a '"bent pipe" transponder relaying signals directly to the ground with minimal processing. This type of repeater is replaced by more complicated designs on satellites with a larger number of beams and where there is digital processing.

Characteristic of Globalstar

1. Eight orbital planes of six satellites each with al,414 km circular orbit inclined at 52°.
2. Orbit period 113 minute;
3. Each satellite mass, approximately 450 kg;
4. Electrical power; 2 sun collected solar arrays with sun tracking solar panel;
5. Antennas: satellite antenna provides 16 spot beam covering several thousand km;

User antenna omnidirectional; gateway antenna traking

6. Frequency bands

a) User links:

* L-band (1610-1626.5) MHz -user-to-satellite;
* S-band (2483.5-2500) MHz -satellite-to-user;
* C-band (6875- 7055) MHz -satellite-to-gateway;

b) Fider links:

* C-band (5091-5250) MHz -gateway-to-satellite.

4. 7 Communications Subsystems

The communications subsystems provides the receive and transmit coverage for the satellite consists of a communications antenna and a communications repeater. The communications

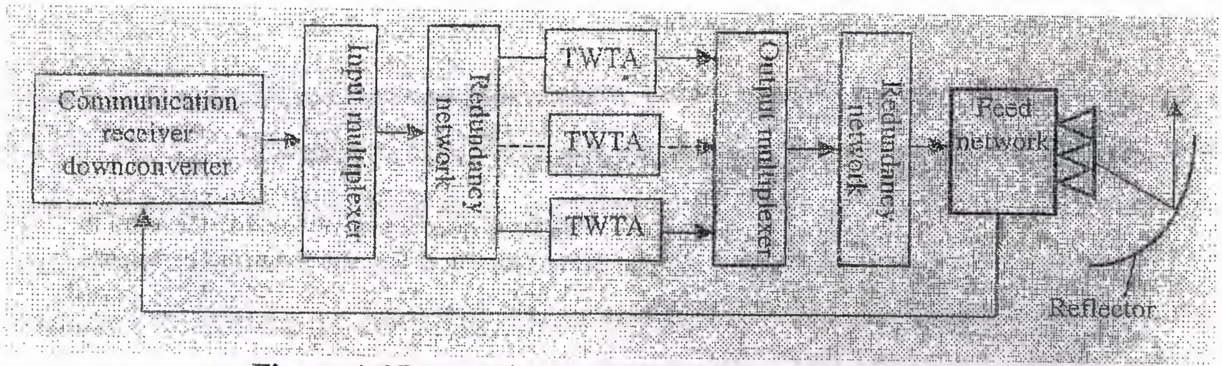


Figure 4.6 Low- noise Amplifier With a Typical Noise

antenna serves an interface between the earth stations on the ground and various satellite subsystems during operation. The main function of the antenna is to provide shaped downlink and uplink beams for transmission and reception of communications signals in the operating frequency bands In addition the antenna may be used to provide a signal link for the satellite telemetry, command, and ranging subsystem which in conjunction with the altitude control subsystem provides beacon tracking signals for precise pointing of the antenna toward the earth coverage areas. Communication Satellite subsystem is given in Figure 4.6.

The communications repeater generally consists of the following modules, as shown schematically in Figure 4.7.

1. A wideband communications receiver/downconverter
2. An input multiplexer
3. Channelized traveling wave tube amplifiers (TWTA)
4. An output multiplexer

The wideband communications receiver/downconverter is designed to operate within the typical 500-MHz bandwidth allocated for C-band (5.9 to 6.4 GHz) and Ku-band (14 to 14.5 GHz) uplink signals and is shown schematically in Figure 9.8 for a Ku-band uplink. The uplink signals are first filtered by a waveguide bandpass filter with about a 600-MHz bandwidth and then amplified by a parametric or a solid-state gallium arsenide field effect transistor (GaAs FET) low-noise amplifier with a typical noise figure of 2 to 4 dB.

The amplified signals are then downconverted to the 11.7 to 12.2-GHz downlink Ku band (3.7 to 4.2 GHz for a downlink C band) by a microwave integrated circuit downconverter. After downconversion, the signals are again amplified by a 11.7 to 12.2 GHz GaAs FET amplifier and passed through a ferrite isolator to the input multiplexer. The input multiplexer is employed to separate the 5üü-MHz bandwidth into individual transponder channels whose bandwidth depends on the satellite's mission. For example, a 500-MHz bandwidth can be

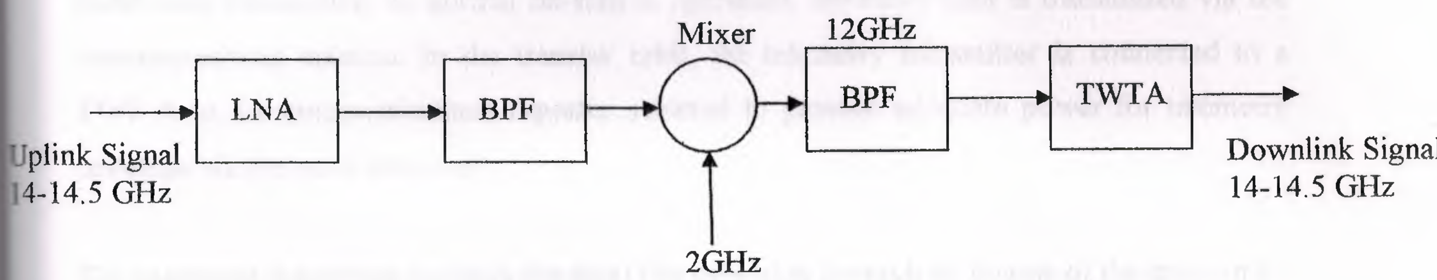


Figure 4.8 Center-To-Center Separator

divided into 8 transponder channels with a center- to-center separation of 61 MHz. With frequency reuse, there are altogether 16 transponder channels in the satellite.

The channelized TWTAs amplify the low-level downlink signals to a high level for transmission back to earth. Driver amplifiers are normally employed in front of the high-power TWT As to allow the communications receiver to be operated in the linear mode. The size of the TWTA depends on the mission and is about 15 to 30 W for a 6I-MHz Ku-band transponder. The TWTA establishes the transponder output power and normally operates near saturation to achieve the desired output power. Thus it is the dominant non linear device in a transponder and can affect the link signal performance considerably. The output downlink signals from the channelized TWTA are combined by the output multiplexer for retransmissions to earth.

A complete communications subsystems employing frequency reuse and consisting of 16 transponders (8 transponders use the horizontal polarization and 8 transponders use the vertical polarization). The number of odd and even transponders in the east and west beams can be selected by using the variable power dividers.

4.8 Telemetry, Command, and Ranging Subsystem

The telemetry subsystem monitors all satellite subsystems and continuously transmits to the earth sufficient information for determination of the satellite altitude, status, and performance as required for satellite and subsystem control. The telemetry transmitter also serves as the downlink transmitter for the ranging tones. The primary telemetry data mode is normal1y pulse code modulation. In normal on-station operation, telemetry data is transmitted via the communications antenna. In the transfer orbit, the telemetry transmitter is connected to a TWT A in the communications repeater selected to provide adequate power for telemetry coverage via the omni antenna.

The command subsystem controls the satel1ite operation through all phases of the mission by receiving and decoding commands from the ground station. It also generates a verification signal and upon receipt of an execute signal carries out the commands. The command subsystem also serves as an uplink receiver for the ranging signals. Again, the omni antenna is used in the transfer orbit for command and ranging and as an on-station backup, while the communications antenna is used on- station for command and ranging.

The ranging subsystem determines the slant range from the ground control station to the satellite for precise transfer and geostationary orbit determination. The slant range is determined by transmitting to the satellite multiple tones modulated onto the command carrier which is received by the command receiver, demodulated, and retransmitted by the telemetry transmitter to the ground control station where the phase difference is accurately measured. During on-station operation ranging is performed via the communications antenna; antenna coverage for ranging during the transfer orbit is provided by the omni antenna.

4.9 Electrical Power Subsystem

The satellite generates power by using a solar array of silicon cells. In a spin-stabilized satellite, the solar array consists of two concentric cylindrical panels of silicon cells. The forward panel is attached to the main structure and is divided into two arrays separated by a thermal radiator band. The aft panel is retracted over the forward panel during a transfer orbit and extended into its operating position in a geostationary orbit. In a transfer orbit, the aft panel provides solar power only. The disadvantage of a spin-stabilized satellite is that only one-third of the solar array is exposed to the sun at any time, resulting in power limitations. For a higher power level a larger satellite is required to provide space for body-mounted solar cells. The three-axis body-stabilized configuration can provide much more power by using deployed solar panels of wings. The array consists of many panels hinged together in two sets. In a transfer orbit, the panels are folded and stowed by restraint bands against the north- and south-facing sides of the satellite. The outermost panel is partially illuminated by the sun and furnishes a small amount of solar power. When the satellite reaches the geostationary orbit, the array is deployed and full power becomes available.

4.10 Earth Station

4.10.1 Antennas

The earth station antenna is one of the important subsystems of the RF terminal because it provides a means of transmitting the modulated RF carrier to the satellite within the uplink frequency spectrum and receiving the RF carrier from the satellite within the downlink frequency spectrum. The earth station antenna must meet three basic requirements:

1. The antenna must have a highly directive gain; that is, it must focus its radiated energy into a narrow beam that illuminate the satellite antenna in both the transmit and receive modes, and hence to provide the required uplink and downlink carrier power. Also, the antenna radiation pattern must have a low sidelobe level to reduce interference into other satellites and terrestrial systems.
2. The antenna must have a low noise temperature so that the effective noise temperature of the receive side of the earth station, which is proportional to the antenna temperature, can be kept low to reduce the noise power within the downlink carrier bandwidth.
3. The antenna must be easily steered so that a tracking system (if required) can be employed to point the antenna beam accurately toward the satellite taking into account the satellite's drift in position. This is essential for minimizing antenna-pointing loss.

Antenna Types

The two most popular earth station antennas that meet the above requirements are the paraboloid antenna with a focal point feed and the Cassegrain antenna.

A paraboloid antenna with a focal point feed is shown in Figure 4.9. This type of antenna consists of a reflector, which is a section of a surface formed by rotating a parabola about its axis, and a feed whose phase centre is located at the focal point of the paraboloid reflector. The size of the antenna is represented by the diameter D of the reflector. The feed is connected to a high power amplifier and a low noise amplifier through an orthogonal mode transducer (OMT) which is three port networks.

This type of antenna is easily steered and offers reasonable gain efficiency in the range of 50

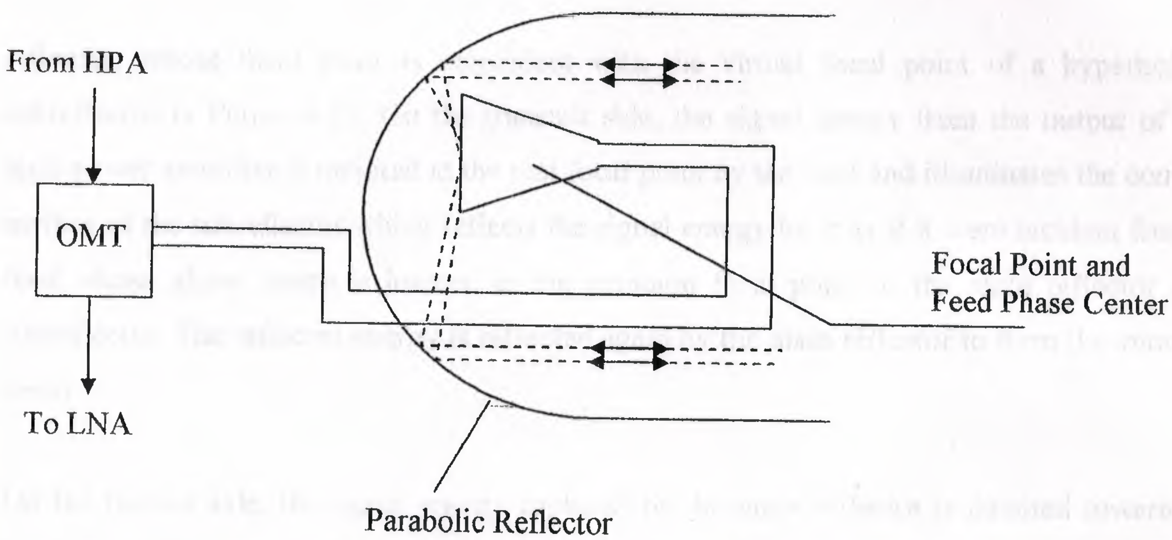


Figure 4.9 a Parabolic Antenna

to 60%. The disadvantage occurs when the antenna points to the satellite at a high elevation angel. In this case, the feed radiation which spills over the edge of the reflector illuminates the ground whose noise temperature can be as high as 2900 K and results in a high antenna noise contribution.

A Cassegrain antenna is a dual-reflector antenna, which consists of a paraboloid main

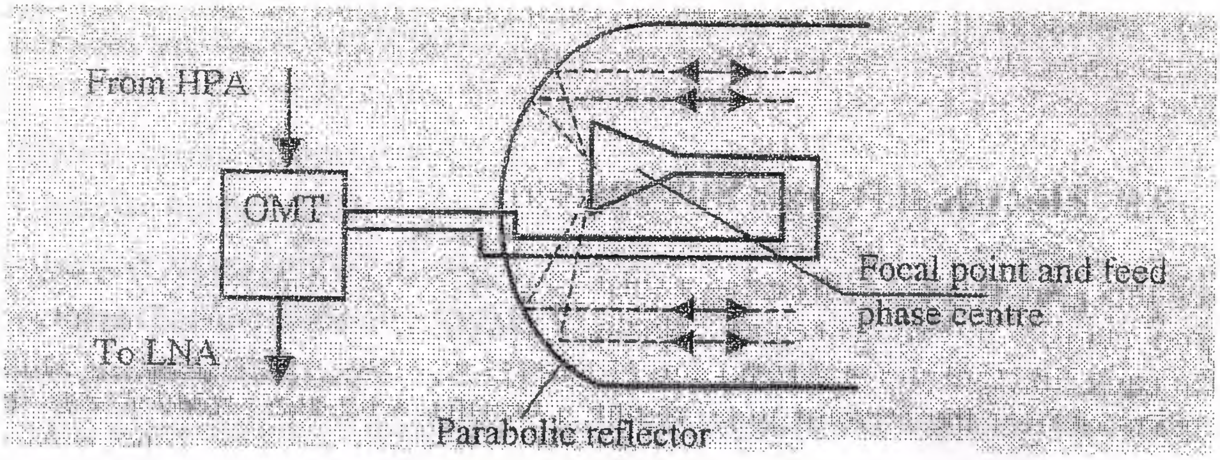


Figure 4.10 a Cassegrain Antenna

reflector, whose focal point is coincident with the virtual focal point of a hyperboloid subreflector in Figure 4.10, On the transmit side, the signal energy from the output of the high-power amplifier is radiated at the real focal point by the feed and illuminates the convex surface of the subreflector which reflects the signal energy back as if it were incident from a feed whose phase centre is located at the common focal point of the main reflector and subreflector. The reflected energy is reflected again by the main reflector to form the antenna beam.

On the receive side, the signal energy captured by the main reflector is directed toward its focal point. However the sub reflector reflects the signal energy back to its real focal point where the phase center of the feed is located. The feed therefore receives the incoming energy and routes it to the input of the low-noise amplifier through the OMT. A Cassegrain antenna is more expensive than a paraboloid antenna because of the addition of the subreflector and the integration of the three antenna elements -the main reflector, subreflector, and feed-to produce an optimum antenna system. However, the Cassegrain antenna offers many advantages over the paraboloid antenna: low noise temperature, pointing accuracy, and flexibility in feed design. Since the spillover energy from the feed is directed toward the sky whose noise temperature is typically less than 30K.

Parameters of antenna

a) Antenna Gain

Gain is perhaps the key performance parameter of an earth station antenna because it directly affects the uplink and downlink carrier power. The gain is given by

$$G = \eta*(3.14*f*D/c)^2 \quad (4.1)$$

where D -antenna diameter (m).

f -radiation frequency (Hz)

c -speed of light = 2.997925 10 m/s

η- antenna aperture efficiency (=0.95)

b) Antenna Pointing Loss

A loss in gain can occur if the antenna-pointing vector is not in line with the satellite position vector as shown in Figure 4.11. The antenna pointing loss can be evaluated from the antenna gain pattern, which is a function of the off -axis angle.

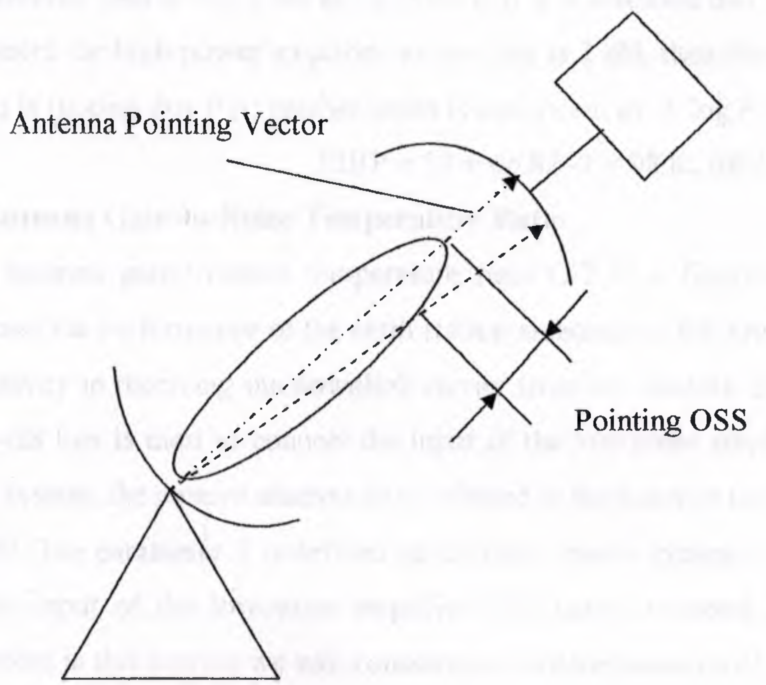


Figure 4.11 Antenna Pointing Loss

Because the earth station antenna is subjected to a wind loading effect and the satellite drifts in orbit, an antenna tracking system is necessary for a large diameter antenna to minimize the pointing error. The antenna tracking system is a closed-loop pointing system; that is, the antenna-pointing vector, which is function of the azimuth and elevation angles, is derived from the received signal. One of the commonly used antenna tracking systems for earth stations is a step track which derives the antenna pointing vector from the signal strength of a satellite beacon.

c) Effective Isotropic Radiated Power

To express the transmitted power of an earth station or a satellite, the effective isotropic radiated power (EIRP) is normally employed. The earth station EIRP is simply the power generated by the high -power amplifier times the gain of the earth station antenna, taking into account the loss in the transmission line (wave guide) that connects the output of the high power amplifier to the feed of the earth station antenna. If we let P(t) denote the input power at the feed of the antenna and G(t) the transmit antenna gain, the earth station EIRP is simply

$$EIRP = P(t) . G(t)$$

For example, consider a 2-kW high-power amplifier and a 20-m Cassegrain antenna whose transmitted gain is 66.82 dB at 14.25GHz. if it is assumed that the loss of the waveguide that connects the high power amplifier to the feed is 1 dB, then the earth station EIRP in decibel watts is (noting that P (t) decibel-watts is equivalent to 10 log P (t) watts).

$$EIRP = 33 + 66.82 - 1 = 98.82 \text{ dBW}$$

d) Antenna Gain-to-Noise Temperature Ratio

The antenna gain-to-noise temperature ratio *G/T* is a figure of merit commonly used to indicate the performance of the earth station antenna and the low-noise amplifier in relation to sensitivity in receiving the downlink carrier from the satellite. If a piece of waveguide with a 0.53-dB loss is used to connect the input of the low noise amplifier to the output port of the feed system, the receive antenna gain referred to the input of the low noise amplifier is simply 65 dB. The parameter T is defined as the earth station system noise temperature referred also to the input of the low-noise amplifier. We have discussed the antenna gain previously, therefore in this section we will concentrate on determination of the earth station system noise temperature.

4.10.2 The High Power Amplifier

One of the most widely used high power amplifiers in earth stations to the traveling wave tube amplifier (TWTA). The traveling wave tube employs the principle of velocity modulation in the form of traveling waves. The RF signal to be amplified travels down a periodic structure called a helix. Electrons emitted from the cathode of the tube are focused into a beam along the axis of the helix by cylindrical magnets and removed at the end by the collector after delivering their energy to the RF field. The helix slows down the propagation velocity of the

RF signal (the velocity of light) to that of the electron beam, which is controlled by the dc voltage at the cathode. Those results in an interaction between the electric field include by the RF signal and the electrons, which results in the transfer of energy from the electron beam to the RF signal causing it to be amplified.

Another type of high-power amplifier used in earth stations is the klystron amplifier, which can provide higher gain and better efficiency than the traveling wave tube amplifier but at a much smaller bandwidth. For low-power amplification are used GaAs FET amplifiers. These are solid state amplifiers and offer much better efficiency than the above two types of amplifiers.

4. 10.3 Up converter

The Upconverter accept the modulated IF carrier and translate its frequency w_0 to the uplink frequency $(w)u$ by mixing w_0 with a local oscillator frequency w_1

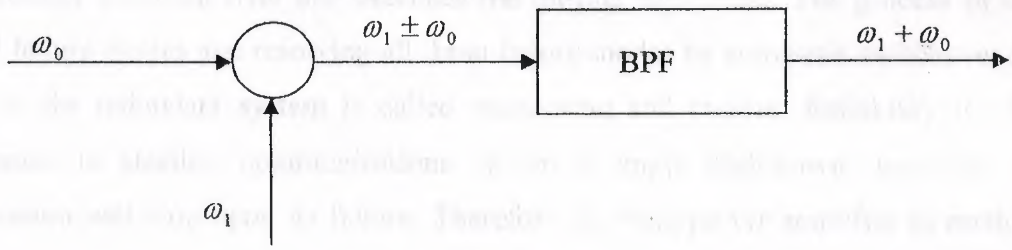


Figure 4.12 Upconvertor

The upconversion may be accomplished with the single or double-conversion preprocess.

4.10.4 Down converter

The downconverter (DC) receives the modulated RF carrier from the low-noise amplifier and translates its radio frequency rod in the downlink frequency spectrum of the satellite to the intermediate frequency w_d. Like upconversion downconversion may be achieved with a single conversion process or with a dual conversion process using mixer.

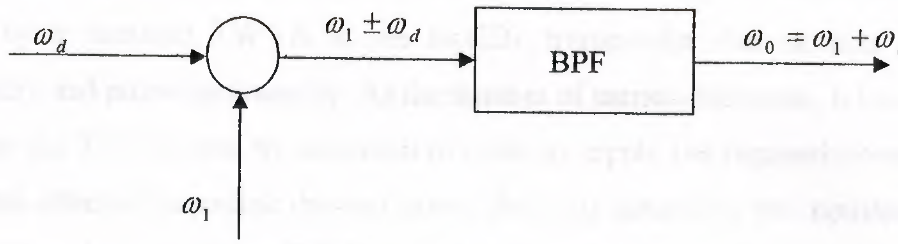


Figure 4.13 Down converter

4.10.5 Redundancy Configurations

As we have seen in previous sections, except for the antenna all earth stations systems namely, the high-power amplifier, the upconverter, and the downconverter, must employ some sort of redundancy to maintain high reliability which is of utmost importance. When the on-line equipment in the redundancy configuration fails. The standby equipment is automatically switched over and becomes the on-line equipment. The process of detecting critical failure modes and resolving all these failure modes by automatic switchover from the failed to the redundant system is called monitoring and control. Reliability is of utmost importance in satellite communications. When a single high-power amplifier is used, transmission will stop upon its failure. Therefore the high-power amplifier in earth stations always employs some sort of redundancy configuration. The most basic redundancy configuration is the 1.1 redundancy.

4.11 Multiple Access

4.11.1 Frequency Division Multiple Access

The simplest and most widely used multiple access technique of satellite communications is frequency division multiple access, where each earth station in a satellite network transmits one or more carriers at, different centre frequencies to the satellite transponder. Each carrier is assigned a frequency band (Bc) with a small guard band (Bg) to avoid overlapping between adjacent carriers, The satellite transponder receives all the carriers in its bandwidth, amplifies them, and retransmits them back to earth, A frequency division multiple access system is shown schematically in Figure 4.14. In this type of system each carrier can employ either analog modulation, such as frequency modulation, or digital modulation, such as phase-shift keying. A major problem in the operation of FDMA satellite systems is the presence of intermodulation products in the carrier bandwidth generated by the amplification of multiple carriers by a common TW'TA in the satellite transponder that exhibits both amplitude nonlinearity and phase nonlinearity. As the number of carriers increases, it becomes necessary to operate the TWTA close to saturation in order to supply the required power per carrier to reduce the effect of downlink thermal noise. But near saturation the input/output amplitude transfer characteristic of the TWT A is highly nonlinear, and consequently the level of intermodulation products is increased and affects the overall performance.

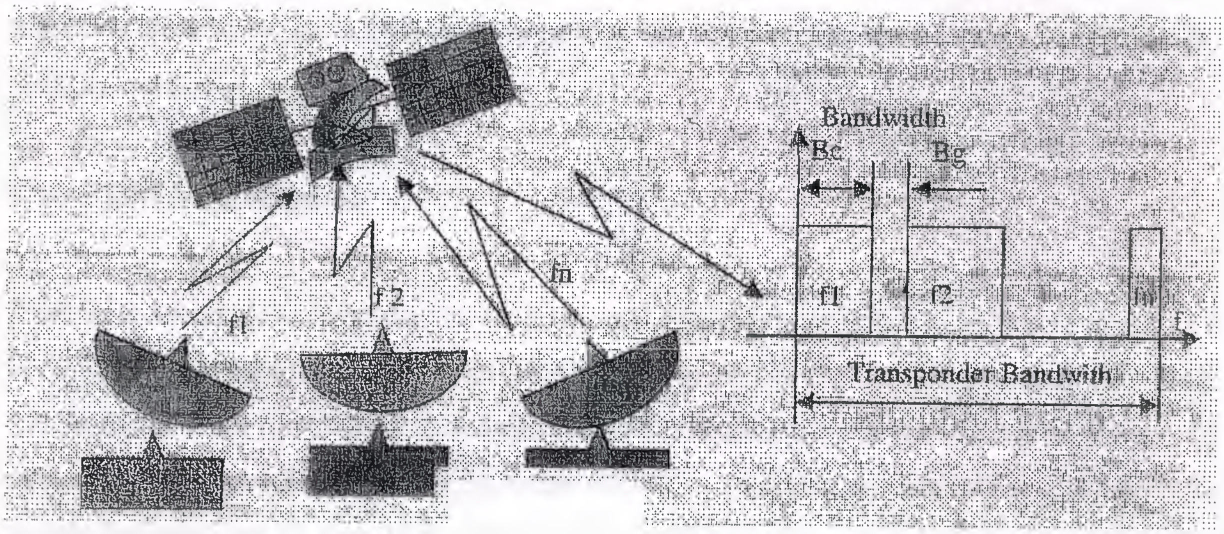


Figure 4.14 Frequency Division Multiple Access System

FDM-FM-FDMA Since the inception of satellites analog modulation, such as frequency modulation, has been used for carrier modulation in satellite communications using FDMA; it will probably be employed in existing equipment for years to come despite advances in the development of digital satellite systems. There are two main FDMA techniques in operation today.

Multichannel-per-carrier transmission, where the transmitting earth station frequency division- multiplexes several single sideband suppressed carrier telephone channels into one carrier baseband assembly which frequency-modulates a RF carrier and is transmitted to a FDMA satellite transponder. This type of operation is referred to as FDM-FM-FDMA. Single-channel-per-carrier transmission, where each telephone channel independently modulates a separate RF carrier and is transmitted to a FDMA satellite transponder. The modulation can be analog, such as FM, or digital, such as PSK.

Single Channel Per Carrier

Unlike FDM-FM-FDMA systems which serve large-capacity links, single-channel-per-carrier systems are more suitable for applications that require only a few channels per link. In these systems each telephone channel independently modulates a separate RF carrier and is transmitted to the satellite transponder on a FDMA basis. A 36-MHz transponder can carry as many as 800 voice channels or more. If the carrier modulation is digital, the performance is measured in terms of the average probability of bit error. For analog carrier modulation, FM is employed. FM-SCPC systems are the most commonly used systems because of their attractiveness in terms of cost and simplicity. The design of a FM-SCPC link can be expressed in terms of the signal-to-noise ratio at the FM demodulator output, as in FDM-FM-FDMA.

FM-FDMA Television

Television broadcasting via satellite in the United States is among the most highly developed in the world. TV programming is distributed on the fixed satellite service portion of the C and Ku bands. In 1983 the Federal Communications Commission approved a frequency band for domestic direct broadcast satellite services (DBS) to provide direct-to-home television: an uplink frequency of 17.3 to 17.8 GHz and a downlink frequency of 12.2 to 12.7 GHz. The DBS downlink portion of the Ku band is adjacent to the 11.7-to 12.2-GHz downlink frequency of the FSS portion of the Ku band. High- power direct broadcast satellites have many characteristics similar to those of communications satellites, except that the DBS downlink radiated power is about 10 dB more per transponder. The powerful television signal lets individual users receive programs with antennas as small as 0.7 m in diameter, which can be mounted on the roof of an average house. The nominal carrier-to-noise ratio is about 14 to 15 dB when used with an earth station G T of 10 dB/K.

Companded FDM-FM-FDMA And SS8-AM-FDMA

The transponder capacity in FDM-FM-FDMA operations can be improved by the use of syllabic compounders. The traditional use of syllabic compounders has been to improve the quality of signal transmission over poor channels. A compounder consists of a compressor at the transmit side of the satellite channel and an expander at the receive side. The compressor is a variable-gain amplifier that gives more gain to weak signals than to strong signals. This results in an improved overall signal to-noise ratio because the Low-Level speech signals are increased in power above the channel noise. On the receive side, the expander restores the signals level by attenuating the Low-Level speech signals. During pauses in the speech signal, channel noise is reduced by the expander, and hence giving further improvement in the overall subjective signal-to-noise ratio. A 36-MHz transponder can accommodate a single FDM-FM-FDMA carrier of 1100 uncompounded channels. On compounding the channels, the capacity is increased to about 2100 channels. With over deviation beyond its allocated bandwidth (with no loss in the quality of the channels), such a transponder can carry about 2900 channels.

Recent use of solid-state power amplifiers with sufficiently linear characteristics to replace nonlinear TWTAs allows the use of compounded single-sideband-amplitude modulation-frequency division multiple accesses (SSB-AM-FDMA) to achieve 6000 channels per transponder of 36-MHz bandwidth for a single carrier. Besides the high capacity, SSB-AM-FDMA offers another major t advantage over FDM-FM-FDMA from a multiple access point of view. The capacity of a satellite transponder using SSB-AM-FDMA is not decreased by multiple accesses, unlike FDM-FM-FDMA. Also, the capacity of small FDM-FM-FDMA carriers cannot be increased by over deviation, because of the cross talks among the carriers. A transponder carrying 6000 SSB-AM-FDMA channels can be accessed, say by four earth stations with 1500 channels, each with no loss in capacity. On the other hand, a four-carrier compounded FDM-FM-FDMA transponder can carry about 1500 channels.

4.11.2 Time Division Multiple Access

Time division multiple accesses are a multiple access protocol in which many earth stations in a satellite communications for transmission via each satellite transponder on a time division basis. All earth stations operating on the same transponder are allowed to transmit traffic bursts in a periodic time frame-the TDMA frame. Over the length of the burst, each earth station has the entire transponder bandwidth available to it for transmission. Transmit the timing of the bursts is carefully synchronized so that all the bursts arriving at the satellite transponder from a community of earth stations in the network are .closely spaced in time but do not overlap. The satellite transponder receives one burst at a time, amplifies it, and retransmits it back to earth. Thus every earth station in the satellite beam served by the transponder can receive the entire burst stream and extract the bursts intended for it. A simplified diagram of a TDMA operation is shown in Figure 4.15.

Traffic Burst The traffic bursts (T1, T2, ...) transmitted by the traffic stations carry digital information. Each station accessing a transponder may transmit one or more traffic bursts per TDMA frame and may position them anywhere in the frame according to a burst time plan that coordinates traffic between stations. The length of the traffic burst depends on the amount

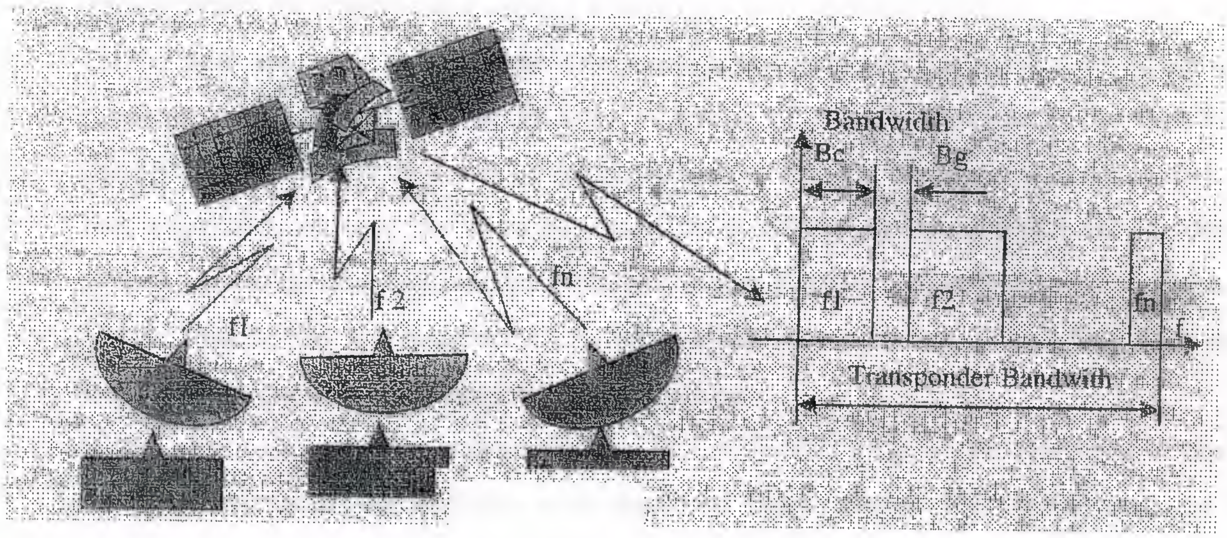


Figure 4.15 TDMA

of information it carries and can be changed if required. The location of the traffic bursts in a frame is referenced to the time of occurrence of the primary reference burst (R1). By detecting the primary reference burst, a traffic station can locate and extract the traffic bursts or portions of traffic bursts intended to it. Also, it can derive the transmit timing of its bursts precisely, so that they arrive at the satellite transponder within their allocated positions in the TDMA frame and avoid overlapping with bursts from other stations.

Guard Time A short guard time is required between bursts originating from several stations that access a common transponder to ensure that the bursts never overlap when they arrive at the transponder. The guard time must be long enough to allow differences in transmit timing accuracy and in the range rate variation of the satellite. The guard time is normally equal to the time interval used to detect receive the timing pulse marking the start of receive a TDMA frame at a station. There is no transmission of information during the guard time.

CONCLUSION

To design a satellite system for trucking applications was in the past no more difficult than to design the transmission system whose role it was meant to fulfill.
This is now changing rapidly. The evolution of the network, the emergence of very competitive alternatives, and the foreseen evolution of the satellite systems themselves call for a reappraisal of their role in the implementation of the backbone digital network. This reappraisal requires the investigation of many system problems which did not use to show up on the tables of satellite communications engineers. The problems are there. But the benefits that this integration may provide are very attractive. This is why ESA is actively studying every implication of these concepts.
In this paper, the need for internetworking of LANS and some of the problems of the existing interconnection facilities were discussed briefly. A satellite based wide area network concept and inter-LAN traffic assessment issues were also discussed.
Given the major evaluation criteria as suggested earlier in the paper, the conclusion is that such a procedure provides a flexible ant efficient space telecommunication network capable of satisfying user requirements. In particular it would provide: minimum transmission delay, user access direct or via ISDN, flexibility in bandwidth-to-service allocation, further extensions.

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