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Graduation Project

MOBILE RADIO SYSTEM

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WIRELESS TELEPHONES

INTRODUCTION

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1.1- ABOUT THIS CHAPTER

This chapter focuses attention on telephones that perform most or all of the functions of the conventional telephone but, are connected with a radio link rather then wired directly.Although the term " wireless" may seem a bit old - fashioned ,it is about the only general name for such telephones since term like "cordless telephones","mobile telephones",and even "radio telephone" have come to mean specific types of telephones without wires .

the mobile telephone

1.2- CORDLESS TELEPHONES

The cordless telephone consiste of base and portable unites

that

The base and portable units of cordless telephones are linked by a low power FM transmitter receiver system. (fig 10.1)

The first type of wireless telephone to be discussed is the "cordless" which is used as an extantion telephone in homes and businesses. (Figure 10 -1) shows that the cordless telephone consists of two parts : a base unit and a portable unit . The connecting wires of a conventional telephone between the portable unit and the base unit are replaced with low -power radio transmissions.

using FM modulation principle

The radio link is completed by the transmission of a carrier that is frequency modulated (FM) with the information to be transmited. The carrier and the modulation principles are the same as described for the modem in chapter 9. The only difference is that the frequency of the carrier is much higher for the cordless telephone .

The cordless telephone is one of two types . For one type ,the portable unit has no key pad ,and only the voice comminications are radio linked . The key pad on the base unit .The other type has the key pad on the portable unit and all the normal functions of a telephone are radio linked to the base unit .

The telephones are electronic telephones as have been described in this book with the radio frequency transmission and modulating and demodulating electronic circuits added for the radio link. The have pulse generators and / or DTMF generators for dialing , electonic single - or dual - frequency ringers , and electronic speech circuits . Some have special futures that allow the telephone to be used as an intercom . Some have speech and / or ringer volume controls and some have special security futures to prevent unauthorized use. Cordless telephones are available with the same futures as the electronic telephones that use cords. However , the cordless type must be plugged in to a power source rather than operating from the line . Most have redial future and some have a memory to store several numbers. The cord type electronic telephone usually obtains the operating power directly from the telephone line ~~as has been described earlier ;~~ however , the cordless telephone , because of its greater power requirements , usually obtains operating power from the household AC outlet . This is a minor disadvantage since the cordless telephone won't operate if a utility power failure occurs.

Cordless telephone have the following features:
- speach and volume control
- security features to prevent unautorized use
- LSD display shown
  • Caller phone numbers
  • Date and time
  • Review incomming and outcoming calls
- down, up and select functions
- Keeping 50 incoming, 50 outcoming numbers
- DTMF autodetecting system

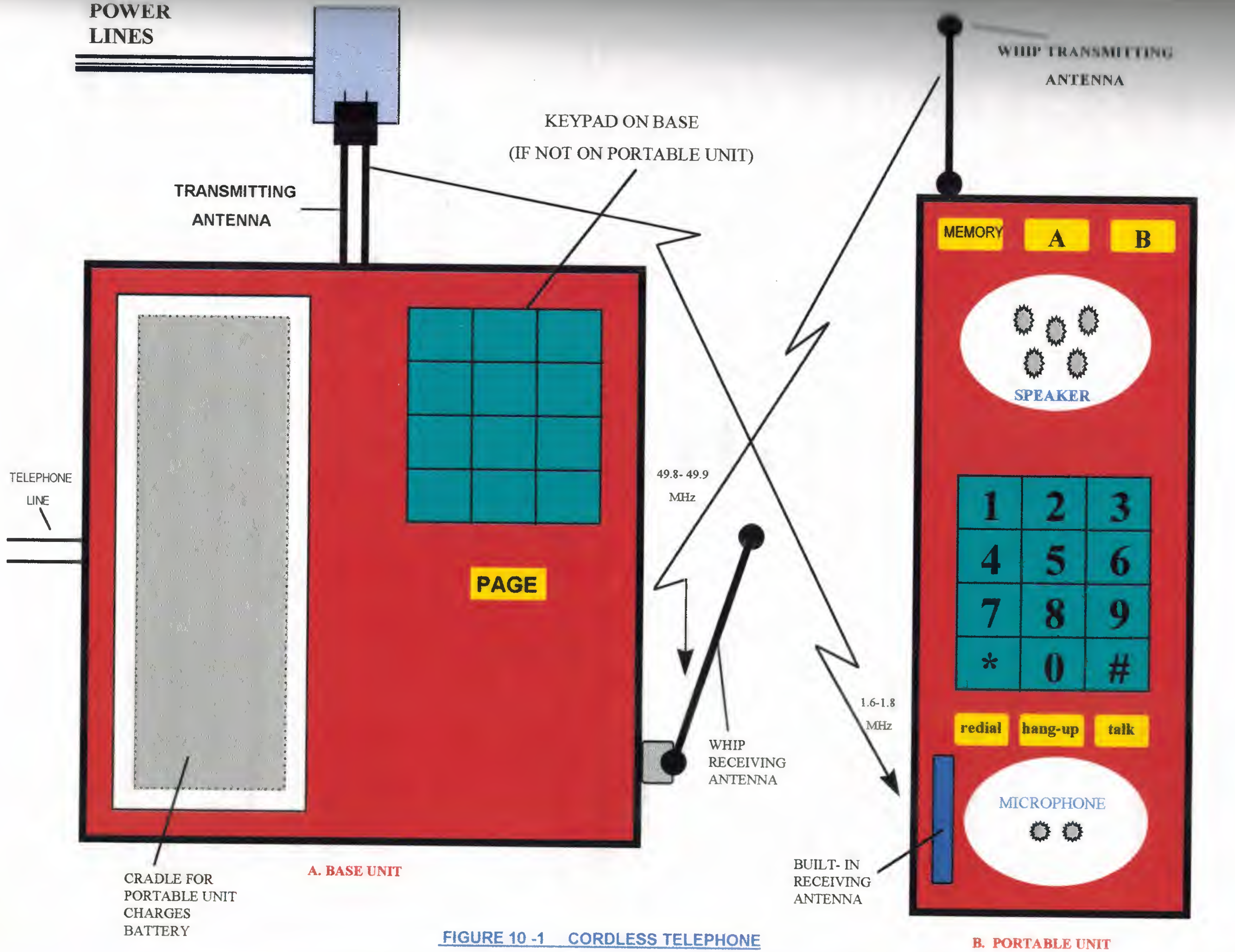


FIGURE 10 -1 CORDLESS TELEPHONE

1.3- BASE UNIT

As shown in figure 10- 1 . The base unit connects directly to the telephone line to complete the local loop to the central office A 2-wire to 4-wire hybrid arrangement couples the local loop to the separate transmitte and receive sections in the base unit . The base unit transmits on a frequency of 1.6 to 1.8 MHz . It uses the AC power line that supplies the power for the base station electronics as a transmiting antenna. The base unit transmites on a carrier frequency in the range from 1.6 to 1.8 MHz and the household electirical wiring is used as the transmitting antenna for the base unit. The nominal 1.7 MHz frequency modulated signal is fed from the base unit transmiter to the AC line through capacitors which block the line current from the base unit transmiter while passing the 1.7 MHz output to the line . This use of the house wiring as an antenna is not unique to cordless telephones: It also is used for wireless intercoms. This method provides good reception within and near the house as well as outside near power lines that are on the same side of the utility company's distribution transformer as the house circuit. This may include a neighbor's house wiring ; thus ,the potential exists for interference if that person also has a cordless telephone ; more about this later .

1.4- PORTABLE UNIT

An internal loopstick antenna (like that used in standard radio receivers) in the portable unit receives the nominal 1.7 MHz tranmission from the base unit over arrange from 50 to 1000 feet . The range depends not only on the manufacturer's design , but also on such things as whether the house wiring is enclosed in metal conduit and whether foil - backed insulation is used in the walls . The ringing or voice signal is recovered by demodulation and drives the speaker in the portable unit . The portable unit is powered by a battery which is recharged when placed in the receptacle in the base unit.

The portable unit receives the locally radiated r.f. signal from the base unit's antenna (the house wiring) with its built - in loopstick antenna , much like a portable radio the portable unit is usually in a standby mode which corresponds to the on-hook condition of a telephone set . When the ringer sounds , the user operates a talk swicth which turns on the transmitter in the portable unit. This transmitter transmits on a frequency in the range of 49.8 to 49.9 MHz and outputs the signal on the whip antenna. (since the whip is used only for transmit, it can be collapsed out of the way when the portable unit is on standby. If the portable unit is only for voice tranmission , it may have an internal antenna and its range is shorter .) A similar whip antenna on the base unit receives the FM signal from the portable unit ,demodulates it and appilies the off - hook signal to the telephone local loop. Dialing transmits modulation tones to the base unit, which sends either tones or pulses over the telephone lines after the two parties have been succesfully connected , the transmitter and receiver operate simultaneously .

When the user dials the number for outgoing calls ,the dial pulses produce tones which modulate the carrier for transmission to the base unit . The base unit recovers the tones by modulation. If DTMF service is used, the tones are sent on the telephone line. If pulse service is used the tones are converted to pulses and the telephone line is pulsed . When the connections between calling and called parties are established , both transmitters and receivers operate at the same time the permit two - way conversation.

1.5- FREQUENCY

Although a few cordless telephones use the same frequency for transmission in both directions, most use two different frequencies in the ranges given above.Unless multiplexing is used , the use of a single frequency provides only half-duplex (one - way - at - a - time) transmission while the use of two frequencies allows full - duplex (simultaneous two - way) transmission just like a wired telephone. A choice of several frequencies is available so that neighbours can use different frequencies to prevent interference and eavesdropping. Also, signaling is done by guard tones in sequences which are selectable and unlikely to be duplicated in the neighborhood.

2.1- MOBILE TELEPHONES

Mobile telephones maybe thought of as cordless telephones with
сложный, тщательный elevated - возвышенный
elaborate portable and base units. High - power transmitters and elevated 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 more than 60 subscribers Per base station , are the major differences between cordless telephones and mobile telephones .

2.2- BASE UNIT

Figure 10-2 shows a mobile telephone system . 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. The mobile telephone base unit can operate on many channels simultaneously and can easily cover the avarege city with a power of several 100 watts. It covers a circular area of up to 30 miles in radius for clear reliable communications, but transmitters with the same frequency are not spaced closer than about 60 to 100 miles because of the noise interference levels.

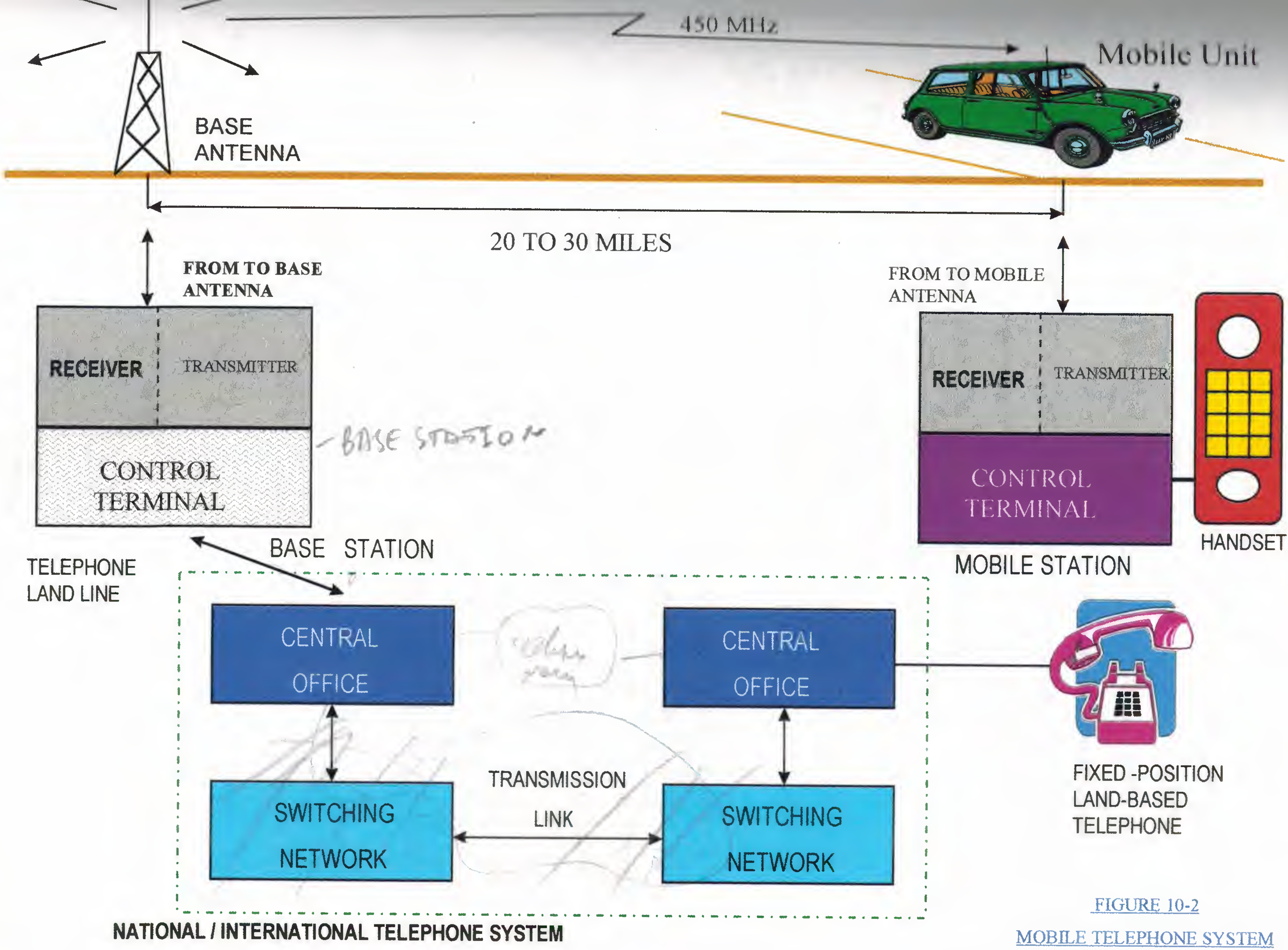


FIGURE 10-2
MOBILE TELEPHONE SYSTEM

The base unit receiver contains the necessary electronics to present its control terminal with a good audio signal. The control terminal interfaces the voice and control signals to the standart telephone circuits. The receiver base station contains filters , high - gain amplifiers, and demodulators ~~to provide a usable~~ voice signal to the telephone line .The control terminal contains the necessary detector and timing and logic circuits to control the transmissions link between the base unit and mobile units. ~~As a result telephone calls are coupled to and from the standart telephone system just like calls that are carried completely over wired facilities.~~ The control terminal has the neccessary interface circuits so that a call initiated at a mobile unit is interconnected through the national or international telefon system to the called party just as any other telephone call .

The national and international telephone system facilities are owned by the respective telephone companies . The base units and the mobile units may be

owned by the telephone 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 telephone system just like any other customer. This cost is then included in the charge by the RCC to the eventual user of the mobile units.

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

2.3- MOBILE UNIT

control terminal ~~and~~

the receiver

The mobile unit contains ~~a~~ receiver , ~~a~~ transmitter, ~~control logic ,~~ ~~control unit,~~ and antennas. ~~For the user it operates pretty much like and ordinary electronic telephones.~~

The mobile unit in ~~the user's vehicle consist 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, key pad and swicthes; antennas and the interconnecting cables. The control units performs all of the functions associated with normal telephone use .A modern control head with automatic functions is illustrated in Figure 10-3

The mobile telephone user with automatic control places and receives calls in the same manner as with an ordinary telephone .When the hand set is lifted to place a call , the radio unit automaticaly select 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 telephone system , and can then dial the number and proceed as if the telephone were direct wired. An incoming call to be mobile unit is signaled by a ringing tone and is answered simply by lifting the hand set and talking. Thus the automatic mobile telephone is as easily used as a home telephone . The mobile telephone combines the mobililty of the radio link and the world - wide switched network of the existing telephone system to provide a communication link to any other telephone in the world.

2.4- HOME AREA AND ROAMING

As previously stated, the mobile system is designed for optimum use within a 20 - 30 mil radius of the base antenna . This is called the subscriber's home area and a subscriber usually would remain in the home area. However if the subscriber moves out of the home area in to another area , the subscriber is referred to as a roamer and a different mode of operation applies .

Each mobile telephone has a unique telephone number which includes the home area's base station identification. When someone calls the mobile unit the calling party is connected first to the transmitter serving the subscriber's home area . As long as the subscriber is within radio range of that system,all is well ; otherwise , the base station won't get an answer from the mobile unit and the caller will get a no- answer signal. If the subscriber roams outside the home area , he / she can still be reached if a similar mobile telephone system exist in that area, provided proper advance arrangements have been made .

If a subscriber goes outside the range of his base station, his mobile telephone can only reached through another similar adjacent mobile base station system, provided advanced arrangements have been made.

Calls to roamers are usually placed by calling special number for the mobile servis operator who knows the roamer's location. The operator manually patches the call through base station serving the area of the roamer's location. Some systems can not handle roamers due to overload of their channels, and some system don't allow roamers .

3.1- DETAILED OPERATION

Different signaling techniques must be use in a mobile telefon system then in a wired facility. Since there are no wires connecting the telephone

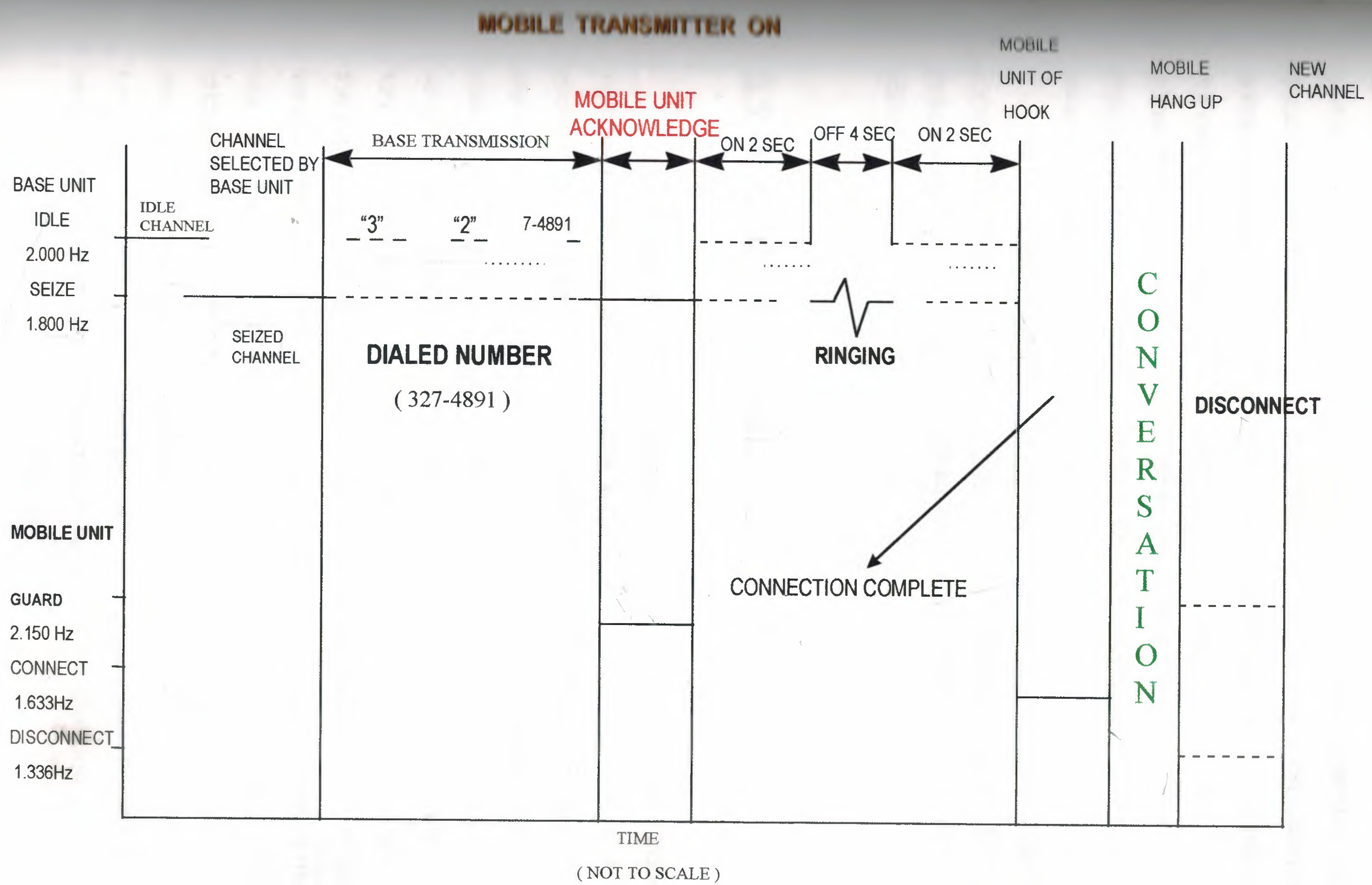


FIGURE 10 -4

BASE TO MOBILE SEQUENCE

to the network , both speech and signaling must be transmitted via radio . For wireless operation, tones are used for those signaling functions otherwise performed by voltage and current in hard- wired systems. This is accomplished through the use of special tones rather than applying a voltage level or detecting a current. The tones are selected so as not to be mistaken for other signaling tones, such as DTMF . The proper tone transmitted to the mobile unit will, for example , ring the mobile telephone to indicate an incoming call just as with a stantard telephone.A different tone is used to indicate off - hook , busy, etc.

The Improved Mobile Telephone System (IMTS) uses in -band signaling tones from 1300Hz to 2200Hz. The older mobile telephone system (MTS) had in-band signaling tones in the 600 Hz to 1500 Hz range . Some systems use 2805 Hz is manuel operation.

3.2- INCOMING CALL

To gain a better understanding of the system operation , let's trace an incoming call from a wire facilitiy subscriber through the base unit to a mobile unit .The base station controls all activity on all channels and can transmit on any idle channel . Regardless of how many channels are idle , it selects only one and places a 2000 Hz idle tone on it as shown figure 10 -4. All on- hook mobile units that are turned on automatically search for the idle tone and lock on the idle channel because this is the channel over which the next call in either direction will be completed .The base station selects on idle channel and modulates it with a 2 KHz tone. This becomes a market idle channel. This channel has now been reserved for the next land-originating telephone call.After locking on the idle channel , all on-hook mobile units "listen" for their number on that channel . When an idle channel becomes busy for a call in either direction , the base station control terminal selects another unused channel and marks it with the idle tone. All on -hook mobile units then move to the new idle channel .This process is repeated each time a new call is initiated as long as unused channels are available .

After the person calling the mobile subscriber dials the mobile units telephone number , the call is processed through the switched telephone network as in a normal landline call .Following the sequence in figure 10-4. When the call reaches the control terminal , the terminal seizes the idle channel , and indicates seizure by removing the idle tone from that channel and applying the 1800 Hz seize - ton .The seize - tone prevents mobile units from seizing the channel to originate a call . The control terminal than out - pulses the mobile units number over the base station transmitter at ten pulses per second , with idle -tone representing a mark (which corresponds to the make intervall in the pulsing) and seize - ton representing a space (correspanding to the break interval). When the land originating call reaches the control terminal the idle tone is replaced by a 1.8 KHz seize ton which seizes the reserved channel. The control terminal then sends out via the transmitter the mobile unit's .

Each on-hook mobile unit receiving the number transmission compares the received number to its unit number. As soon as a digit mismatch is detected, the mobile unit abandons that channel and searches for the new idle channel. Thus upon completion of the number transmission, all mobile units except the one called will have abandoned the seized channel and will be monitoring the new idle channel . Only the one mobile unit with a number match remains locked on that channel. The others automatically abandon it.

When the mobile unit receives its correct seven - digit address , the mobile supervisory unit turns on the mobile transmitter and sends the acknowledgement signal, 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 , the seize tone is removed and the call abandoned . However, upon receipt of the mobile acknowlegement

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 is discontinued and the call abandoned. When it receives Its unique number ,the mobile transmitter automatically broadcast a 2150 KHz acknowledgement. Ringing signals are then broadcast to the mobile unit.

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 . 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. When the subcriber hangs -up at the end of call , the mobile supervisory unit sends a disconnect signal - 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 .If the mobile unit goes off - hook to answer , another tone frequency burst is sent, allowing voice comminications to commence .Upon hang - up (on-hook), the mobile unit sends a disconnect signal by alternating disconnect tone and tone guard.

3.3- OUTGOING CALL

The sequence for a call originated by a mobile subscriber is illustrated by figure 10 -5. 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.The identification section of figure 10 - 5 is where the 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 pulses of the guard tone mixed in with the number pulses are for parity checking. The remainig functions of figure 10 -5 are similar to those of figure 10 - 4. 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 , dialing the number and the waiting for the called party to answer .

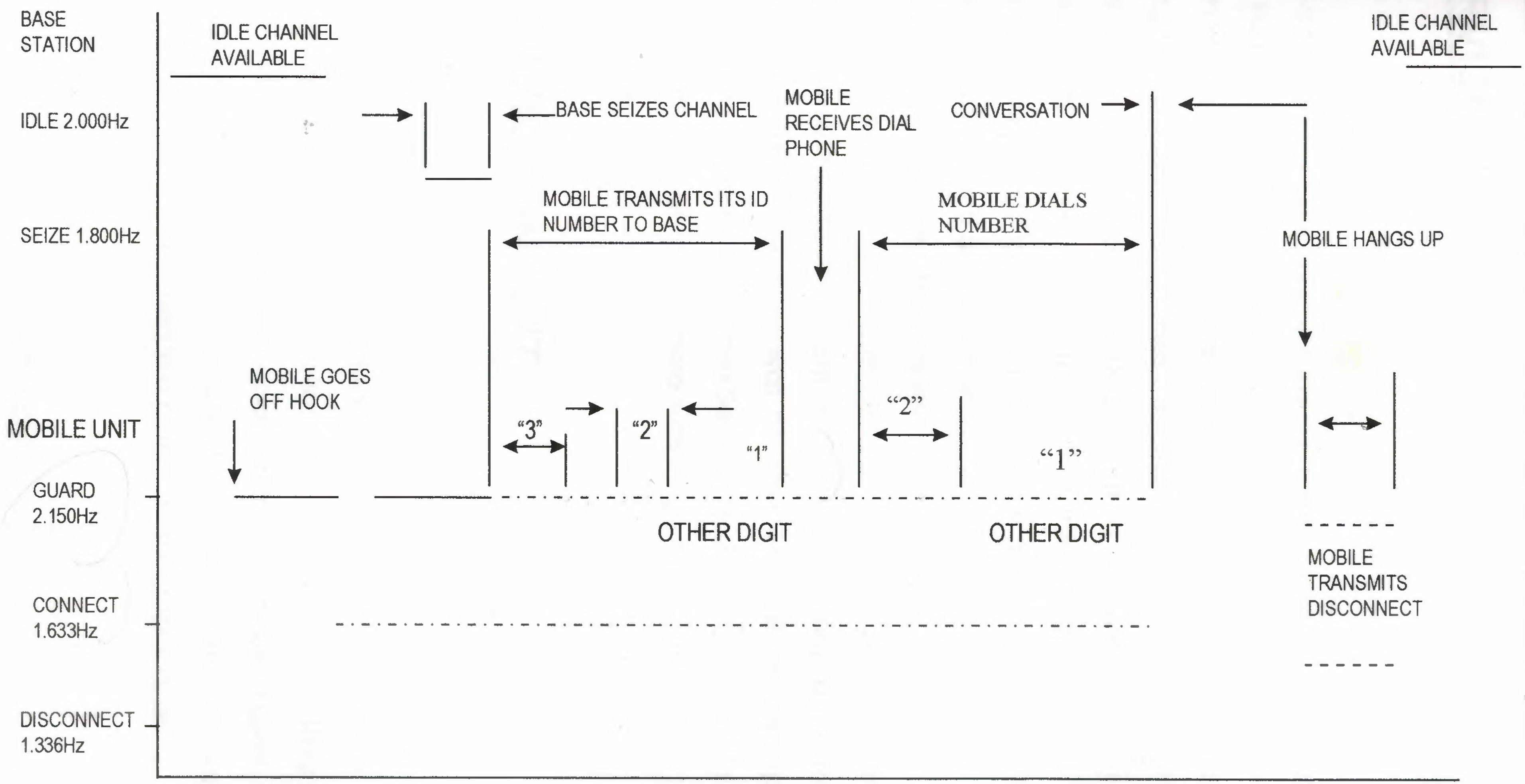


FIGURE 10-5

MOBILE TO BASE SEQUENCE

4.1- CELLULAR MOBILE TELEPHONE SERVICE

Mobile telephone service always has been a scarce luxury. Subscribers pay from 10 to 20 times more for mobile service than for residential telephone service , yet most urban telephone carriers and RCC s have long waiting lists for mobile telephones . In Chicago, for example , only 2000 mobile users can be accommodated, yet at list ten times that many desire service at present rates. The reason is that there simply are not enough channels to handle the demand , and the few dozen available are spread over several bands and divided among different type of carriers. The solution is not simply to assign new frequencies and build more transmitters because the spectrum space for new frequencies is simply not available ; besides , this would not eliminate the restirictions on roamers . Clearly ,an entirely new approach to mobile telephony was needed. The cellular concept , also called the Advanced Mobile Phone Service (AMPS) is a method to provide high quality mobile service for more subscribers at an affordable cost and the more freedom for roamers .

4.2- BASIC CONCEPT

The basic concept of the AMPS cellular system is to reduce the area covered by the transmitter by reducing the power of transmission . In this way, concentrated areas of population can have more transmitting stations, and thus more channels , because each transmitter handles a given number of conversations . In addition ,because transmitters cover less area , the same frequency can be re - used in a common geographical area.

4.3- SYSTEM STRUCTURE

The basic system arrangement is shown in figure 10 -6 . The service area is divided in two regions called cells, each of which has equipment to switch transmit, and receive calls to / from any mobile unit located in the cell . Each cell transmitter and the receiver operates on a given channel. Each channel is used for many simultanious conversations in cells which are not adjacent to one another , but are

far enough a part to avoid excessive interference . Thus a system with a relatively small number of subscribers can use large cells , and as demand grows, the cells are divided in to smaller ones. By dividing a city in to many cells , each serviced by a low power transceiver base station , the number of available channels over the city is increased enormously .

4.4- THE MTSO

The cell sites are interconnected and controlled by a central Mobile Telecommunications Switching Office (MTSO) , which is basically a telephone switching office as far as hardware is concerned , but as shown in figure 10-6

it uses a substantial amount of additional digital equipment programmed for cellular control . It not only connects the system to the telephone network, but also records call information for billing purposes . The MTSO is linked to the cell sites by a group of voice trunks for conversations , together with one or more data links for signaling and control. The MTSO controls not only the cell sites via radio commands, but also many functions of the mobile units . Advanced Mobile Phone Service (AMPS) is a registered service mark of AT&T Co. A central mobile telecomunication switching office (MTSO)performs all of the functions of a normal switching office and also controls each of the cell transceiver functions.

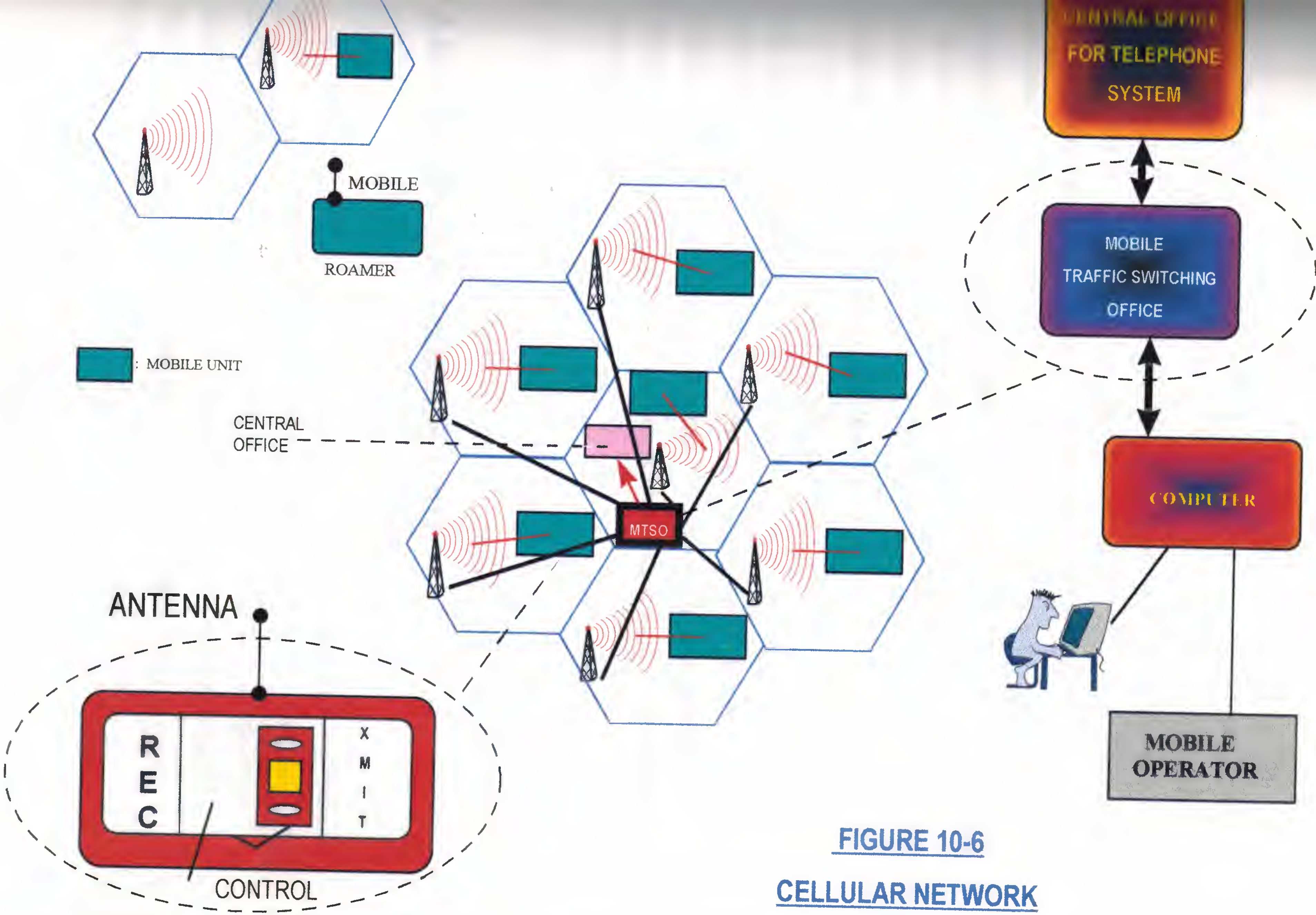


FIGURE 10-6

CELLULAR NETWORK

4.5- MOBILE UNITS

The mobile units consists of a control unit , a transceiver , and appropriate antennas. The transceiver contains circuits that can tune to any of the 666 FM channels in the 800 MHz range assigned to the cellular system . Each cell site has at least one set up channel dedicated for signaling between the cell and its mobile units . The remaining channels are used for conversation. Each mobile unit is assigned a 10 digit number , identical inform to any other telephone number. Callers to the mobile unit will dial the local or long- distance number for desired mobile unit. The mobile user will dial 7 or 10 digits with a zero or a one prefix , where applicable , as if calling from a fixed telephone .

Whenever a mobile unit is turn on but not in use, the mobile control unit monitors the data being transmitted on a set up channel selected from among the several stantard set up frequencies on the bases of signal strength. If signal strenght becomes marginal as the mobile unit aproaches a cell boundary, the mobile control finds a setup channel with a stronger signal.

4.6- CALLING A MOBILE UNIT

A mobile unit iscalled by transmitting its number over the setup channel. When the mobile recognizes itsnumber, it quickly seizes the strongest setup channel and transmits an acknowledgement response. The cell site then uses the seized setup channel to transmit the voice-channel assignment to the mobile. The mobile and cell site switch tothe voice-channel radio frequency and the voice channel is used for ringing, off-hook and subsequent conversation. The sequence is similiar when the mobile user originates the call. It begins with the mobile control unit seizing a setup channel when the mobile unit goes off-hook. Then the voice channel selection signaling and conversation occur in the same way.

During the call , the system at the serving cell site examines the signal strength once every few seconds . If the signal level becomes too low, the MTSO lookes for a cell site closer to the mobile unit to handle the call, based on the location and direction of travel information from the serving cell side. The actual handoff from one cell to the next occurs so rapidly that the user can not tell it has occurred.

When a call originates from a fixed telephone the number is sent out over the set-up channel , to which all mobile units are tuned . The mobile unit responds to its number by transmitting an acknowledgement , and then receives data telling it what voice channel has been assigned . Once on the assigned channel, the usually signaling functions and voice transmission occur.

4.7- ROAMERS

The system is designed to make handling of roamers automatic ; indeed , this is the principal goal of the cellular approach. Locating and handoff are concept that come directly from the use of small cells. "Locating" In this sense is not the determination of precise geographic location - although that is obviously a factor ; rather , it is the process of determining whether a moving active user should continue to be served by his current channel and transmitter , or " handed off " to either another channel , cell , or both. The decision is made automatically by a computer , based on signal quality and potential interference , and involves sampling the signal from the mobile unit .

The mobile unit automaticaly hands off the call to a cell and a channel that provides the optimum communication quality. The MTSO computer continiuously analyzes signal quality and makes the appropriate changes without any interruption in service .

With the cellular system, a subscriber could make a call from his car while driving in the countryside toward a city , continue through the city's downtown , and not hang up until well beyond the city on the other side During the entire time, the transmission would be clear with no dead spots . More importantly ,the swicthing of transmitters and frequencies during the conversation would be entirely automatic,with no interruptions and no action required by the user or an operator .

Wherever there is a system to serve it, a roaming unit wiil be able to obtain completely automatic service; however a call from a land telephone to a mobile unit which has roamed to another metropolitan area presents additional problems. While it would be technicaly possible for the system to determine automatically where the mobile unit is , and to connect it automatically to the land party ,there are two reasons for not doing so. First, the caller will expect to pay only a local charge if a local number is dialed. Second, the mobile user may not want to be identified to be at a particular location automatically by the system without an approval. Therefore the system will complete the connection only if the extra charge is agreed to, and when possible to do so without unauthorized disclosure of the service area to which the mobile unit has roamed .

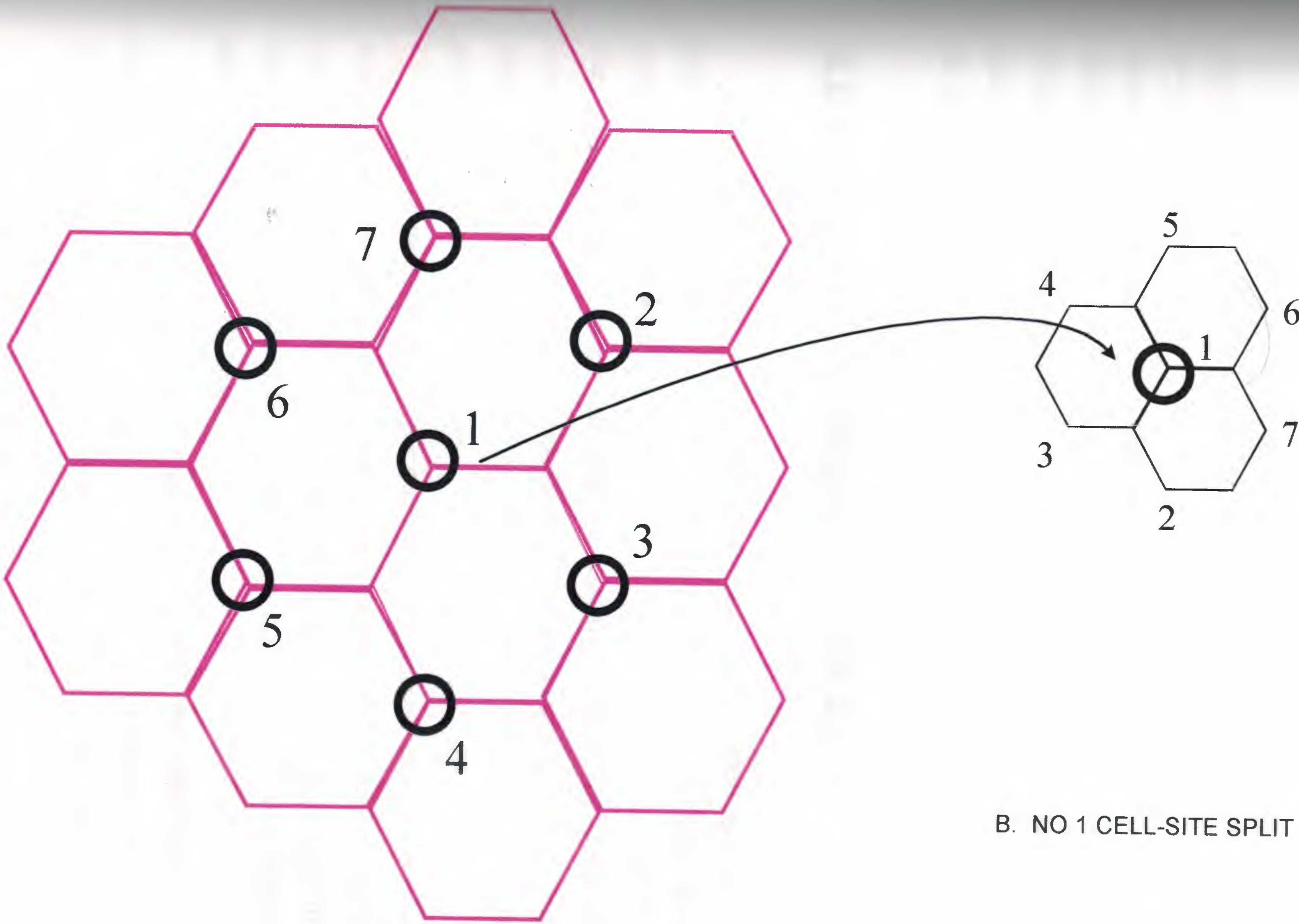
4.8- UNIQUE FEATURES

There are two essential elements of the cellular concept which are unique ; frequency reuse and cell splitting.

Frequency reuse means using the same frequency or channel simultaneously for different conversations , in the same general geographic area .The idea of having more then one transmission on a given frequency is not new ; it is done in virtually all radio services. What is unique to cellular is the closeness of the user ; two users of the same frequency maybe only a few dozen miles apart, rather than hundreds of miles. This is done by using relatively low-power transmitters on multiple sites, rather than single high - power trasmitter .

Each transmitter covers only its own cell , and cell sufficiently far a part maybe using the same frequency.

Cell splitting is based on the notion that cell sizes are not fixed and may vary in the same area or over time . The principle is shown in figure 10-7 . Initially ,all the cells in an area may be relatively large as shown in figure 10-7a . When the avarege number of users in some cells becomes too large to be handled with proper service quality , the overloaded cells are split into smaller cells by adding more transmitters , as shown in figure 10 -7b. The same MTSO can continue to serve all of the cell sites , but expansion of its computer and switching facilities probably will be required . Multiple frequency reuse is possible because of the lower transmitter power radiated in each cell , and by not using the same frequency in adjacent cells.
The cellular system can be expanded because cell splitting may occur as demand increases.



A. INITIAL CELL- SIDE PATTERN

B. NO 1 CELL-SITE SPLIT

FIGURE 10-7

CELL SPLITTING

5.0- GLOBAL SERVICE AND MACHINES (GSM) ARCHITECTURE

GSM as a modern telecommunications system is a complex object. To the multiservice aspect it shares with ISDN, it adds all the difficulties coming from cellular network. As such , its specification, its implementation and its operation are no simple tasks. Neither is its description. In the course of the specification of GSM, much effort was expended to sort out this complexity. Through we try to limit the use of the technical jargon ,it is unavoidable; when a concept or an object has to be referred to every third line , it is best to give it a name.

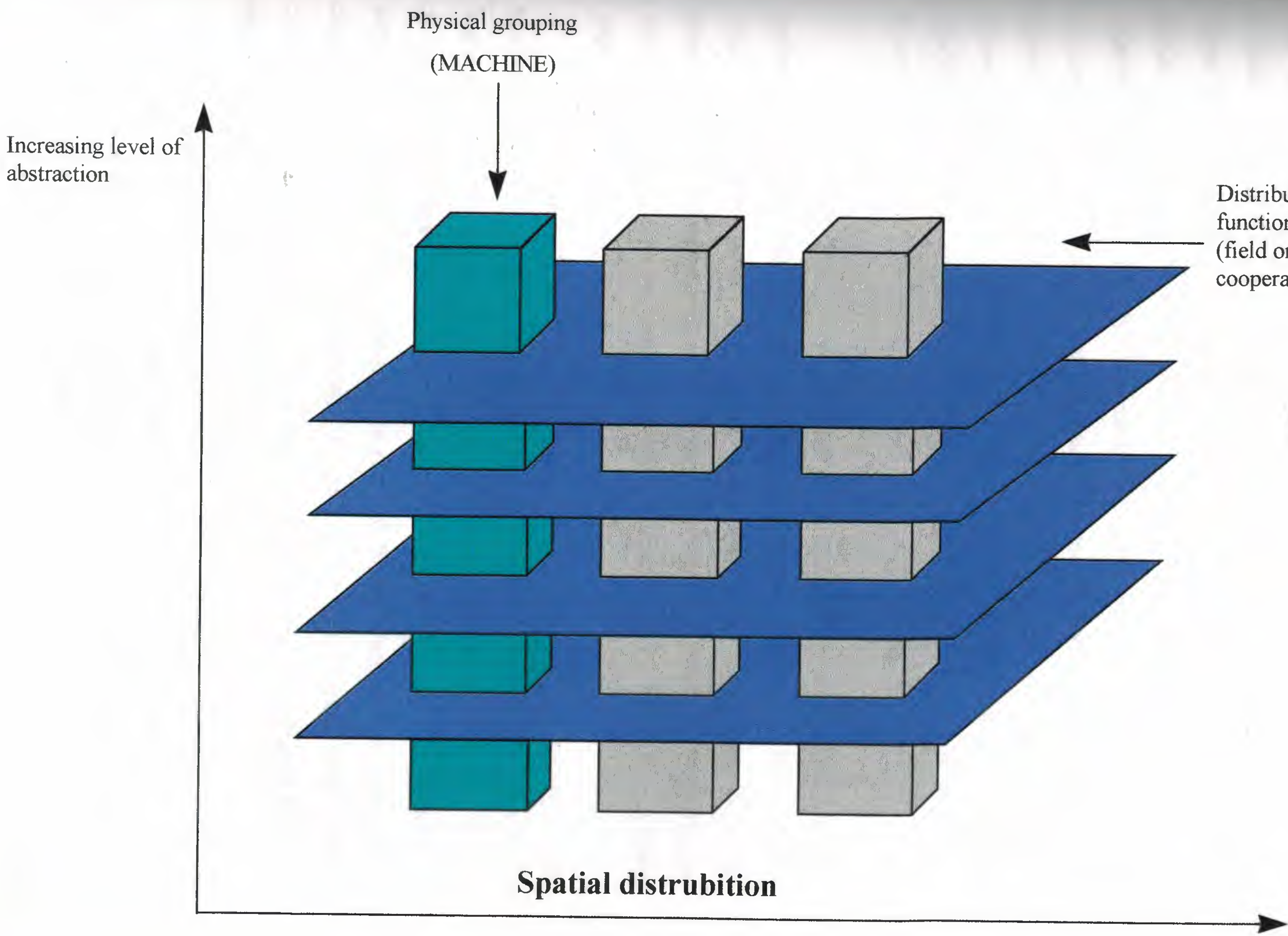
5.0.1- THE THREE DESCRIPTION AXES :

From one point of view , a telecommunication system is a collection of electronic boards transferring analogue electrical signals through wires or electromagnetic waves. But there is more to a system than this "reductionist" approach equating the system with electromagnetic field value or transistor states .The opposite view point would consist in looking at the system as a black box, seen only through its interfaces with the external world .

GSM is more than a concetanation of sub-systems: some areas involve many pieces of equipment and cannot be described satisfactorily by looking at each sub-system independently. Therefore , we must additionally look at how GSM operates from two different view points: a static one and a dynamic one..

The Static view point enables us to identify and describe several functions which are fulfilled through the cooperation of several machines . The term machine is used here to refer to an assembly of inter connected

Figure 2.1 Two dimensionalview of network



system components, physically close to each other , working together to perform identifiable tasks, The term function is often use in the technical literature (and in the specifications) to refer to some abstract machine .The use here is closer to the basic meaning of the word. A function is something to fulfill , an activity . In a sound architecture ,a machine corresponds to some function. However , other structures should also be identified, which group similar functions in a system .These functional groupings will put together elementary functions, possibly from different machines , which fulfill by cooperation the same goal. The usual representation ,such as used in the layering method of the Open System Interconnection model , consist in showing functional grouping over several machines as "horizontal" layers , a machine being a "vertical " structure in this representation , as shown in figure 2.1 .

The Dynamic view consist in looking at the events affecting the system .Events happen at many scales , from microseconds for transmission aspects, to years when one views the deployment of a network .The description of these events, their organization and of the way they trigger other events in turn is very important understanding how the system performs its functions.

The GSM architecture , which is the subject of this chapter , will first be described in term of machines ,then through a functional layer view . The subsequent chapters (the bulk of the book) will be devoted to the study of functional planes in detail ,going deeper into the role of each machine within each plane, with a substantial part of these chapters containing a description of event sequences . Having thus covered the three complementary axes (as shown in figure 2.2.) , the description should enable the reader to get a full and consistent picture of GSM.

Now although this three axes concept helps in structuring the description of system, it is not sufficient to tackle the overall complexity. The number of possible physical groupings (that is to say machines or abstract

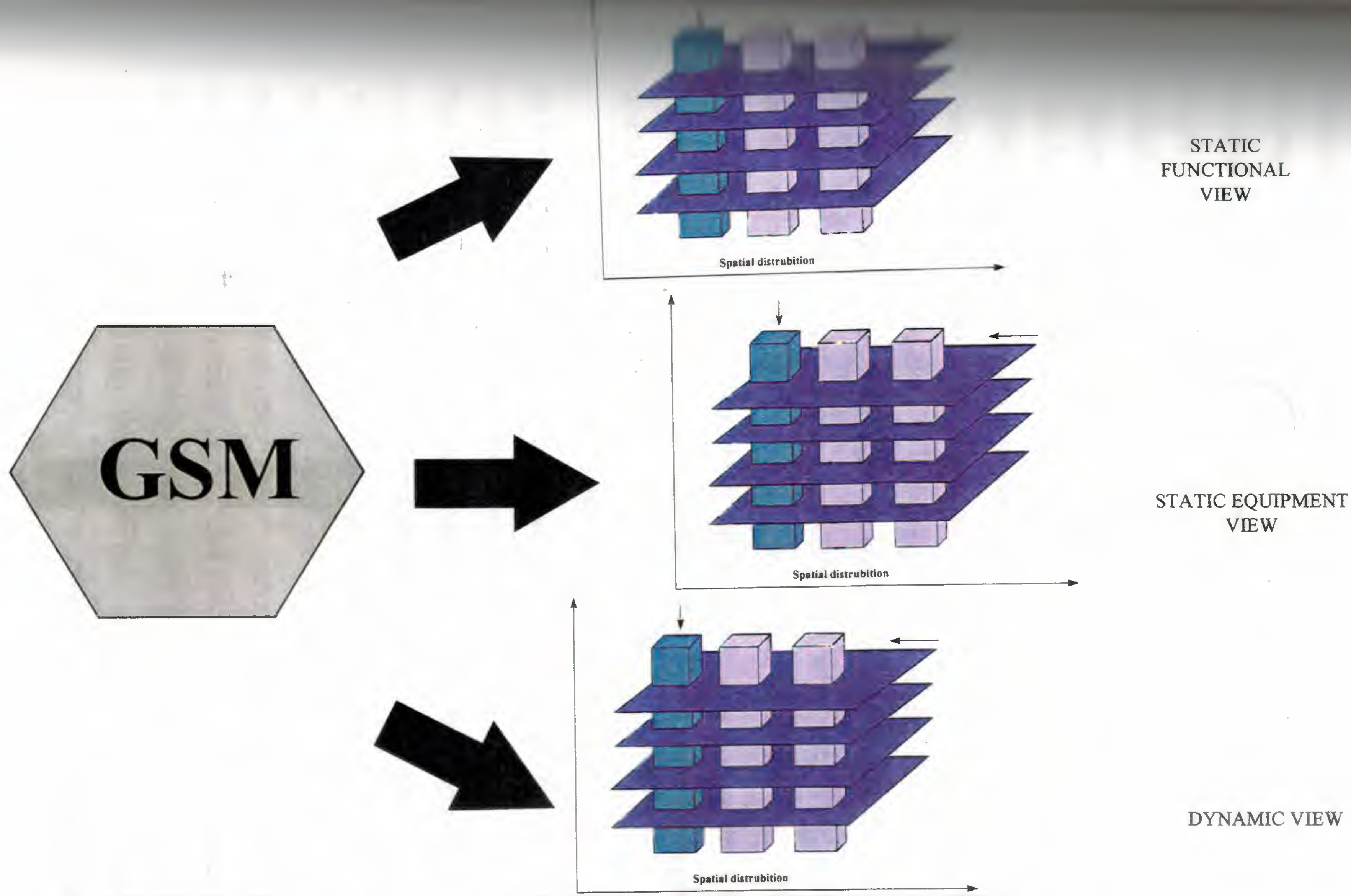


FIGURE 2.2 THE THREE AXES OF THE DESCRIPTION

portions of machines) or of planes or of event sequences ,is still rather big. The second trick we will use is a recursive description or top-down approach. Taking the horizontal axis as a first example , this approach consist of describing the system in a few (less than 5 when possible) sub systems, analyzing the interactions at this level while taking each sub systems as a black box, and then applying this method for each sub system similarly, functional planes will be composed of sub planes , themselves composed of sub - sub planes, and so on. Likewise for the temporal sequences , which can be analyzed from the large scale down to the small scale. This allows us to identify first general , compound events, of substantial duration (e.g. the call set up, its release) then stepping up the resolution (e.g. procedures , like channel allocation , start of ciphering), and so on down to elementary message exchanges , and finally to transmission and bit modulation . This is the "reductionist" part of the methodology , which must be balanced at each step by the analysis of the relationships between the sub parts, in order not to lose the large scale view. This will be helped considerably by the three -axes approach , each axis embodying relationship between entities along the other axes.

Physical groupings and their borders are two sides of the same coin. In fact , borders between machines are extremely important in a system such as GSM, since the specifications specify in fact mainly the behavior of the system as seen on interfaces between machines and not the internal working of this machines (through this can be derived to a large extent from the external behavior) . It is then an important function of the architectural model to define the system interfaces. An interface represents the frontier between two machines which are in contact via "transmission medium " . It should be noted that the specifications use in some place a wider meaning for this term ,as for instance when referring to the MSC to HLR interface (where the interface may well include a full signaling support network). In the following , both the machines and the interfaces involved at each stage will be indicated , with a particular emphasis on the interfaces for which the exchange of information is specified in the Specifications.

Interactions between functional layers can also be described in terms of interfaces . Because this interactions happen inside machines , they are not to be specified. However a detailed and formal description o.g. such interactions can often be found in the specifications (as the notation of primitives between layers) . In some cases this description takes up a substantial portion of the specifications , and is justified by the quest for as little ambiguous as possible. It should be understood that in the Specifications the description of the interactions between layers is a model , which does not constrain implementation in any way, though it can be useful guideline.

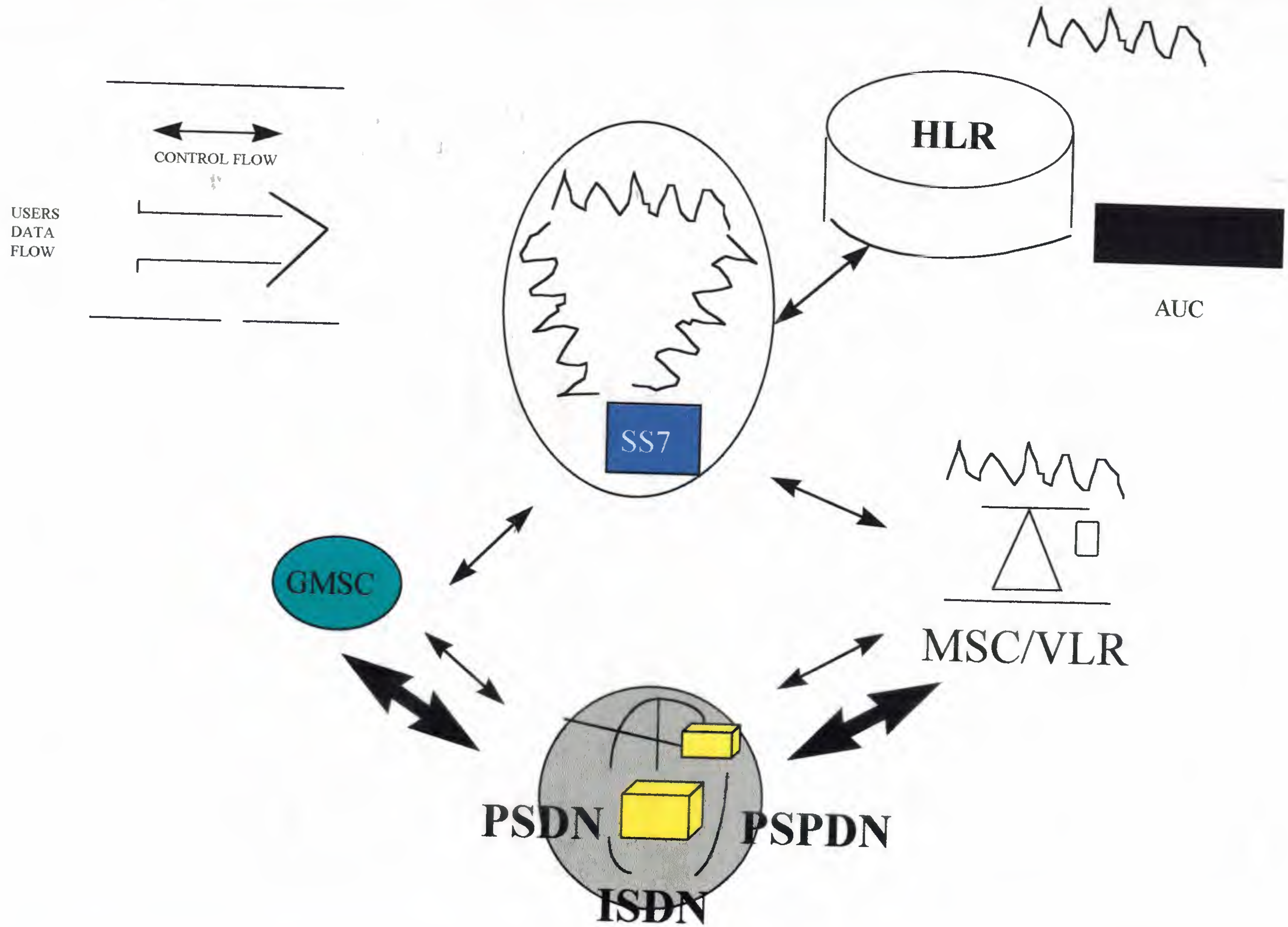
5.0.2- FRONTIERS OF THE SYSTEM : WHERE ARE THE BORDERS OF GSM ?

When looked at from a distance ,GSM is part of the Global Telecommunications Network , itself a part of the human organization. As such GSM is in direct contact with users (human beings or machines which are being provided with telecommunications services through GSM) , with other telecommunications networks (e.g. the global telephony network) and with the personal of the operating companies. These are indeed three main external interfaces of GSM, as shown in Figure 2.3.

Other interfaces with the external world exist, such as the contact of machines with air , ground and power supplies (which we may term environmental aspects) as well as other systems using the radio spectrum (electromagnetic compatibility -EMC-aspects). These pragmatic aspects , which are far from negligible for manufacturers and operators, if not for users, are not directly related to the provision of telecommunication services and will not be dealth with here. Let us now look at the three main border lines , with respectively users , other telecommunication networks and operators.

On the user side the limit lies somewhere between the user himself , who can be excluded from GSM, and the radio inter interface which represents the principal part of the specifications .The mobile stations are only partly specified by the specifications : an example is that terminal equipment functional entities similar to those defined is ISDN are not defined specifically for GSM : another example is the man machine interface of mobile stations, which is in no way specified in a binding manner in the specifications, and could include functions which have nothing to do with

FIGURE 2.3


CONTROL FLOW
USERS
DATA
FLOW
HLR
AUC
SS7
GMSC
MSC/VLR
PSDN
PSPDN
ISDN

GSM .The point of contact between GSM and the user lies therefore somewhere inside the mobile stations.

GSM is specified mainly as an access network ,enabling the setting up of calls between GSM subscribers and subscribers to other telecommunication networks. For practical reasons ,machines belonging to GSM are most often kept separate from machines belonging to other networks ; this comes from the division, which is now the general rule , between GSM operators and the fixed PSTN/ISDN operators ,even in companies of marked administrative origin such as FRANCE TELEKOM or DEUTSHE BUNDES TELEKOM. Other choices exist ; one could imagine telecommunication switches performing GSM functions as well as managing PSTN/ISDN subscribers. This is not excluded by the specifications ,but no canonical GSM architecture does not consider this possibility, and the interface between GSM and other telecommunication networks is clearly defined.

This interface includes three aspects. The first and major one concerns the point where communications transit between GSM and another network ; the GSM machine (a switching exchange) at the corresponding contact point is referred to as a "mobile services switching center " (MSC) . The second aspect of interfacing with fixed networks concerns the provision of basic telecommunication transport between GSM machines. In many instances, regulations do not allow GSM operators to operate the terrestrial links between their machines, often leaving them no other choice but leasing lines from a fixed network operator. This second point will be ignored, since in practice it has no affect on the functioning of GSM . The third aspect of interconnection with fixed networks, which is also within the realm of using external networks as support for GSM functions, concerns the routing of non-call related signaling between diferent GSM networks. For instance, the management of location data for GSM users roaming abroad, requires an exchange of data with no direct relation with given calls to be transmitted through an international signaling network .

There again GSM operators of many countries are forbidden to access the international signaling network directly , and must therefore interface with a fixed network operator in their country to transmit this signaling information between GSM entities.(MSCs , location registers) of different networks.

Although the point of contact (at least for the transit of calls) is well defined, the interfaces between GSM and external networks are not specified in the specifications. CCITT recommendations, as well as ETSI standards based on them, include such specifications, but usually national variants exist. Fixed network operators have their own peculiarities, and the machines in contact with their networks must be customized to meet the exact interface requirements specific to each network.

The border between GSM and operator personnel. As in the case of the user interface, the border is basically between machines and the human employees themselves. The set of the machines intervening between this boundary and the machines handling the telecommunications traffic (as well as some parts of these machines) are globally referred to as the Operation and Maintenance Sub-System (OSS). It includes various entities such as work stations (or terminals) handling the man machine interface with operator personnel , and dedicated computers managing a number of tasks required for operation and maintenance of system, as well as parts of the software of traffic handling machines themselves.

Most of the OSS aspects are not specified by the specifications . Only a small part of the interfaces between traffic handling entities defined in the GSM architecture is related to OSS functions, and interfaces between these entities and OSS machines are only partially specified.

5.0.3- INTERNAL GSM ORGANIZATION

This quick study of the GSM borders already hints at a first split of the internal GSM domain into sub systems. The mobile station (MS) and the OSS have already been identified as manifest sub systems. The remaining part consist of infrastructure machines whose roles are to provide and to manage transmission path between mobile stations and either fixed networks or other mobile stations, and to provide the means for the users to set up communications along these transmission paths.

The canonical GSM architecture distinguishes two parts: the BBS (Base Station Sub System) and the NSS (Network and Switching Sub System*) . The BSS is in the charge of providing and managing transmission paths between the mobile stations and NSS machines (namely the MSCs) , including in particular the management of the radio interface between mobile stations to the relevant network or to other mobile stations. The NSS is not in direct contact with external networks.

(* The term NSS is used by many operators and manufacturers, though not in the specifications) The interface between the BBS and the mobile station is already introduced radio interface , where as the interface between the BSS and the NSS has been named the A interface in the specifications. The MS,BSS and NSS form the operational part of the system, whereas the OSS provides means for the operator to control them. This model is shown in Figure 2.4 .

On this scale, the interactions between the sub systems can be grouped in two main categories. The bottom part of the figure corresponds to a chain:

external networks←→ NSS←→BSS ←→MS←→USERS

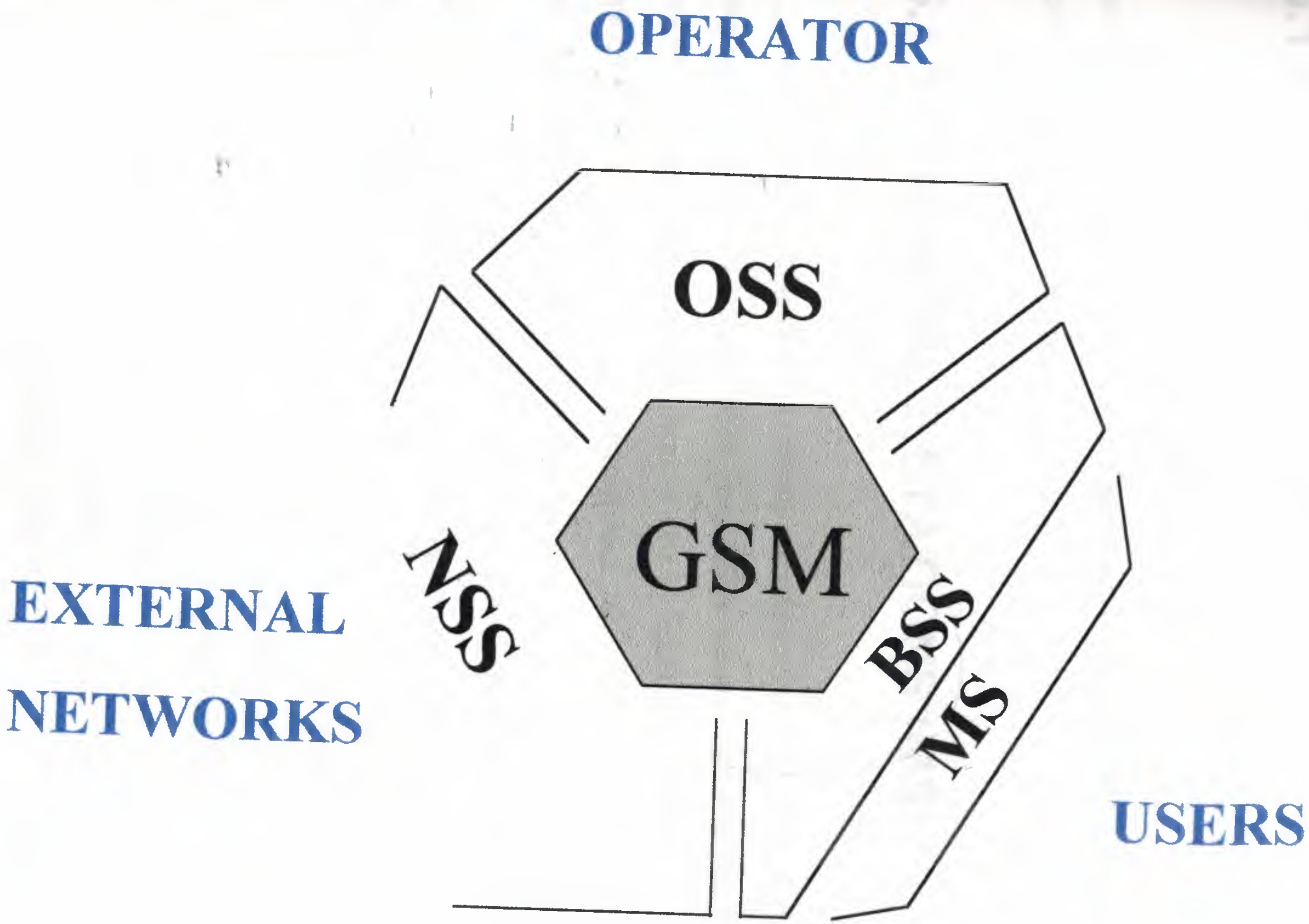


FIGURE 2.4 GSM SUB SYSTEM ORGANIZATION

Whose business as a whole is to provide transmission paths and means to establish them. This is operational part of the system, handling the telecommunications traffic. Above it we find the control part, composed of the OSS and the operator, which interacts with the traffic handling part by observing and modifying it so as to maintain or improve its functioning.

5.1.0- SUB SYSTEMS :

The main purpose is to introduce a number of terms , not to exhaust the subject. The level of detail here is just sufficient to get a general idea of the functional splits.

5.1.1- THE MOBILE STATIONS (MS) :

The mobile station usually represents the only equipment the user over sees from the whole system. Examples taken from the first types of GSM mobile stations to be on the market are in figures 2.5 and 2.6. Mobile station types include not only vehicle-mounted and portable equipment, but also handheld stations, which will probably make up most of the market.

Beside generic radio and processing functions to access the network through the radio interface, a mobile station must offer either an interface to the human user (such as a microphone,loudspeaker, display and key board for the management of speech calls), or an interface to some other terminal equipment (such as an interface towards a personal computer or a faxsimile machine), or both of them. An effort has been made to allow of the shelf terminal equipment to be connected to mobile stations (for instance group 3 faxsimile machines designed for connection to the telephone network), and specific terminal adaptation functions have been specified for this purpose. However, all implementation choices are possible and left open to manufacturers, enabling fully integrated compact mobile stations to coexist with mobile stations featuring standard interfaces.

This leadsto the identification of three main functions as shown in figure 2.7.

*The terminal equipment' carrying out functions specific to the service without any GSM sp[ecific functions e.g. a fax machine.

*The mobile termination carrying out among others all functions releated to transmission on the radio interface :

*Possibly the terminal adaptor which acts as a gate way between the terminal and the mobile termination. A terminal adaptor is introduced when the external interface of the mobile termination follows the ISDN for a terminal installation, and the terminal equipment has a terminal to modem interface.

The functional split between mobile termination , terminal adaptor and terminal equipment is very much related to the transmission needs of each service .

GURE 2.5

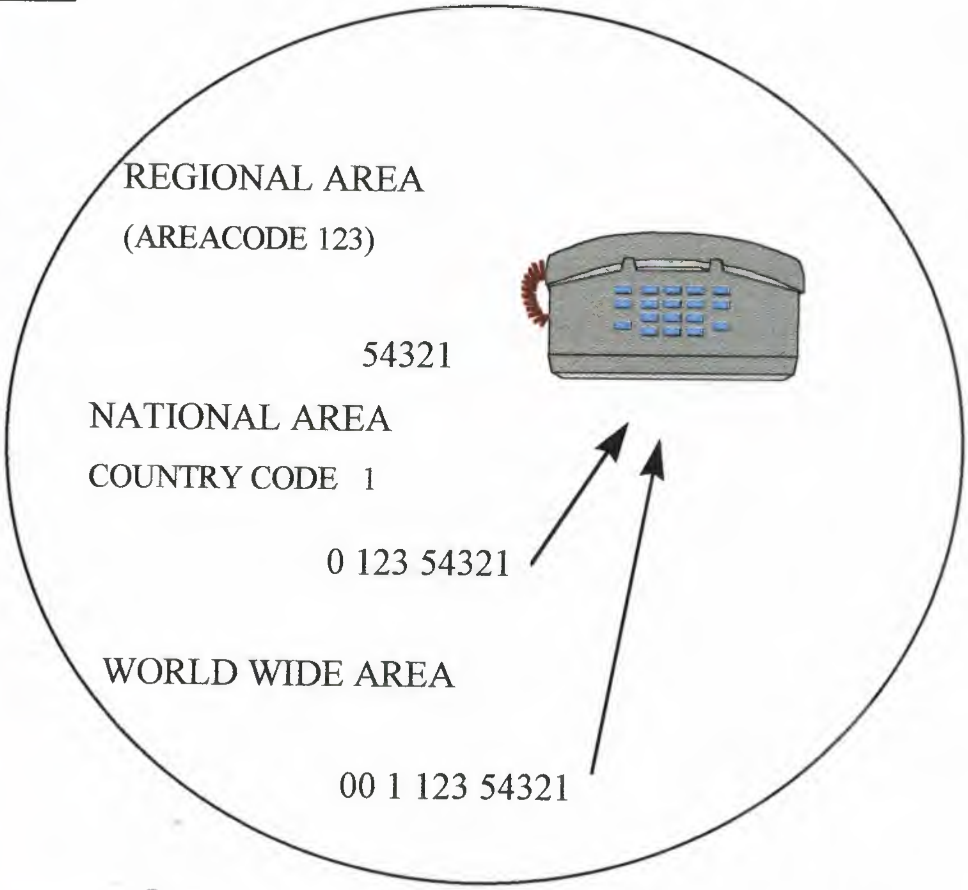
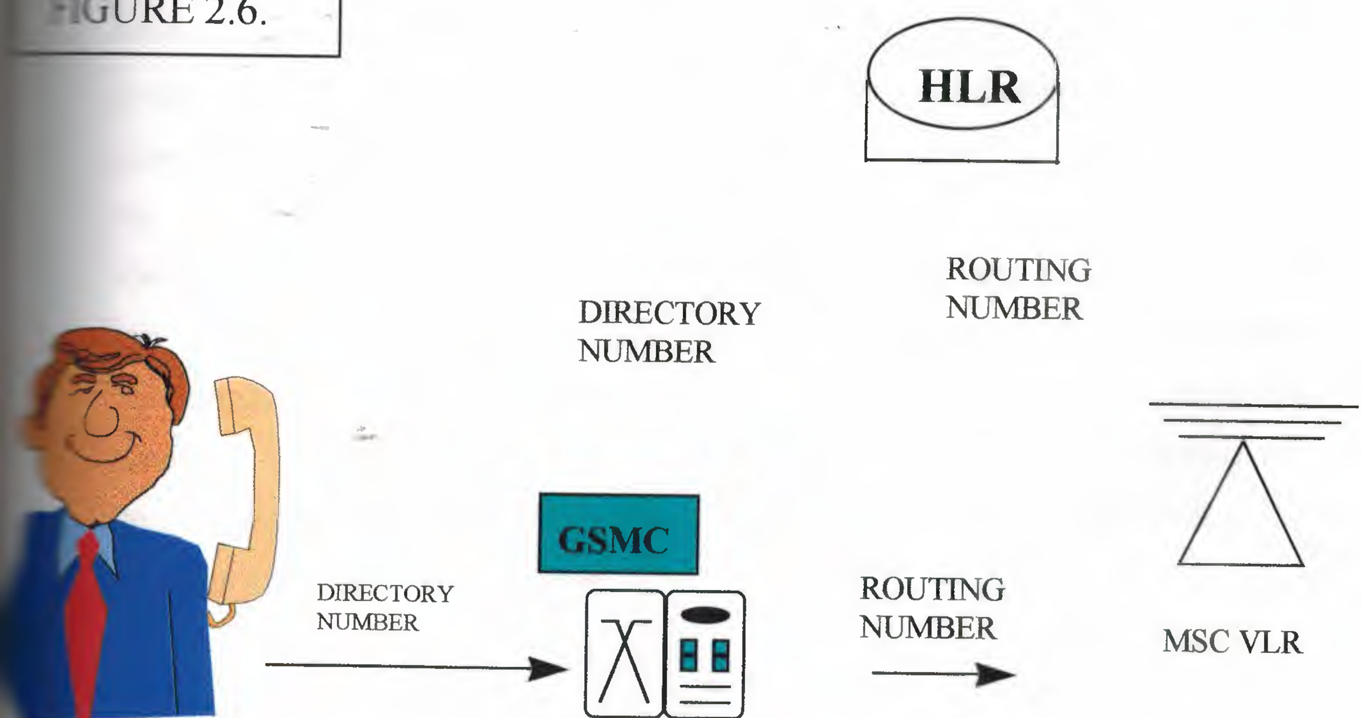


FIGURE 2.6.



Another more significiant architectural aspect of the mobile station relates to the consept of subscriber module or SIM (subscriber identity module a slightly restrictive name as more than identity is involved) . The SIM is basically a smart card (or a cut out thereof) , following ISO standards, containing all the subscriber releated information stored on the user`s side of the radio interface . Its functionalities ,besides this information storage capability ,relate also to the confidentiality area. The rest of the mobile station contains all the generic transmission and signaling means to access the network. The interface between the SIM and the rest of the equipment is fully specified in the Specifications, and is simply referred to as the " SIM-ME " interface (ME stands for mobile equipment .

The term Mobile Station (MS) shall generally include a mobile equipment and a SIM ,although a rare case exist where a mobile station could be operated without a SIM for the handling of anonymous emergency calls when so permited by the network.

The consept of a removable storage device for subscriber data has far reaching consequences . In previous cellular systems, except for the German C network which introduced the smart card concept at the time when it is making its way in GSM committees, the personalisation of the mobile station required a non trivial intervention, only possible for technical specialist and not for the operator`s administrative clerks. This situation lead to several drawbacks. A mobile station could only be sold by specialist dealers, able not only to install the equipment in a vehicle but also to act as an intermediary between the user and the service provider to personalise the equipment. Should the mobile station fail (unfortunately not such a rare event) it was diffucult to provide the user with a replacement during the repair period and almost impossible to allow the user to keep the same directory number during this time.

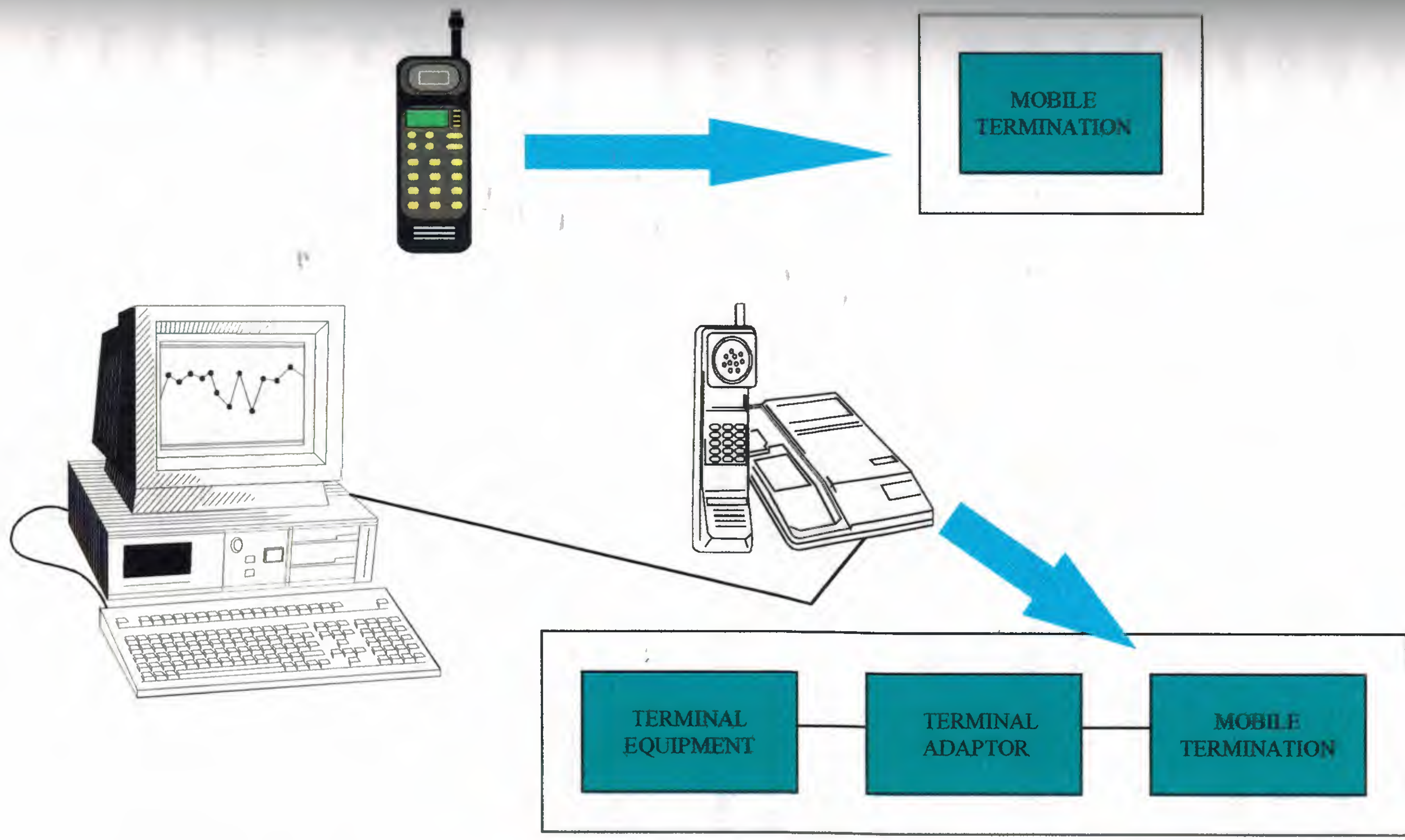


FIGURE 2.7 MOBILE STATION FUNCTIONAL ARCHITECTURE

5.1.2- THE BASE STATION (BSS):

Largely speaking, the base station subsystems groups the infrastructure machines which are specific to the radio cellular aspects of GSM . The BSS is in direct contact with mobile stations through the radio interface. As such, it includes the machines in charge of transmission and reception on the radio path and the management thereof. On the other side , the BSS is in contact with the switches of the NSS. The role of the BSS can be summarized as to connect the mobile station and the NSS and the hence the mobile station`s user with other telecommunications users. The BSS has to be controlled and is thus also in contact with the OSS. The external interfaces of the BSS are summarized in Figure 2.8.

According to the canonical GSM architecture, the BSS includes two types of machines ; the BTS (Base Tranceiver Station), in contact with the mobile stations through the radio interface ,and the BSC (Base Station Controller), the latter being in contact with the swithes of the NSS . The functional split is basically between a transmission equipment ,the BTS and a managing equipment ,the BSC.

The BTS comprises radio transmission and reception devices, up to and including the antennas, and also all the signal processing specific to the radio interface. BTS scan be considered as complex radio modems, and have little other function. A typical first-generation BTS consists of a few racks (2m high and 80cm wide) containing all electronic devices necessary for the transmission functions, as shown in figure 2.9. for a GSM900 BTS and figure 2.10 for a DCS 1800 BTS. The antennas are usually a few tens of meters away, on a mast, and the racks are connected to it throught a feeder cable. A one-rak first-generation BTS is typically able to handled three to five radio carriers, carrying between 20 and 40 simultaneous communications. Reducing the BTS volume is important to keep down the cost the cell sites, and progress can be expected in this area.

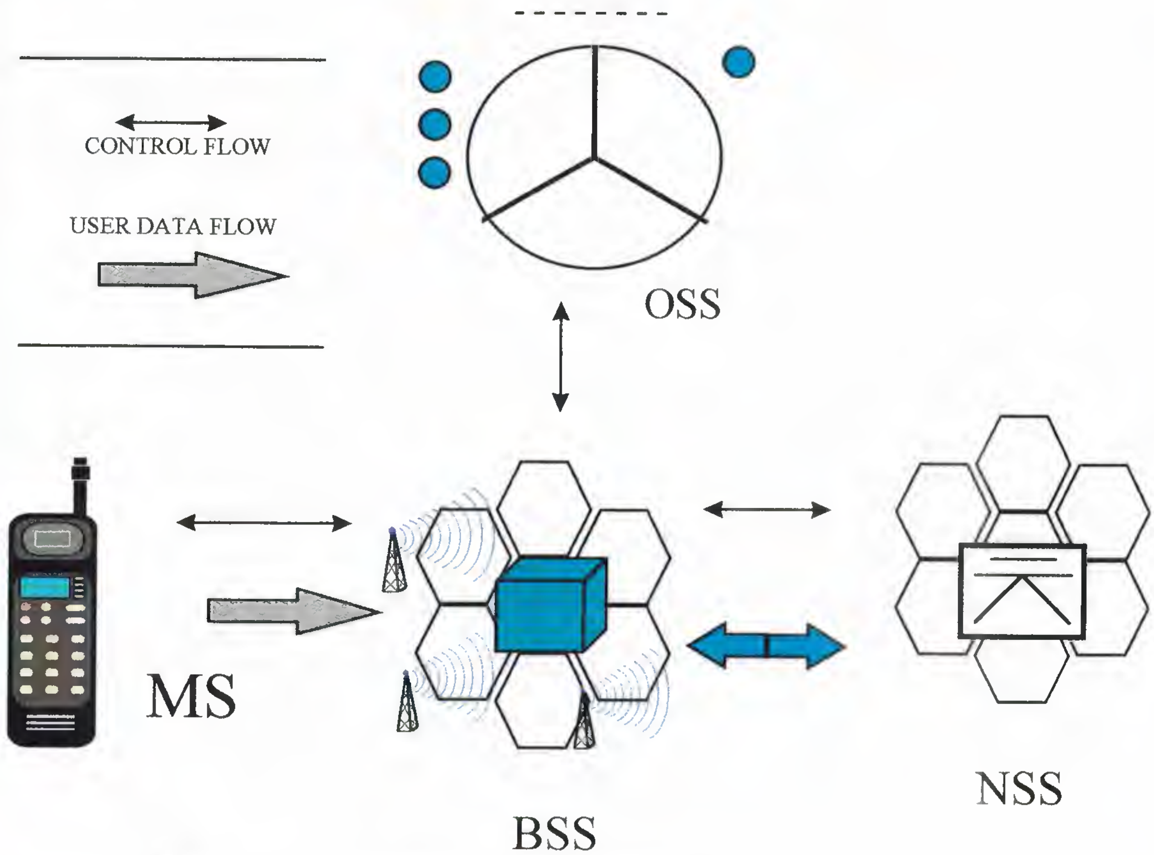


FIGURE 2.8. THE EXTERNAL ENVIRONMENT OF THE BSS

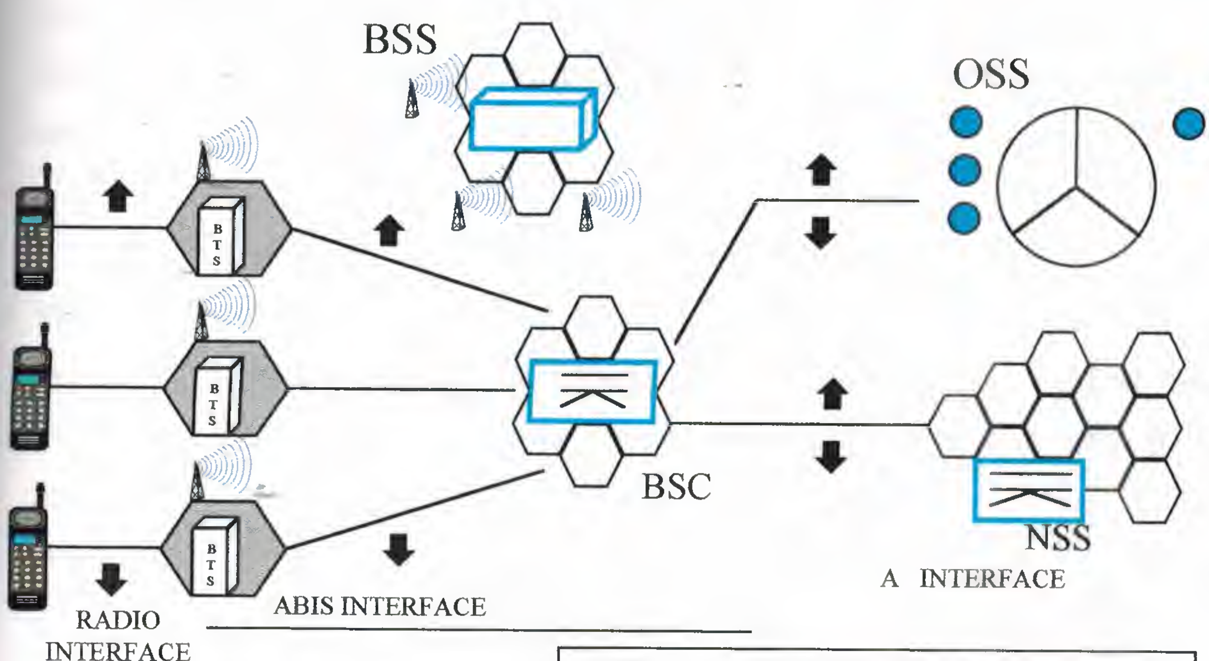


FIGURE 2.11. BSS COMPONENTS AND INTERFACES

An important component of the BSS, wich is the considered in the canonical GSM architecture as a part of the BTS, is the TRAU, or Trascoder/rate Adapter Unit. The TRAU is the equipment which the GSM-specific speech encoding and decoding is carried out, as weel as the rate adptation in case of data. Although te Specifications consider the TRAU as a sub-part of the BTS, it can be sited away from the BTS, and even more so since in many cases it is actually between the BTS and the MSC.

The internal stucture fo the BSS is represented in figure 2.11. On top of the BTS, it shows the second "canonical" component of the BSS, the BSC is in charge of all the radio interface management through the remote command of the BTS and the MS, mainly the allocation and release of the radio channels and the handover management. The BSC is connected, on one side, to several BTSs and on the other side, to the NSS (more exactly to an MSC).

A BSC is in fact a small switch with a substantial computational capability. Its main roles are the management of the channels on the radio interface, and of the handovers. A typical BSC consist of one or two racks, as shown in Figure 2.12 and can manage up to some tens of BTSs, depending on their traffic capacity.

The concept of the interface between BSC and MSC, called the a interface, was introduced fairly early in the GSM standard elaboration process. Only later was it decided to also standardise the interface between BTS and BSC, and this interface therefore bears the (not any more meaningful than A!) name of "Abis"
interface.

In the GSM vocabulary a BSS means the set one BSC and all the BTSs under its control , not to be confused with the BSS as the sub system including all the BSCs and BTSs.

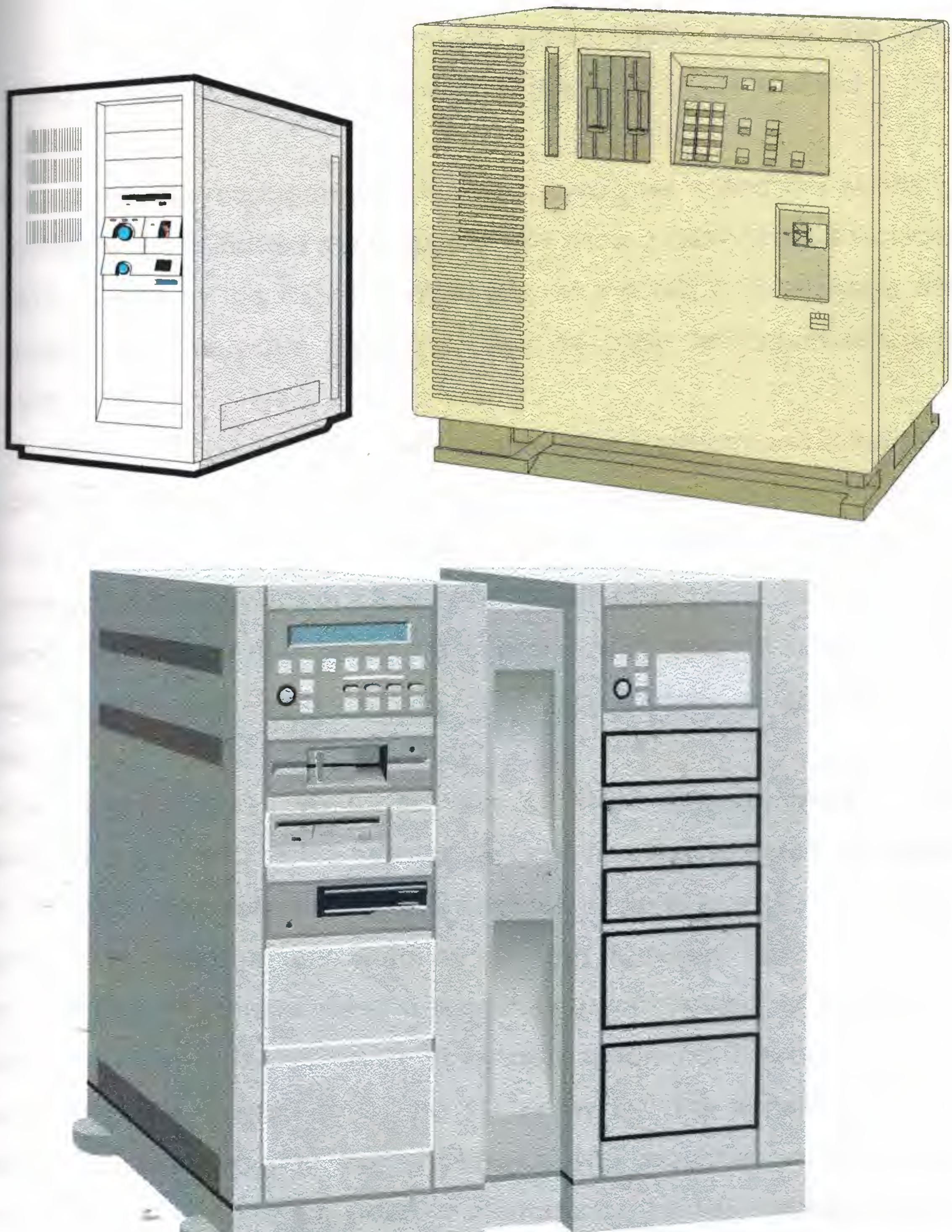


FIGURE 2.9-2.10 A GSM BTS AND A DCS 1800 BTS

5.1.3- THE NETWORK AND SWITCHING (NSS) :

The network and switching sub system (NSS) includes the main switching functions of GSM , as well as the data bases needed for subscriber data and mobility management . It is also sometimes called the switching sub-system which is indeed more appropriate since a GSM network includes the BSS as well as the NSS. The main role of the NSS is to manage the communications between the GSM users an other telecommunications network users.

Within the NSS ,the basic switching function is performed by the MSC (mobile services switching centre) , whose main function is to coordinate the setting up calls to and from GSM users. The MSC has interfaces with the BSS on one side (through which it is in contact with GSM users) , and with the external networks on the other. The interface with external networks fob communication with users outside GSM may require a gateway for adaptation (interworking functions , IWF) , the role of which may be more or less substantial depending on the type of user data and the network it interfaces with. The NSS also needs to interface external networks to make use of their capability to transport user data or signaling between GSM entities. In particular , the NSS makes yse of a signaling support network, at least partly external to GSM ,following the CCITT signaling system no :7 protocols (and therefore usually referred to as the SS7 network) ; this signaling network enables co-operative interworking between NSS machines within one or several GSM networks . The external interfaces of the NSS are represented schematically in Figure 2.13.

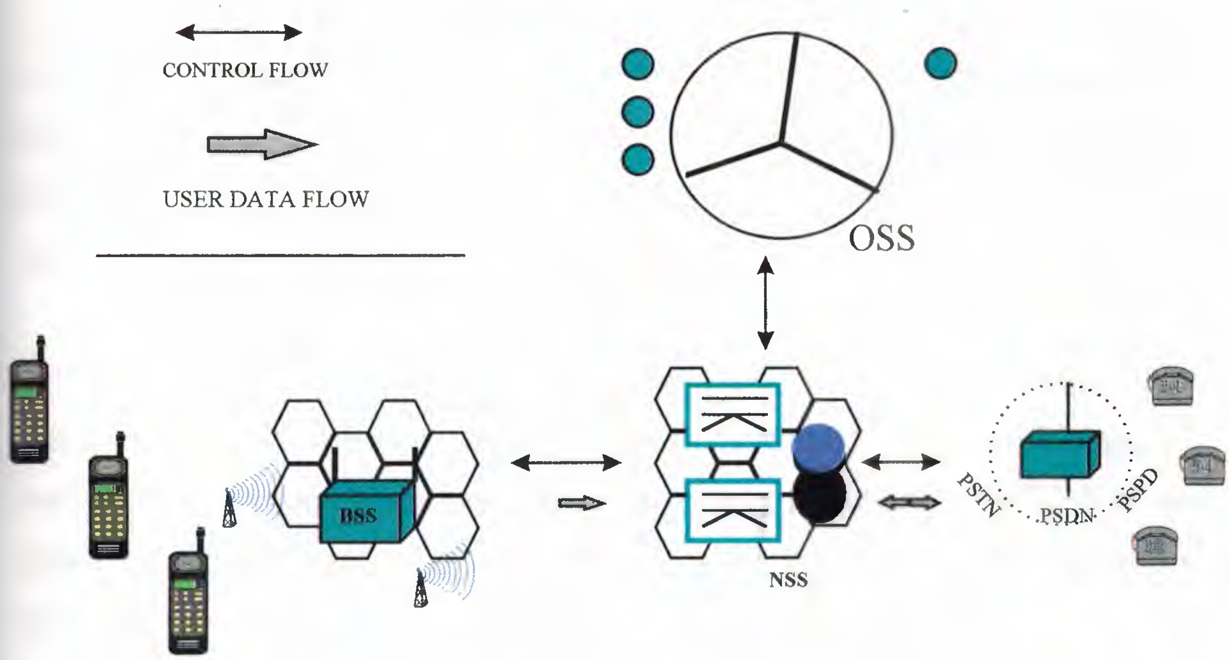


FIGURE 2.13 THE EXTERNAL ENVIRONMENT OF THE NSS

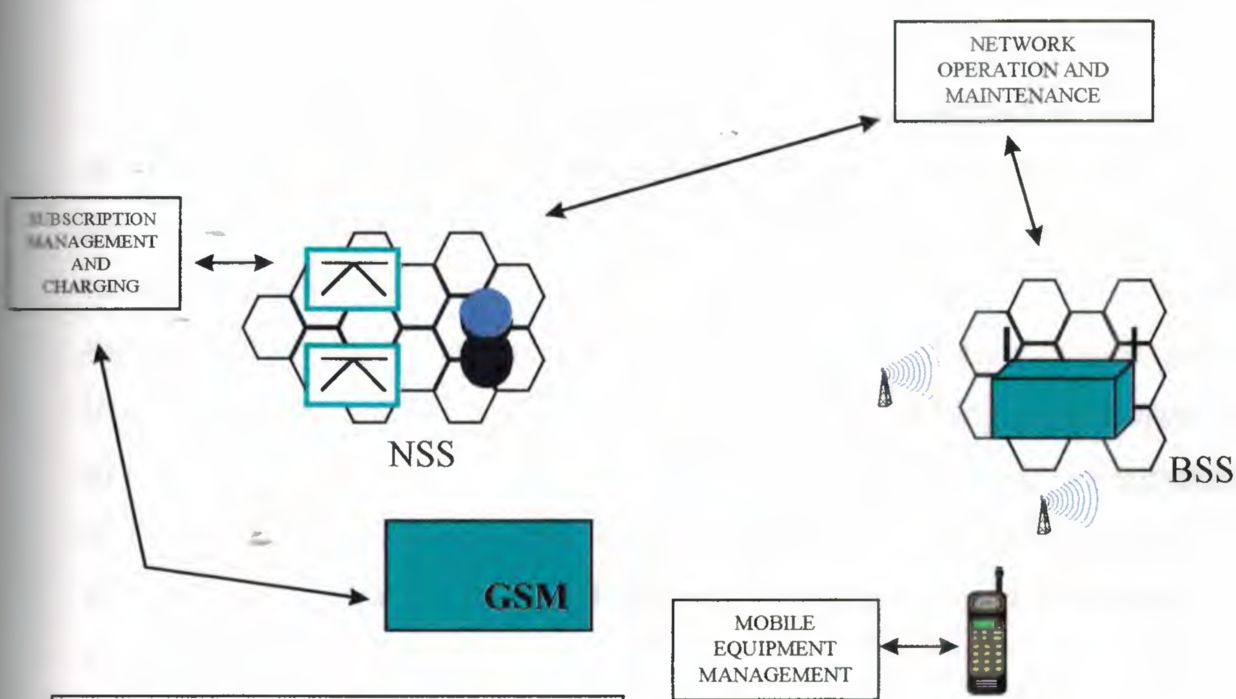


FIGURE 2.16. OSS ORGANIZATION

As a piece of equipment, an MSC controls a few BSCs and is usually a rather big switching machine. With a medium population penetration percentage, a typical MSC at the date of writing is suitable for covering a regional capital and its surrounding, totalling say 1 million inhabitants. Such an MSC includes about half a dozen racks.

The interconection of the MSC with certain networks requires the adaptation of the GSM transmission peculiarities to those of the partner network. This adaptations are the Interworking Functions (IWF). This term refers by extension to the functional entity in charge of them. It basically consist of a transmission and protocol adaptation equipment. It enables interconnection with network such as PSPDNs (Packet-Switched Public Data Networks) or CSPDNs (Circuit-Switched Public Data Networks), but it is also exits when the partner network is simply the PSTN or ISDN. Interworking functions may be implemented together with the MSC function, or they may be performed by a separate equipment. In the second case, the interface between MSC and IWF is left open by the Specifications.

Besides MSC, the NSS includes data bases. Subscriber information relevant to the porvision of telecommunications services is held on the infrastructure side in the HLR (Home Location Register), independently of the actual location of the subscriber. The HLR also incudes some information related to the current location of the subscriber. As a physical machine, an HLR is typically a standalone computer, without switching capabilities, and able to handle hundreds of thousands of subscribers. A functional subdivision of the HLR identifies the Authentication Centre (AuC), the role of which is limited to the management of security data for the authentication of subscribers.

The second database function identified in GSM is the VLR (Visitors Location Registers), linked to one or ore MSCs, and in charge of temporarily storing subscription data for those subscribers currently situated in the service area of the corresponding MSC (s), as well as holding data on their location at a more precise level than the HLR.

But the NSS contains more than MSCs, VLRs and HLRs. In order to set-up a call towards a GSM user, this call is first routed to a gateway switch, referred to as GMSC, without any knowledge of the whereabouts of the subscriber. The gateway switches are in charge of fetching the location information and of routing the call towards the MSC through which the subscriber can obtain service at this instant (the Visited MSC). To do this, they must first find the right HLR, knowing only the directory number of the GSM subscriber, and interrogate it. The gateway switch has an interface with external networks for which it provides gatewaying as well as with the SS7 signalling network to interwork with other NSS entities. The term GMSC is somewhat misleading, because the GMSC funotion is not by technical necessity linked to an MSC. It could be thought of as an independent equipment, or as a function integrated an a digital telephony switch.

Having seen the pieces, let us look at the glue. Depending upon national regulations, a GSM operator may or may not be allowed to operate the full SS7 network between NSS machines . If the GSM operator has the full control of this signalling network,than Signalling Transfer Points. (STPs) will probably the part of the NSS functions and could be implemented either as stand alone nodes or in the same machines as the MSCs,in order to optimise the cost of the signalling transfer transport between NSS entities.(MSC/VLRs,GMSCs,HLRs...)

Similary, depending upon terms of its licence , a GSM operator may have the right to implement its own Network for routing calls between GMSC and MSC or even fo routing outgoing calls as near as possible to the destination point before using the fixed network. In the case Transit

Exchanges (TE not to be confused with Terminal Equipment as used fob the mobile station architecture) may be part of the GSM network as well and there again may be implemented a stand-alone nodes or together with some MSCs. As a summary figure 2.15 shows the main components of a GSM NSS and the interconnections between them.

5.1.4- OSS ORGANISATIONS

Network operation and equipment maintenance concern all machines (including by the way the oss machines), whereas subscription management has impact on at least the HLR.

The canonical architecture described in the specifications is not as specific on operation in maintenance aspect of the GSM as it is on the rest. A wide latitude is left to operators and manufacturers. One sound reason for this is that the issue is not specific to GSM .Operation and maintanence functions are necessarily implemented in existing networks, and a lot of standardisation work on the issue is going on fob general application to telecommunications networks.

Up to some point in the past, operation and maintenance actions were performed locally by invervention on the side of each machine. To this avail , each equipment was provided with some man machine interface ,such as through a local terminal .The coordination of the actions on different machines was managed by human beings. Such an approach could still be adopted fob GSM, at least at the very beginning , for experimental networks .For instance , subscription management could be done by manually entering the subscription data on an on-site terminal connected to the HLR. With the local only approach the OSS functions are simply spread in the BBS and NSS machines , and the only "OSS" machines are the man machine devices such as terminals .

Now , with the evalution of the tecnology and of the complexity of telecommunications systems, the range of possible actions on the system as well as the quantity of information to be processed have increased tremendously . The local only approach is then inefficient as soon as the number of machines to operate becomes significiant . Some centralisation is required , and this calls fob seperate pieces of equipment intervening between several of the traffic handling machines and the man machine devices . Moreover ,these pieces of equipment can perform some of the coordination functions instead of human operator thus offering a better guarantie of consistency between the configuration of the different machines . The ultimate centralised approach is the concept of TMN (Telecommunication Management Network) , where all operation and maintenance machines compose a network which as a whole is linked to all the traffic handling machines .

Once some centralisation is appilied interfaces between traffic handling machines and OSS machines appear requiring some specifications . It is at this level that most of the substance related to operation and maintenance can be found in the Specifications. This part of the standard has been designed with the ideas of TMN in mind, in order to enable a smooth integration of GSM networks with advanced operation and maintenance machines.

Network operation and maintenance proper calls fob mediation between the operator personnel and all the machines. If TMN principles are followed, the operation network is linked on one side to the telecommunications machines (MSCs, BCSs, HLRs, and others, but not the BTSs which are accessed through the parent BSC),. Onthe other side, the operation network is linked to workstations acting as man-mabhine interfaces. The Specifications identify the OSS machines directly in contact with BSS or NSS machines. These dedicated machines are called the OMCs (Operation and Maintenance Centres), and are linked to a few telecommunications machines of the same category (and usually of the same manufacturer), e.g:an "OMC-R" (OMC-Radio) would be in charge of a few BSCs and, through them, the corresponding BTSs. In a simple management architecture, the OMC is autonomous, and includes the man-machine interface fob the control of the traffic handling machines it is linked to. Such an OMC is typically a standalone workstation. The OMCs can also become simply the gateways to an overall management network, acting as mediation devices, following the terminology of TMN.

Subscription management corresponds to tasks which are independent from the other operation and maintenance functions, and which may then be supported by machines separate from those involved in network operation. Subscription management has two facets, subscriber data management and call charging. The Specifications do not address at all the first aspect, and only lightly the second. Different architectural approaches will be adopted by the different operators.

Subscriber data management involves only the HLR (including the AuC for the security related data) and dedicated OSS machinees, for instance a network linking the HLR and the-machine devices in the commercial agencies where subscribers are dealt with. This network, if it exists, can be autonomous, in contact with the rest of the system only at the HLR. The SIM is also affected at this stage. It has to be initiallised consistently with the information held in the infrastructure. The SIM has then twop phases in its life. In a first "administrative" phase, it is dealt, with by the subscription management system, bur it is not active in the traffic handling part of the system. Then, once initialised, it is operational and can be inserted into a mobile equipment.

The second facet of subscription management is call charging. In a mobile environment, the call tickets relative to a given GSM subscriber can be issued by many different machines, including all the MSCs the user may visit.

Finally, the last domain of operation and maintenance is the management of mobile stations. A part of it is done within nework operation, through the infrastructure machines. There is however one machine identified in the canonical GSM architecture and specific to mobile station management, the EIR (Equipment Identity Register). It is a database which stores data related to Mobile Equipment. As explained earlier, a mobile station consists of a SIM and a Mobile Equipment (ME). Subs cribers data management concerns the SIM and is handled by the HLR and VLR. Because SIMs can be moved between Mes, overseeing subscribers does not imply overseeing mobile stations. Though the management of the mobile equipment is not absolutey necessary for the operation of telecommunications services, it holds some interest, e.g.when searching for stolen mobile equipment or when monitoring mobile station misbehaviour. For this purpose, the EIR is in charge of storing the relevant ME-related data. It interfaces with other NSS entities and with the network operation system. The interface with NSS machine is, again, Though the SS7 signalling support network.This is the reason why the EIR is often considered as part of the NSS, though its fuctional role sets it within the OSS. Figure 2.16 summarises the general organisation of the OSS.

5.2.0- FUNCTIONAL PLANES :

Up to this point description of the GSM architecture has focused on the physical grouping of funstions. The division of the system into there sub-system already bears a strong functional flavour. This reflects the fact that the chice leading to grouping some functions in the same machines depends on the closeness of those functions. However, a number of functions are by essence distributed, and can be fulfilled only by the co-operation of distant machines. This is obvious in any telecommunications system, since the basic function, the management of communications, is distributed. As a consequence, most functional entities described so far perform tasks in several spread functinal areas. For instance, machines such as the MSC or the BSS have necessarily some operation and maintenance functions; another example is the involvement of the MSC in BSS-related functions such as handover.

5.2.1- LAYER MODELLING PROTOCOL ARCHITECTURE

In the telecommunications domain, a powerful method to obtain a functional grouping is to use the Open System Interconnection model. Functions are grouped in functional planes, represented as stacked one upon the oyher. The lowest plane, devoted to the physical transmission of information between distant entities, relies on physical hardware media, whereas the highest one represents the view of external users. Each plane (or layer) provide services to the next layer up, these services being themselves enhancements of the services provided by the next layer below. Machines or entities are represented vertically, the intersection between entity E and layer L corresponding to the functions fufilled by E to contribute to the objectives of L.

Besides the hierarchical organisation, based on the notion of service provided by one layer to another, there is an underlying temporal organisation. In generali lower layers correspond to functions having a short time scale, whereas the upper layers will group long time scale functions.

Within each layer, the entities co-operate to provide the required service, through information exchanges. The rules of these exchanges are specified at referance points where the information flow crosses an interface between different entities. This rules are called the signalling protocols.

The distinction between an interface and a protocol is important. An interface represents to point of contact between two adjacent entities, and as such it can bear information flows pertaining to several different pairs of entities, e.g: several protocols. For instance in GSM is the transit point for messages pertaining to several protocols: between MS and BTS (for transmission), but also between MS and BSC (for the management of the transmission over the radio interface), MS and MSC (for the management of "users" mobility, and of the communications), or even MS and HLR (for setting Supplementary Services parameters), as shown in figure 2.17.

This is an example of the analysis of an interface as a protocol stack, each element of the stack being related to the intersection of a functional plane and of the interface. Still, signalling messages pertaining to a given protocol may be visible on several interfaces along their path, if the corresponding peer entities are not adjacent. The protocol then appears on several interfaces. Thus, the specification of a protocol is somewhat distinct from the specification of an interface. The specification of an interface can be reduced to the description of its protocol stack. This conceptual distinction is ill resolved in the Specifications, where often what is called an "interface" specification is in fact a "protocol" specification. For the purpose of protocol specification,the slicing of functions in planes must lead to fairly thin "slices", in order for each one to be consistent and also to escape from too big a complexity for the corresponding protocols.

Five planes will be distinguished in this part, as shown in figure 2.18. At the bottom lies the basis of any telecommunications system, i.e: the transmission plane. It provides transmission means for the communication

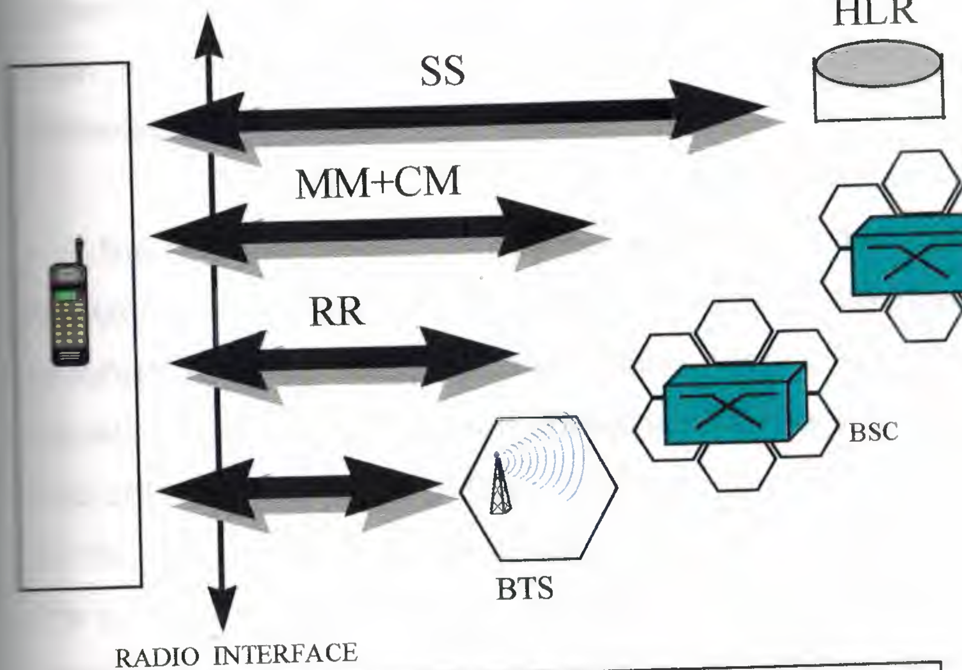


FIGURE 2.17. PROTOCOLS VERSUS INTERFACES

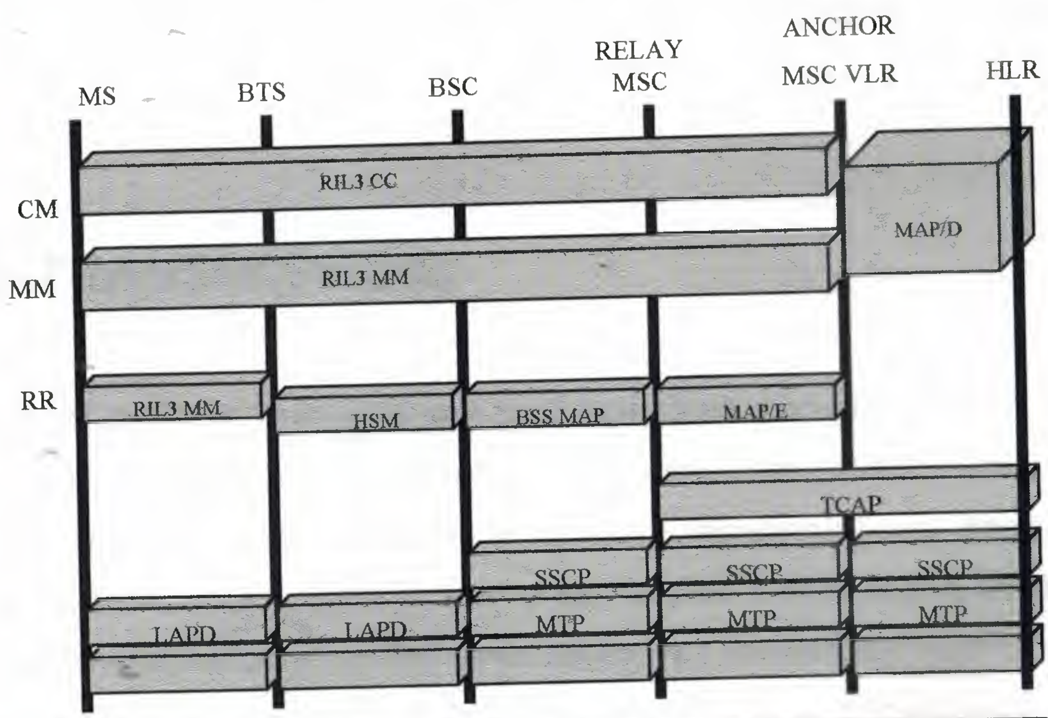


FIGURE 2.20. GSM SIGNALLING ARCHITECTURE

GSM MACHINES (SHOWN AS VERTICAL LINES) AND FUNCTIONAL LAYERS (SHOWN AS HORIZONTAL LAYERS OF BRICKS) DEMARCATE PROTOCOLS A STACK OF WHICH CAN BE DEFINED ON EACH OF THE INTERFACES.

needs of the users, as well as for information transfer between the co-operating machines. Transmission is a domain for very short time scale events, from microseconds (e.g: bit modulation) to seconds (for message transmission).

The next plane up is concerned with the management of transmission resources. In the telecommunications network, these functions are usually grouped with the communication management functions, because fixed circuit management represents a small portion thereof. However, in the case of a cellular system such as GSM, the management of transmission resources on the radio path is a complex issue and it warrants its own functional plane. This is called the radio resource management layer, or RR layer. The RR layer provides stable links between the mobile stations and the MSCs, coping the particular with the movements of the users during the calls (handovers) . The BSS performs most of the RR functions. From atemporal point of view, this plane and the two next ones deal with events on the call, that is to say from seconds to minutes.

5.2.2- TRANSMISSION:

Some of the GSM machines are concerned with transmission only. An obvious example is the transcoder and rate adaptor unit (TRAU), which is only concerned in adapting speech or data representations. Another example is provided by a transit exchange, whose role is limited to the routing of signalling exchanges between distant NSS entities. But most other GSM machines also play a more or less complex role in transmission.The mobile station obviously does so, and so does the BSC, the MSC and the interworking function (IWF), which may all be along the transmission path between two users. Conversely, some of the machines have no reletion to transmission, except for the minimum needs concerning signalling with the other machines. This include the data bases (HLR, VLR, EIR) and the OSS in general.

As already mentioned, the transmission plane includes two more or less independent functions. The first one is the prodive the means to carry user information (whether speech or data) on all segments, along the path followed by a communication. The second one is to provide the means to carry signalling messages between entities. The transort of the signalling is needed between adjacent macines (i.e:MS to BTS, BTS to BSC, BSC to MSC), but also through networks suh as the SS7 network used between NSS machines.

Include in this plane are aspects indeed traditionally called transmission, such as modulation, coding, multiplexng, but also other aspects such as low level protocols to format data, to ensure proper sequencing, to corret errors through the repetitions and the route information throughout networks.

5.2.3- RADIO RESOURCE MANAGEMENT(RR):

The role of the radio resource management layer is to establish and release stable connections between mobile stations and an MSC for the duration of a call for instance, and to maintain them despite usermovements. It must cope with a limited radio resource and share it dynamically between all needs. The functions of the RR layer are mainly performed by the MS and the BSC. In addition, since the responsibility for the handover process lies entirely within the RR layer, part of the functions implementing the MSC are within the RR domain, in particular the ones related to inter-MSC handovers.

5.2.4- MOBILITY MANAGEMENT

The machines concerned with mobility management are mainly the mobile station (and more precisely the SIM inside the mobile station) the HLR and the MSC/VLR. The management of the security functions are done by the same machines and more particularly by the AuC inside the HLR. The BSS is not concerned with the MM plane .

5.2.5- COMMUNICATION MANAGEMENT (CM)

The functions of the communication management layer or CM layer consist in setting up calls between users their request , as well as of course maintaning these calls and releasing them, It includes the means for the user to have some control over the management of the calls the origates or receivers , through the suplementary services the variety of the communication management functions makes it easier to describe as three sub-domains.

5.2.6- CALL CONTROL

The MSC/VLRs , GMSCs,IWLFs and HLRs, through the basic call management functions ,are able to manage most of the circuit oriented services to GSM users ,including speech and circuit data. This functional core represents a sub part of the CM layer, and is called CC (Call Control) in the specifications .

An important aspect of communication management beside establishing , maintaining and releasing calls , is the routing function e.g. to choise of transmission segments linking distant users and their concentration through switching entities , GSM mostly relies on external networks to perform this task , interfacing these networks through MSCs and GMSCs . The IWF may also have a switching function for communications to ands from, the networks it interfaces with .Call management requires access to the subscription data , in order to check the profile of the subscriber, and therefore the HLR also intervenes in the CM layer.

5.2.7- SUPLEMENTARY SERVICES MANAGEMENT :

A second aspect of the CM layer concerns the management of the so called suplementary services. Users in GSM have some control on the way their calls are handed by the network , this potentially is described as " SUPLEMENTARY SERVICES ", each one of them corresponding to some specific variation of the way the basic service is rendered to the user. The impact of suplementary services on calls is mainly a CC function. The entities involved in SS management are very few ; the mobile station and HLR are the only entities involved.

5.2.8- SHORT MESSAGE SERVICE

The last aspect of the CM layer is related to the point to point short message services (SMS-PP) . For the purpose of these services , GSM is in contact with a Short Message Servis Centre.

(SM-SC) A service centre may be connected to several GSM networks. IN each of these ,one or several function entities are in charge of interfacing the SM-SC . However , the Specifications have special terms for the gateway functions when applied to short messages. They define two types of such entities : the SMS-GMSC for mobile terminating short messages (SMS-MT/PP) and the SMS-IWMSC FOR mobile orginating short messages (SMS-MO/PP) . The ole of the SMS-GMSC is identical to the role of the GMSC for incoming speech or data calls. The role of the SMS-IWMSC is much less obvious and adds little value to the service except providing a fixed GSM point of interconnection for an SM-SC , rater than enforcing its connection to the SS7 network which would enable information transfer with any MSC.

5.2.9- INTERFACES AND PROTOCOLS AN OVERVIEW :

Basically ,this section lists the main GSM protocols , with our terminology. The Specifications do not always give an abbreviation or a name to the protocols . This is the main reason for our introduction of new terminology, in area already riddled with acronyms and other jargon terms. However ,there is a need for a short term used consistently to refer to each protocol.

Figure 2.20 shows an overview of the signalling architecture in the machines of a GSM transmission chain , as far as the telecommunications functions are the concerned (i.e. without showing explicitly the OAM functions). The horizontal axis coresponds to spatial distribution, starting with the mobile station on the left most part of the diagram and going through the various infrastructure machines on the way . The vertical axis coresponds to the functional planes , starting from the bottom with transission and going up through the different layers described in the previous section.

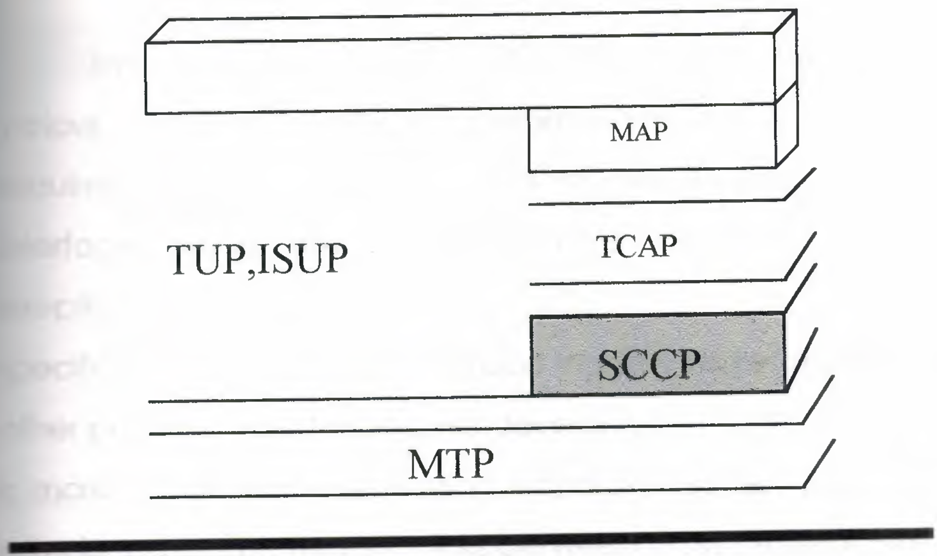


FIGURE 2.21. STACK OF PROTOCOLS ON THE SS7 INTERFACE

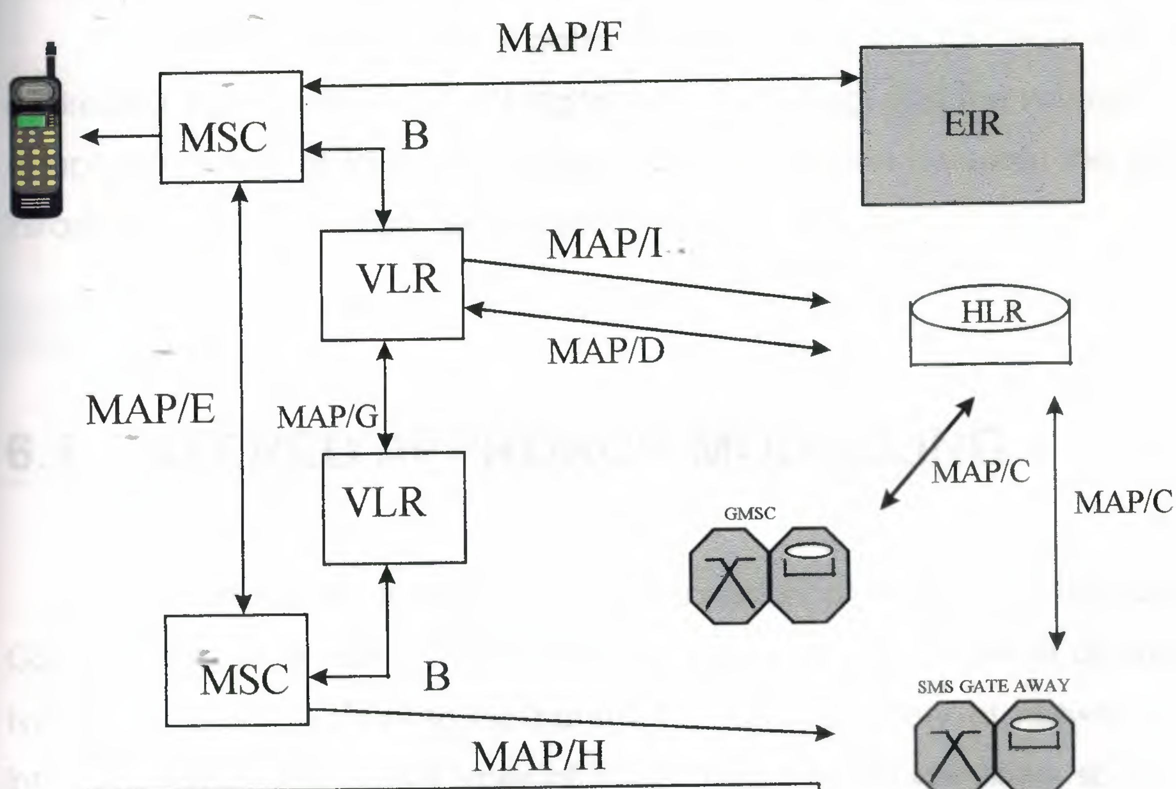


FIGURE 2.22 MAP/C TO MAP/I PROTOCOLS

6.0- TRANSMISSION

The multiservice nature of GSM repuires that it interconnects with various kinds of external networks, each with their own transmission requirements. As far as internal interfaces are concerned, the radio interfaces usually the focal point in a cellular network. GSM is no exeption, and the

specifications of its radio interface include more original features than any other public radio interface yet developed. Though transmission on fixed links is more constranied by existing standards, some new features have been introduced on the terrestial links between GSM infrastructure machines.

Despite this variety, the whole system must provide consistent end-to-end transmission paths, taking into account different oftimisation schemes on the successive along the way. This calls for translation functions between some of the transmission segments, and is a source of complexity.

The way in which user is data transformed along the way will be addressed; this includes the GSM digital speech coding, and the various rate adaptation schemes for data services. Then transmission between the GSM infrastructure machines will be detailed.

6.1. LAYERED APPROACH MODELLING

The transmission of GSM give an impression of complexity. Because GSM is aimed at providing multiple services, there are a number of different types of information flow to be transmitted, and a variety of networks to interface with. Three broad types of information can be identified; speech; various information formats such as text, images, facsimile, computer file, messages and so on, grouped under the vague term of data; and finally the

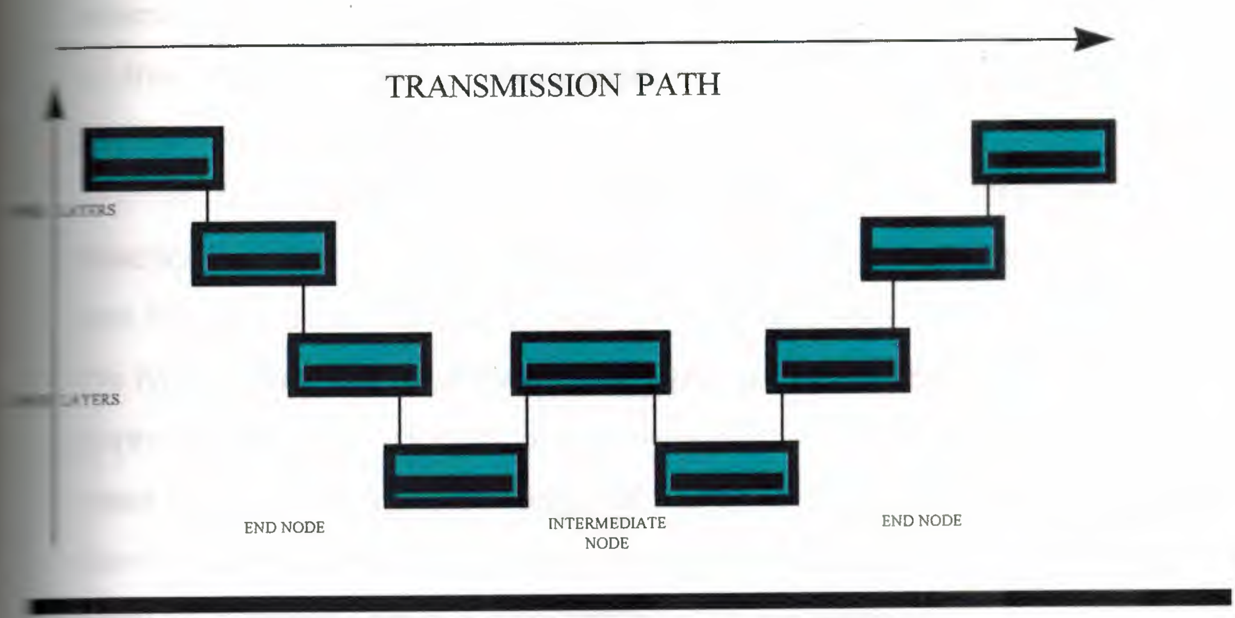


FIGURE 3.1 LAYERED APPROACH

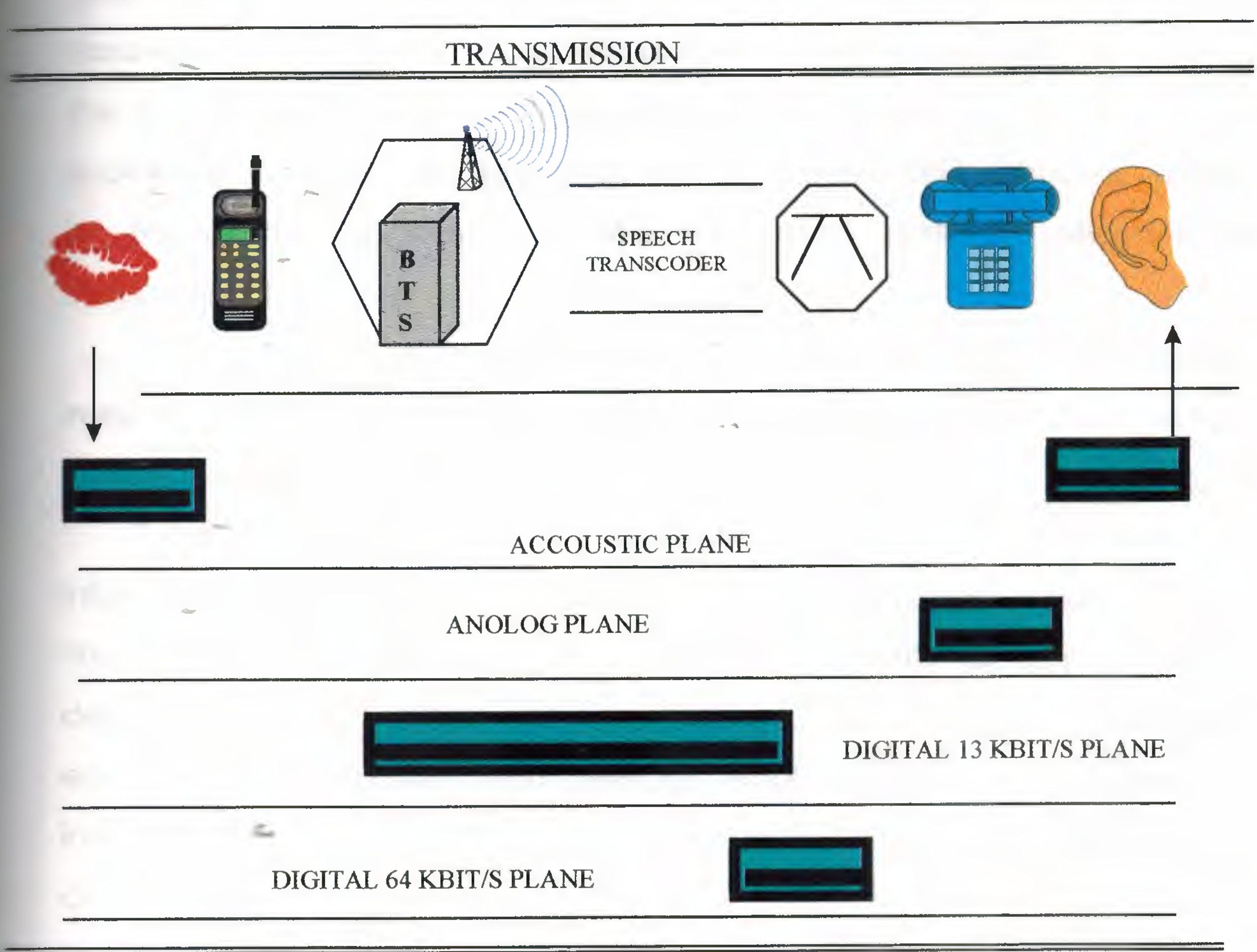


FIGURE 3.2 SPEECH REPRESENTATIONS

internal signalling messages. Depending on the possible transit network, and on the network of the correspondent, the same kind of information may be transmitted in different ways.

In fact, this seemingly complex jungle can be structured in two directions. A first "horizontal" structure stems from the need for consistent end-to-end connections A number of them defined, according, typically, to the type of transmitted information layering design approach. With such an approach, a given portion of the transmission paths uses machines which need not have a full knowledge of the type of data to be given physical connection can be peeled little by little like the different layers of an onion, each layer representing a different level of knowledge.

If we represent the layering along a vertical axis, with the raw information at the bottom, and the distilled product at the top, and if the transmission path follows a horizontal axis, we obtain a representation such as the one in figure 3.1. Intermediate nodes need not be concerned with the knowledge of upper layer semantics. As a interfaces can be much simplified, by taking into account only the information attributes relevant for transporting it.

If we apply this model, a first split along the vertical axis consists inseparating two domaings, which will be described in the two main sections of this chapter:

* The upper and outermost end-to-end domain is concerned with the information as dealth with by the final users. This domain is by nature rather "horizontal", i.e: each type of information warrants its own individual description (speech, differents kinds of data,....). When describing the data end-to-end transmission path, special emphasis well be put on the interforking between GSM and other networks. At this stage, GSM will be considered as a
"black box" obeying certain rules.

*The lower and innermost transmission domain is concerned with the way in which the information, regardless of its upper-layer semantics, is transported inside GSM. In the corresponding section, the GSM "black box" well be opened, gradually unfolding its internal organisation and interfaces.

6.2- TRANSMISSION OF SPEECH SIGNAL BETWEEN GSM AND USERS

First the transmission of speech between a GSM user and another user in the GSM or in some telecommunications network accessible through the PSTN or the ISDN. Then the other types of transmission will be addressed, that is to say the transmission of non-speech data between GSM users in network such as the PSTN, GSM, ISDN, packer or circuit data networks.

Telephony is by far the most popular service offered by public networks, including fixed networks and mobile celluler networks. After a general presentation of how voice is transported.

From mouth to ear through these networks.

Starting with the GSM subscriber's end, these transmission planes unfold as follows (figure 3.2): acoustic transmission, analog transmission, digital transmission at 13 kbit/s (this transmission being performed in two different ways over the radio path and between the base station and the speech transcoder), and finally another digital transmission mode in which speech is represented by a 64 kbit/s signal.

Closest to the end user is the acoustic trasmission plane; digital transmission at 13 kbit/s and 64 kbit/s as well as analog transmission are found furter down the transmission path. The transmission means vary from one interface to another, even with in the same transmission plane.

6.3- TRANSMISSION SEGMENTS AND INTERWORKING FUNCTIONS :

In this description, the transmission mode at 13 kbit/s is clearly specific to GSM, and is therefore part of the GSM domain. The precise border between the GSM domain and the external world is however rather subjective, and is done by defining two reference points within the transmission path. The first point is between the user's mouth and the microphone: the handset is considered to be an integral part of the GSM domain. The second reference point is between the MSC and the switch of the external network to which the MSC is directly connected.

The 64 kbit/s digital coding is part of every basic telecommunication curriculum. The analog signal is sampled at a rate of 8 kbit/s; this operation limits the bandwidth to 4 kHz. Each sample (which has an analog value) is given an integer value after the application of a logarithmic compression law known as A law. Each value is coded as an 8 bits symbol. The outcome is then a flow at a rate of 8 kbytes/s, i.e: 64 kbitt, The transcoding between the analog signal and its digital A law representation includes an analog process (the pre-emphasis), sampling, a linear analog-to-digital conversion of the samples giving a result on 13 bits, and finally a coding process which transforms the 13 bits samples into an 8 bits code. All this processes are specified in more detail in the CCITT Recommendations, in particular in Recommendations CCITT G.11/G.714.

The effect of the speech transmission methods used in the PSTN and even in the ISDN cannot be neglected by the GSM transmission, since they do not provide a true reconstruction of the original acoustic signal. In both cases, the high frequencies of the signal are filtered out. Fortunately, this does not raise a problem. The lower part of the spectrum is also distorted, and this has some undersireble consepuences. In the analog case, the 0-300

Hz. Band is completely filtered out. Even with digital transmission, this band usually disappears in the fixed correspondent's handset. The main consequence is that the two directions, from a GSM user to a PSTN user and vice versa, are not identical. In the first case, the signal is processed first by GSM machines before any distortions introduced by the other network. The result is that transmission quality is, barring other differences, better in the mobile to fixed direction.

Another network interworking problem for speech communication is the problem of echo. The final segment in the PSTN uses a two-wire cable, and there is necessarilly a 2-weri/4-wire adaptation somewhere, which is a first source of echo. The other sources are local to the terminal installation. The GSM transmission introduced a large delay, amounting roughly to half that which is encountered with a communication involving a satellite link. Since it is considered this echo is particularly disturbing to the user when delay is more than 25 ms, an echo contro function must then be provided at the boundary between GSM and the PSTN to avoid the negative impact of a delayed echo.

6.4- DATA SERVICE:

Non-speech services, or data services, cover the exchange of a lot of different types of information. Data transmission encompasses the exchange of text, of drawings, of computer files, of animated images, of messages, and so on, An important part of the information processing is done at the two extremities, in a machine most often outside the scope of the Specifications. We call such a machine "Terminal equipment" or more simply "terminal",though in some cases it can be a complex installation, such as a videotex server or a message handling system.

The main functions performed by a data terminal in the realm of end-to-end information are the following:

* Source coding, which transforms back and forth text, images, sound, etc. In the international currency of the world of information which are the binary digits;

*End-to-end protocol, dealing with the organisation of the communication, juggling with such concepts as pages, sessions, languages;

*And, most important, the presentation of the information to the user, by display, sound generation, printing, and so on.

In most cases it is possible to confine such processes to the ends of the transmission chain. This enables the reduction of the number of different cases which need to be taken into account by eliminating the need to study the intermediate transmission devices. The relevant characteristics distinguishing the different cases are few, and include the bit rate, the acceptable transmission delay (fixed of variable) and the maximum acceptable degradation due to transmission errors. The concept of bearer capability is used to describe and to refer to what is provided by the intervening equipment, i.e: the transportation of information between two user-network interfaces. A bearer capabilitiy is then characterised mainly by the attributes listed above.

GSM can be connected to a variety of external networks, since we are not yet in the promised land of broadband ISDN where a single international long-distance network supports all possible telecommunication services. Examples of network include the good old PSTN (this ubiquitous telephone network which is still the principal carrier of data transmissions), Packet Switched Publec Data Networks (CSPDNs).

The existence of an external network divides the transmission path in two segments. The segment between the GSM user terminal and the boundry point is entirely within GSM. But the other segment, from the boundry point to the other terminal, is entirely outside the control of GSM, and follows transmission rules are specified to the external network.

To reduce the number of cases dealt with by transmission equipment within GSM, despite the variety of interworking cases, two generic functions are inserted on each side of the GSM segment, as shown in figure 3.3.

6.4.1- INTERCONNECTION WITH THE PSTN

Data (i.e: binary digit flows) can be transmitted over the speech-oriented Public Switched Telephone Network (PSTN) via the use of audio modems, which transform the bit stream in an analog signal which is costrained to the 300-3400 Hz bandwidth carried by the PSTN. Most of the data services, including fax, videotex, are transported by the PSTN. A data connection between two PSTN users has the configuration shown in figure 3.4. case [a].

When considering a connection between a GSM user and a PSTN user it is clear the we must keep the half of the configuration concerning the PSTN user as if he/she was in connection with another PSTN user. An audio modem must then appear somewhere between the GSM user and the PSTN-GSM interworking point, as shown in figure 3.4 case [b]. The problem is then to design the GSM part of the transmission path, with the constrains imposed by the radio interface. For different reasons, chiefly because of the difficulty in designing an efficient and robust transmission scheme over radio for audio modem signals, the choice was to put the refuired modem on the infrastructure side, in fact in the interworking function. This means that, between the TAF and the IWF, the GSM network isonly refuired to transport digitally provided data.

Puttigs modems on the infrastructure side necessarily restricts the freedom of the user to use any kind of audio modems (as in the PSTN), since

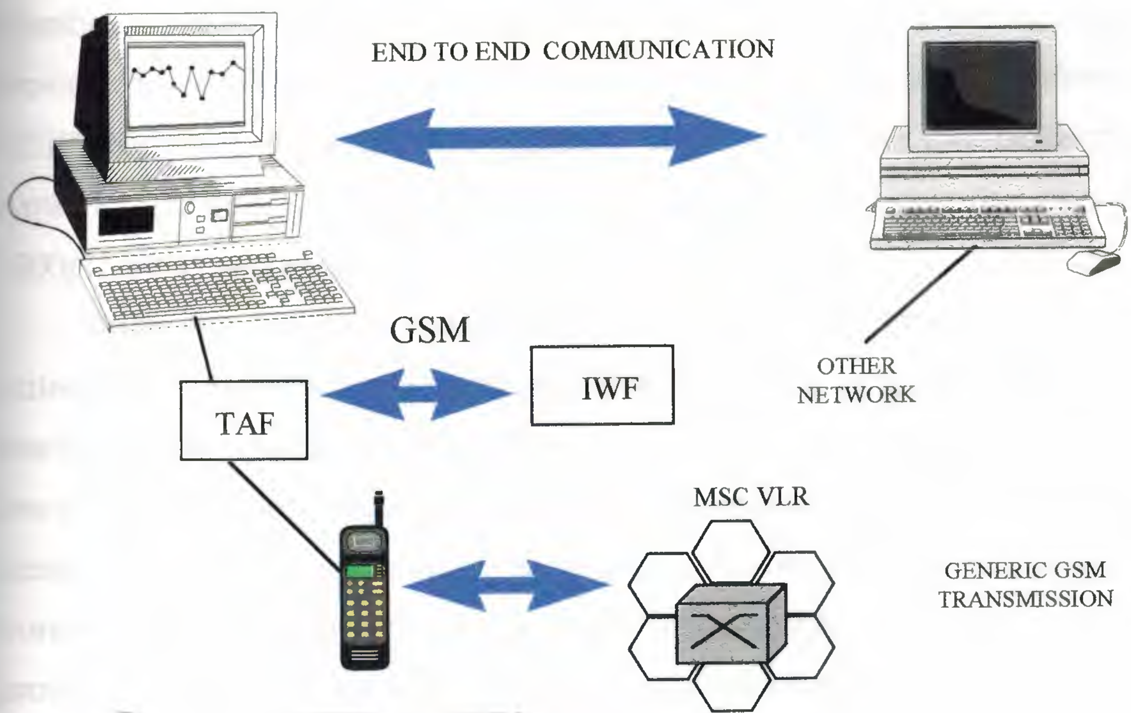


FIGURE 3.3 KOY

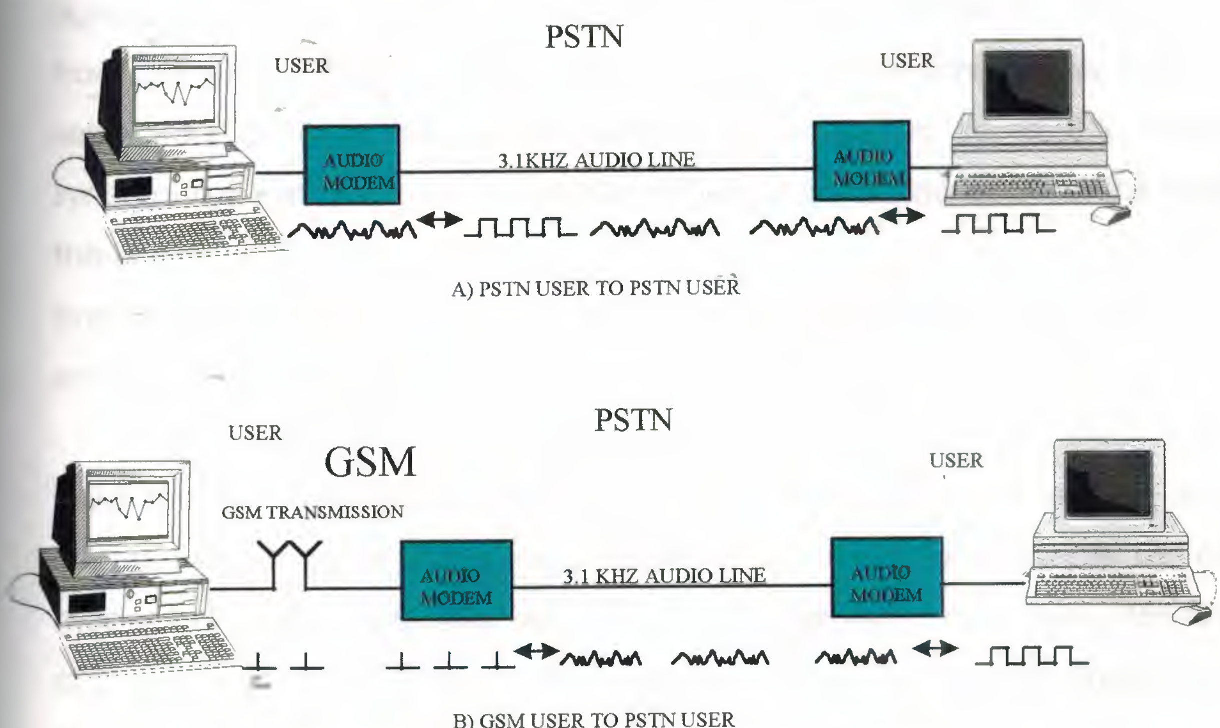


FIGURE 3.4 INTERCONNECTION WITH THE PSTN

they are constrained to use the types offered by the operator. However, in general, only standardised modems are used in the PSTN, and amoung all possible modulations, only a handful have survived the standardisation process. Moreover, the available rates are in a limited series (defined by CCITT Rec. X.1), in which the typically used values are the bidirectional symetric 300 bit/s, 1200 bit/s, 4800 bit/s, and 9600 bit/s, and the asymetric 1200/75 bit/s used in particular for the videotex service.

Between the two modems, the steram formatting falls into two categories, the saynchronous format. The difference of names comes from the fact that in the asynchronous format the instant of transmission of the bits are not aligned on a regular time base, whereas they are in the synchronous case. In fact, the difference is more profound, as the saynchronous transmission is a small protocol by itself, grouping bits into chorocters, and providin flow control for example. The unit of transmission is a character, a grouping of 7 to 9 bits. A character is preceded and followed by special signals, the start and the sto "bits". The data rate represents the rate of transmission of bits within a character, but successive characters may be separated by a period of any length, as shown in figure 3.5. In the synchronous case, bits are transmitted regularly and continously. In this case, the unit of

transmission is the bit, and it is up to end-to-end conventions to group bits in one way or another.

These aspects are important because the start/stop format, used by the asynchronous services are still, despite their ancestal character, the most widely used form of data transmission, being used for instance by the modern videotex, on the serial port of personal computers, for the connection of most computer terminals, or in the interface with smart cards (uncluding the SIM-ME interface in GSM).

A finite nad limited list of modem types, rates and transmission modes which can be supported for the interworking between GSM and the PSTN has then been established. It covers all the rates previously listed, with in each case, the modes allowed by other CCITT Recommendations. The list is given in table 3.1.

Modem type	Rate	Mode of Transmission
V.21	300 bit/s	asynchronous
V.22	1200 bit/s	asynchronous,
V.22bis	2400 bit/s	synchronous
V.23	1200/75 bit/s	synchronous
V.26ter	2400 bit/s	asynchronous
V.32	4800/9600 bit/s	synchronous
		synchronous

TABLE 3.1- Audio modem types supported by the Specifications

The most widely used audio modem tpes are supported by GSM, including both asynchronuus modems at rates up to 9600 bit/s.

The interconnection with the Integrated Services Digital Network (ISDN) is a must for a modern digital communisation system. Speech raises no problem, but this is not the case for data services. The basic data service supported by the ISDN uses the capacity of a bi-directional 64 kbit/s channel. By deliberate coice, GSM is not able to provide this service. Because it would use at least four times as much spectrum as a speech channel (and eight times in the future with the half rate speech coding scheme), a heavy price would have been charged to the user of this service. In these conditions, it is difficult to imagine it would have been much used. Moreover, it was clear from the start that the inclusion of a 64 kbit/s capability would have seriously impacted the design of the transmission system and increased the totol system cost.

Eventhough the services are similar in both networks, the interconneciton between the GSM and the ISDN, both using digital transmission, is radically different from the interconnection between either of them and the PSTN. Between GSM and ISDN there is no need for an audio modem. At the boundry point, the informations carried on a 64 kbit/s bit steam, with a portion of this rate corresponding to the bits exchanged between the end terminals, according to Recommendation V.110. The configuration of the connection, as presented in figure 3.7, case b, shows that each side is almost identical to the case where the other end is in the PSTN (the difference being the can be seen that the GSM transmission replaces the modems and one of the rate adaptation functions.

6.4.2- INTERCONNECTION WITH THE PSPDN :

GSM can offer the possibility to communicate between a GSM subscriber and a Packet Switched Public Data Network (PSPDN) subscriber, or more generally between a GSM subscriber and a subscriber that can be reached throug a PSPDN. This possibility can be provided by different means, depending on the terminal on the mobile station side, and on the level of intervention from the GSM infrastructure. To explain the different cases, it is simpler to start from the ways in which fixed subscribers can access a PSPDN.

There are three different means widely used to access the PSPDN, plus one of potential future utility when the user is an ISDN subscriber. The first is the direct access, where the subscriber is connected physically to the PSPDN. An example of an access interface, widely used, is the one specified by CCITT Recommendation X.25. The configuration (shown in figure 3.8. case a) usually includes a modem in the subscribers premises, but this modem is not an audio modem and it is part of the PSPDN. The terminal exchanges data with the network according to high-level packet protocol (X.25 levels 2 and 3). The subscribers is basically identified by the access line.

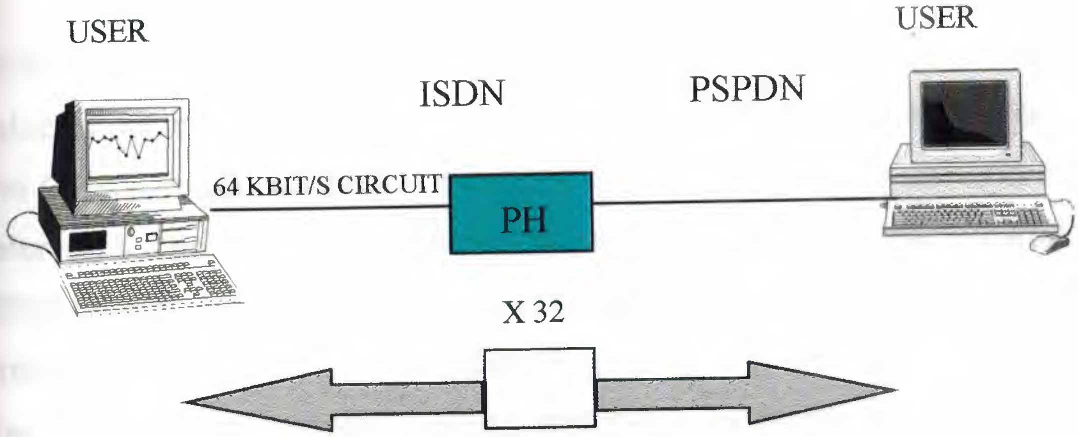


FIGURE 3.5

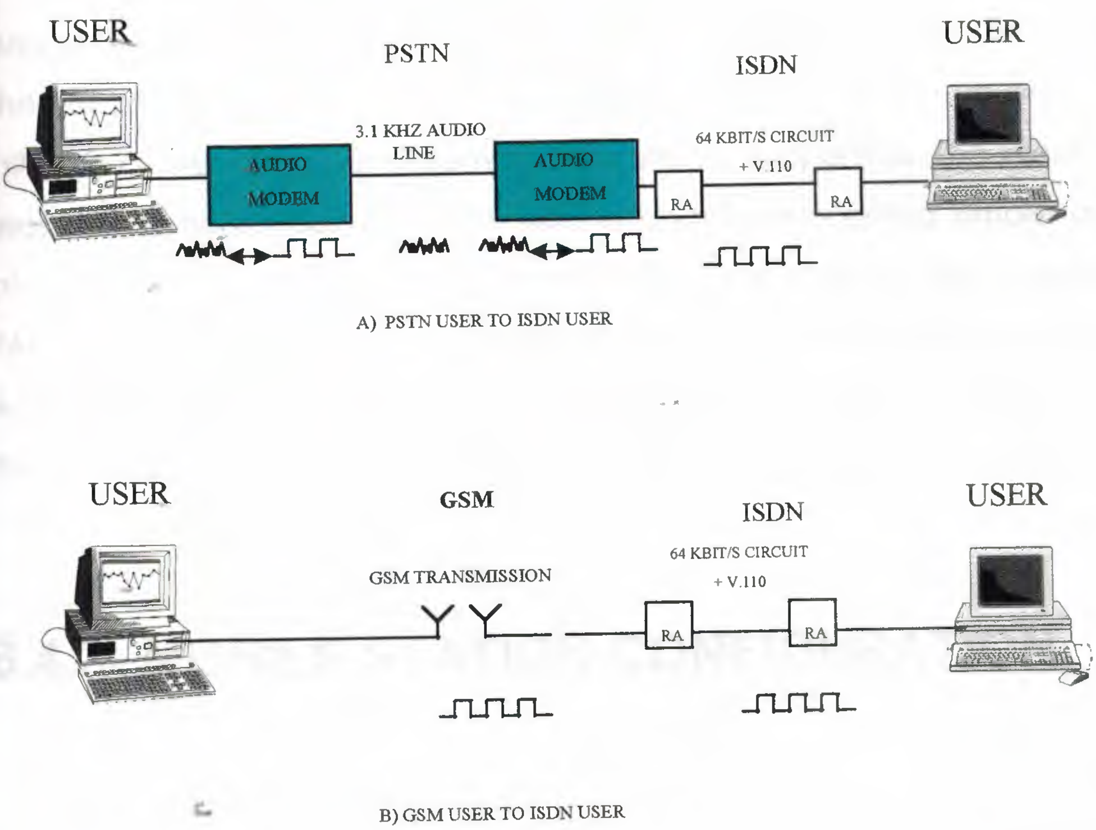


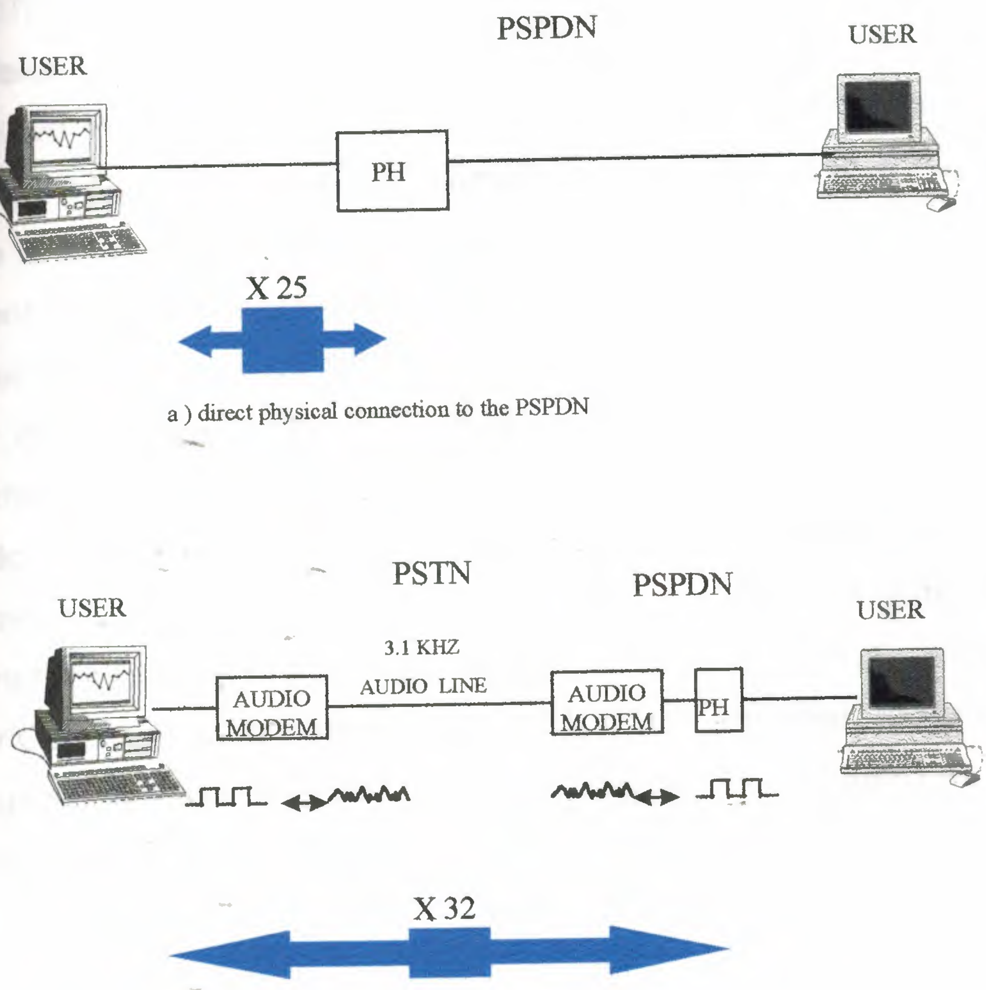
FIGURE 3.7 INTERCONNECTION WITH ISDN

The second method is a variation of te first, where the access is via the PSTN (figure 3.8, case b). Audio modems must be used. The connection is established first through the PSTN, as a normal PSTN communication. When the PSTN number addressing the PSPDN entry port. Then a second number, referring to the end correspondent, will be provided from the terminal to the PSPDN (this is the notion of double numbering). The major difference with the previous case is that is not possible to identify the subscriber by the access line. This is why the access protocol is slightly modified in this case to convey the subscriber identity. X.25 thus modified is the protocol specified by CCITT X.32. However the terminal and the PSPDN still use a packet protocol for communcation.

A third method makes use of a PAD function (a Packet Assembler Disassembler). It enables the user to have a simle terminal, which does not support a packet protocol. The major disadvantages of this solution are that the transmission data rate is limited, and that calls can be set up only at the initiative of the subscriber (incoming calls are not supported). The transmission uses an asynchronous protocol, with a character-oriented simple access ptorocol on top, which is used for numbering for example (an example of PAD access protocol is the one specified by CCITT Recommendations X.28). Access to a PAD can be direct, but the principle application is for the access through the PSTN. The configuration is then as presented in figure 3.9.

6.4.3- MOBILE STATION CONFIGURATION

Circuit Switched Publis Data Networks are, as their name indicates, telecommunication networks devuted to data, and use circuit transmission, like the ISDN or the PSTN, as opposed to packet transmission like in PSPDNs. The standardised user to network interface access for CSPDN follows CCITT X.21 or X.21 bis.



a) direct physical connection to the PSPDN

b) packet access to the PSPDN through the PSTN

FIGURE 3.8 PSPDN PACKET ACCESS

GSM provides the specifications for the support of X.21 and X.21 bis minals in the mobile station, as well as for interworking with CSPDNs. This is ne in a way which very similar to the previous cases, nothing that for PDNs only synchronous access is provided, with rates equal to 2400, 4800 9600 bit/s. ISDN specifications already provide for these functions, and M simply inherited them. The rate adaptation specification in this case is CITT X.30, which is very close to the synchronous case of V.110: the ferences are almost entirely in the additional information signals, which e not the same in CCITT X.21 and in CCITT V.24, but the structure of the me is exactly the same. The differences are only visible in the TAP and in e IWF, so the intervening machines are not involved. There are two ways entified in the specifications for accessing CSPDNs. Either GSM is directly terconnected, or a CSPDN is accessed through ISTN. There is no provision r access through the PSTN. In all cases the data flow at the interface etween the IWF and the BSS follows the ISDN specifications. No further daptation is needed as the connection goes on through the ISDN. In the ase of direct interconnection, the IWF perform sthe rate adaptation nctions for the translation between the ISDN format (X.30) and the format or CSPDN, which is defined by CCITT X.71. In both cases the interworking is igital, without specific modems. Figure 3.13 illustrates the two nterconnection configurations

3.5- TRANSMISSION INSIDE GSM:

The inner part of the GSM transmission system extends from a point ome where in the mobile station (inside the TAP for data services, and where speech is an acustic signal for the speech case) and the interworking

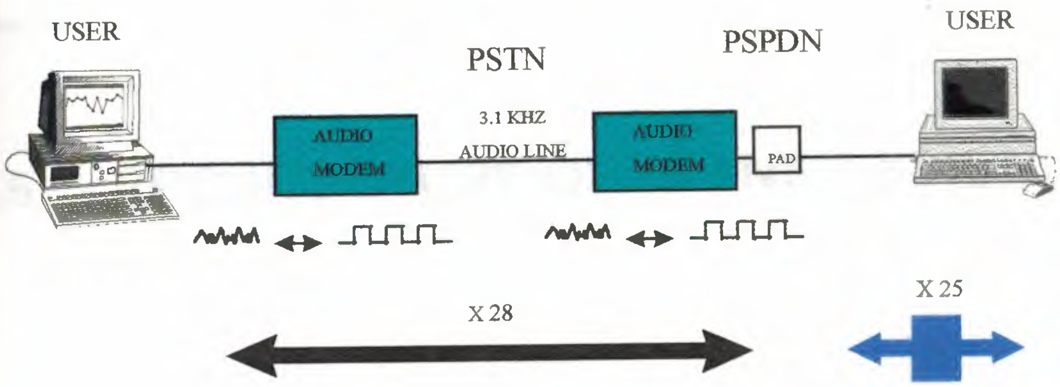
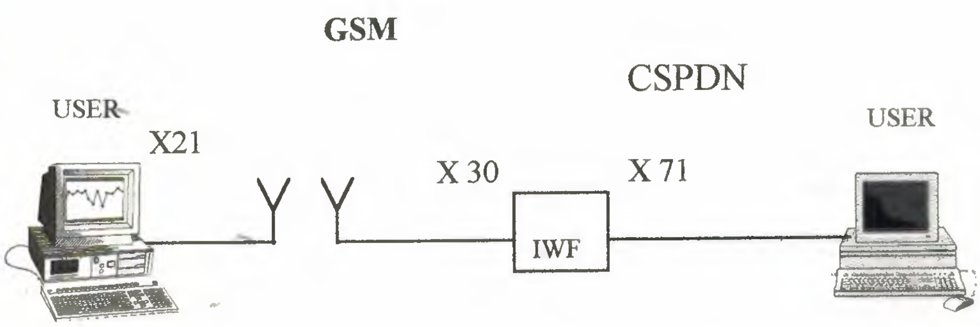
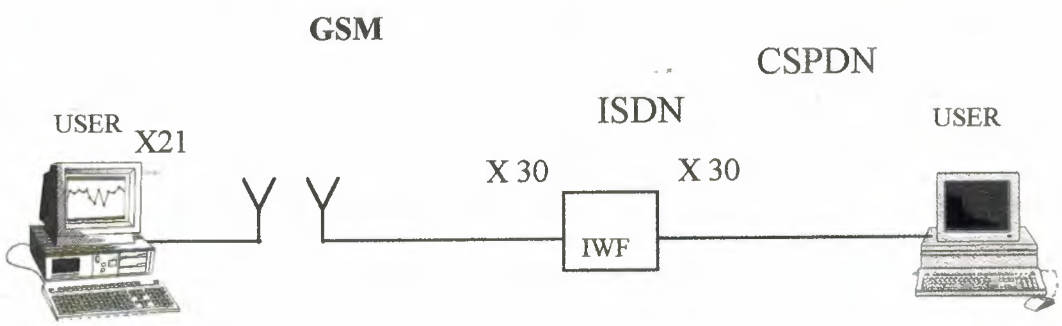


FIGURE 3.9 PAD ACCESS TO A PSPDN



A0 DIRECT ACCESS TO CSPDN FROM GSM



B) ACCESS TO PSPDN VIA ISDN FROM GSM

FIGURE 3.13 CSPDN INTERCONNECTION

ACCESS TO A CSPDN CAN BE PROVIDED DIRECTLY OR THROUGH THE ISDN.

point between GSM and external networks. Between this two points lie several machines and several interfaces.

6.5.1- ARCHITECTURE OF TRANSMISSION SYSTEM

The IWF is a set of fuctions fulfilling the adaptations necessary between GSM and external networks. As will be seen, it can be rather limited for speech toward the PSTN, and for basic data when interfacing with ISDN. But in other cases, such as facsimile. Interworking Functions can be quite extensive. The IWF as a function lies somewhere between the MSC and the external network it interfaces with. A first implementation approach is simply to put the IWF in the MSC, and this is the usual approach for simple cases such as speech. For the complex cases, it can also be imagined to have special machines devoted only to the interworking functions, and linked to several MSCs. This centralised approach is sensible if the traffic through the IWF is but a small proportion of the overall traffic. This inplamentation is not precluded by the Specifications, but there is no standard specification of an MCS/IWF interface, and any such interfaces will be pro prietary.

Another identified possibility is when the external interface of the Mobile Termination (MT) is the ISDN "S" interface, to which off-the-shelf ISDN terminal equipment can be directly connected. In thes case, the machine is called MT1 (Mobile Termination type1). A terminal using a modem to terminal interface can still be connected to an MT1, provided an ISDN Terminal Adaptor (TA) is inserted. In this case, the Terminal Adaptations Functions (TAF) are spread between the MT1 (where a synchronous adaptation is performed) and the TA (where for instance the synchronous/asynchronous adaptation is performed). The different mobile station configurations are illustrated in figure 3.14.

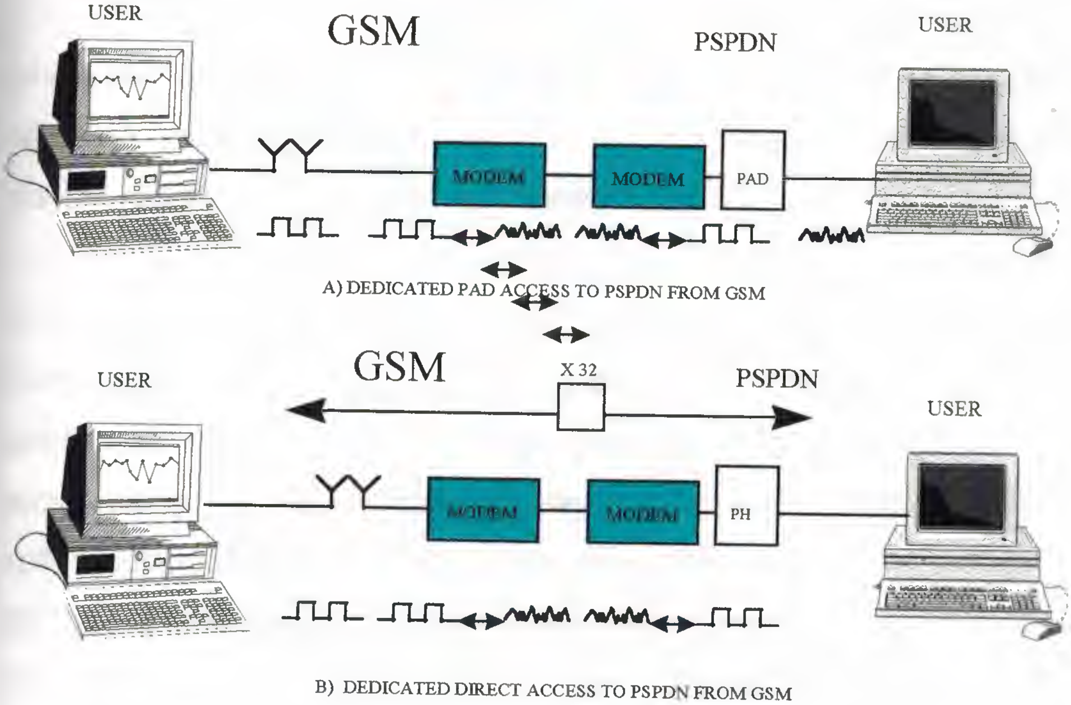


FIGURE 3.14.

POSITION OF THE TRAU

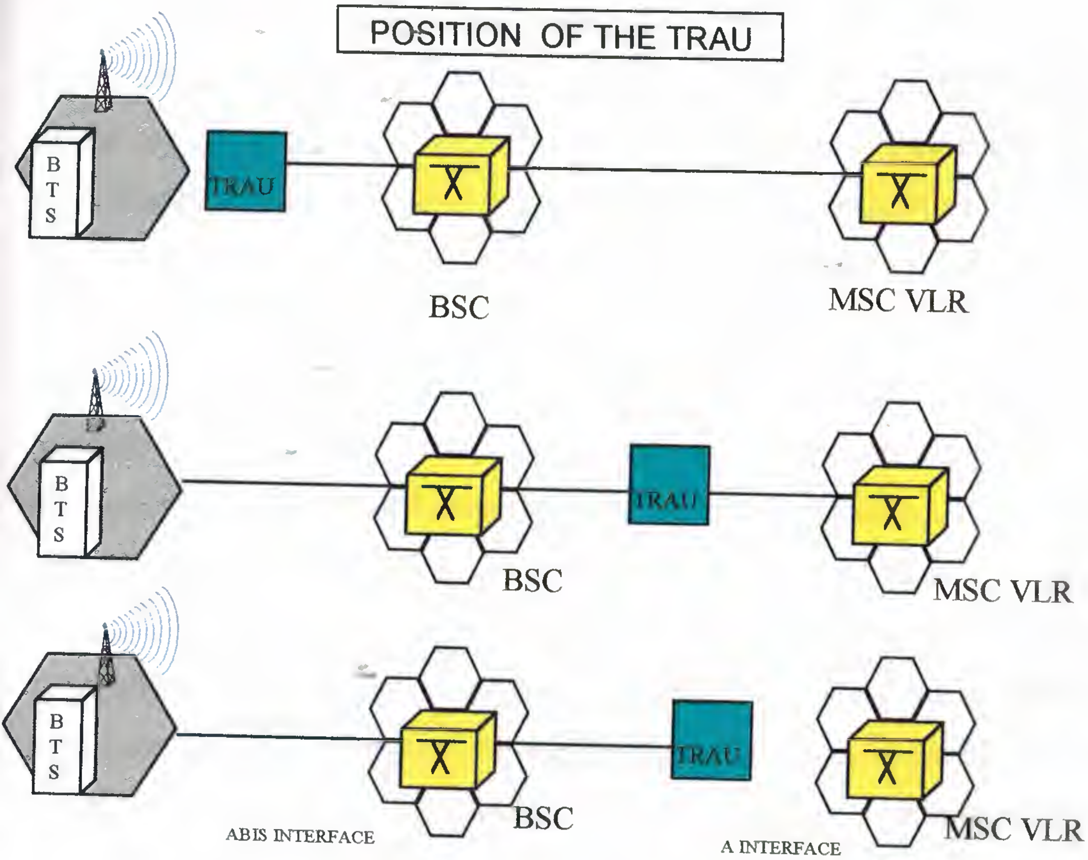


FIGURE 3.15

Which will be our basis of work in the following for data applications,is when the TAP is totally integrated with the generic functions, and interface with the terminal through a classic modem to interface. This integrated device is called MT2 (Mobile Termination type 2).

Along the transmission path, the canonical architecture of GSM distinguishes the BTS (Base Transceiver Station), the BSC (Base Station Controller) and the MSC. Between the mobile station and the BTS is a clear references point, the radio interface, where the information crosses the space riding the 900 or 1800 MHz electromagnetic waves. The BTS/BSC/MSC split is adequate for the study of the signalling aspects. But the MSC and the BSC have little role to play in the transmission chain. Historically, the BTS and the IWF were the main actors in the transmission scene, and only basic transmission functions were found between them. Then another piece of equipment was introduced; the TRAU (Transcoder/Rate Adaptor Unit), which is definetely transmission equipment, and which was conceived to be distinct from the BSC or the MSC. The TRAU will take the starring role in this section.

The rationale behind the existence of TRAU, distinct from the MSC and BSC, consist of several points The implicit assumption during the elaboration of the concept of MSC aws that it would be implemented more or less as a modified ISDN switch. As a consequence, the transmission at the level of the MSC is very close to that of the ISDN specifications. In particular only 64 kbit/scircuits are switch. As a consequence, the A interface must conform to the lower layers of the ISDN specifications. Indeed, the 2 Mbit/s standard multiplex structure used on the A interface (and also on the Abis interface) is not specific to GSM, but follows the CCITT G. 703 standard. Their basic usage is to carry 64 kbit/s circuits compliant with the needs of ISDN. The multiplexing is based on a 125 us cycle, each cycle transporting one octet per circuit. This structure is aimed at the transport and switching of 64 kbit/s circuits, but, in addition, enables transport of sub-multiple rates such as 32,16 or even 8 kbit/s. This possibility is effectiively of interest for GSM, which does not require connections of more than 16 kbit/s, and where the cost of internal terrestrial links (between BTS and BSC, and between BSC and MSC), usually leased by the operator, represents a ubstantial part of the operational cost. A transmission method using only 16 kbit/s for user data (signalling is kept on 64 kbit/s links) was then devised, to allow this cost reduction which seems compelling despite some drawbacks. First, this introduces some extra delay for the transmission, and hence lowers the overall speech transmission quality. Second, it introduces a gateway function at the border between 16 and 64 kbit/s, which is really the purpose of the TRAU.

The late introduction of the TRAU, and the will to keep th switching capability of the MSC strictly to the one of an ISDN switch, is the source of its eccentric architectural location. The TRAU may be located in different places along the transmission chain, between the BTS, to which it belongs functionality, and (but not including) the MSC. One may then deduce that the only site possible when not on the BTS site is the BSC site. This is however not quite so; the implementations of many manufacturers include a remote transcoder situated on the MSC site. The BSC, as functional unit, -is then "spread" over its own site and the MSC site, and includes the link between these two sites. Conversely, the BSC-MSC interface (or A interface) is situated on the MSC site, over a very short distance. Figure 3.15 shows the positions of the TRAU relatively to the other BSS machines.

As a consequence, the Specifications strictly speaking do not allow the placing of the transcoder inside the MSC. Every call between two GSM users must then undergo two transformations (from 16 kbit/s to 64 kbit/s and vice versa, entailing for speech two transcoding operations between the 13 kbit/s and the 64 kbit/s representation), even if the two users are connected to the same BTS.

Because the TRAU is the true intermediate equipment for transmission, and because of its architectural predicament, we will not use the notion of A and Abis interfaces in this section, but instead the BTS/TRAU and the TRAU/IWF (or TRAU/MSC) interfaces. The BSC (and the MSC in the case of tada) is simply ignored, as it has no special role as transmission equipment (but some as a switching equipment). The relevant interfaces are in each case the following:

* the radio interface;

* the BTS / TRAU interface (which can be non-existent if the transcoders are situated at the BTS);

* and the TRAU - IWF interface, or more generally the interface between the transcoder and the point of interconnect with other networks.

6.5.2- DIGITAL SPEECH TRANSMISSION

Digital speech transmission over a radio interface in a mobile environment is quite a challange. As already mentioned, a special digital speech coding algorithm is used in GSM, cosen for its low bit rate (13 kbit/s) and its resistance to high error rate conditions. Some emphasis will be given to some side features of voice transmission, such as vocal activity detection and discontinous transmission, which are important for the spectral efficiency of GSM. The rest of the section is devoted to the rate adaptation which enables speech encoded with this algorithm to be carried not only over the radio interface, for which it was originally designed, but also over fixed digital links, between the BTS and the TRAU.

The GSM transmission path for can be divided into the following segments:

a- the mobile station;

b- from the mobile station to the base station: the radio path;

c- from the base station to the (remote) voice transcoder;

d- from the voice transcoder to the MSC:

The junction points separating a to d described above correspond to places where a speech representation is changed to another one. This transcoding points are of major importance here, since the description of a transmission scheme is intimately related to the description of the corresponding transcoding functions. The following transcoding points are identified inside the GSM domain:

* Accoustic to Analogue Electric transcodind, implemented in the microphone, and the reverse Analogue Electric to Accoustic transcoding, implemented in the loudspeaker; this type of transcoding is not however specific to GSM;

* Analogue to 13 kbit/s Digital transcoding (and the reverse opeartion), implemented in the mobile station.

* 13 kbit/s Digital to 64 kbit/s Digital transcoding (and the reverse operation), implemented in the voice transcoder, either in the BTS or in the TRAU.

This does not mean that the signal is transported exactly in the same way on all links between two transcoding points. The signal representation is adapted to the transmission medium in intermediate processing points. The two main adaptations are:

* adaptation of the 13 kbit/s digital representation for transmission on the radio path;

* adaptation of the same 13 kbit/s digital representation to transmission on fixed links between the BTS and the voice transcoder in the TRAU.

6.5.3- SPEECH ON THE RADIO INTERFACE:

The prime concern for the design of the speech transmission means on the radio path aws spectrum efficiency. The goal was to use as low a data rate as possible while providing an acceptable level of quality. Since speech is considered is the prime service, these considerations have heavily influenced the whole design of the system. On the radio interface, two types of raw carrying capabilities are defined, the "full rate! channel (which deserve this title just because it was the first to be specified), and the "half rate" channel, which indeed makes use of half as much radio resource as the previous one. The existence of these two types comes from history; at the time of definition, it was certain that it would be easy and quick to specify a digital speech coding at around 16 kbit/s with the required quality, and it was foreseen that it would be possible to do the same thing with half as many bits some years later. Because of the stress on spectrum efficieny, it was out of the question to forget this halving possibility. It was therefore decided to define the system in two steps, starting with a less efficient coding scheme, but paving the way for a two fold increase in efficiency to be introduced as soon as possible.

In the first phase of GSM, speech is then only defined for the full rate channel. The description of the coding scheme for this type of channel is presented in the following pages. As the end of 1991, commercial use of half rate speech is foreseen for 1994 or 1995.

6.5.4- THE SPEECH CODING & DECODING ALGORITHM ;

The GSM speech coding scheme at 13 kbit/s is called RPE-LTP, which stands for Regular Pulse Excitation-Long Term Prediction. It aims at producing a speech quality similar-when no errors are added-to the fixed telephony network quality, with a much smaller rate in order to optimise the use of the radio spectrum.

6.5.5- GENERAL PRINCIPLES;

The aim of this section is to give the general principles of the GSM speech coding scheme. The bit-exact algorithms for coding and decoding are given is TS GSM 06.10; therefore a specialist shall find no better description than the one in the Specifications. However, the nonspecialist will find here a general idea of the GSM speech represantation.

The easiest way to dive into this subject is to look first at the contents of the transmitted signal, and its translation from 13 kbit/s to 64 kbit/s, which is performed on the decoder's side.

Speech is cut into 20 ms slices; in fact, rather than saying that the transmission rate is 13 kbit/s, it would be more realistic to say that speech is transmitted using groups of 260 bits every 20 ms. Synchronisation (i.e: seperation of the 260 bits blocks. The radio interface makes extensive use of its complex synchronisation scheme; this is why the 13 kbit/s flow, structured in frames of 20 ms, does not include any information helping the receiving entity to determine the frame boundries.

For each block, the output signal is reconstructed by the receiver from an input signal (the excitation signal) which is filtered through a succession of filters (i.e: of linear transform) as shown in figure 3.16.

The Long Term Prediction-or LTP-filter is a very simple filter, which consist in adding the signal and its delayed image multiplied by a factor br, the delay being Nr samples. The values of both Nr and br are transmitted in the speech frame, once for every 5 ms slot.

The Linear Prediction Coding-or LPC - filter is the inverse of an 8^{th} order linear filter. A linear filter of nth order performs a linear combination of the signal and of itself delayed by 1, 2, ..., n samplesat 8 kHz. The coefficients

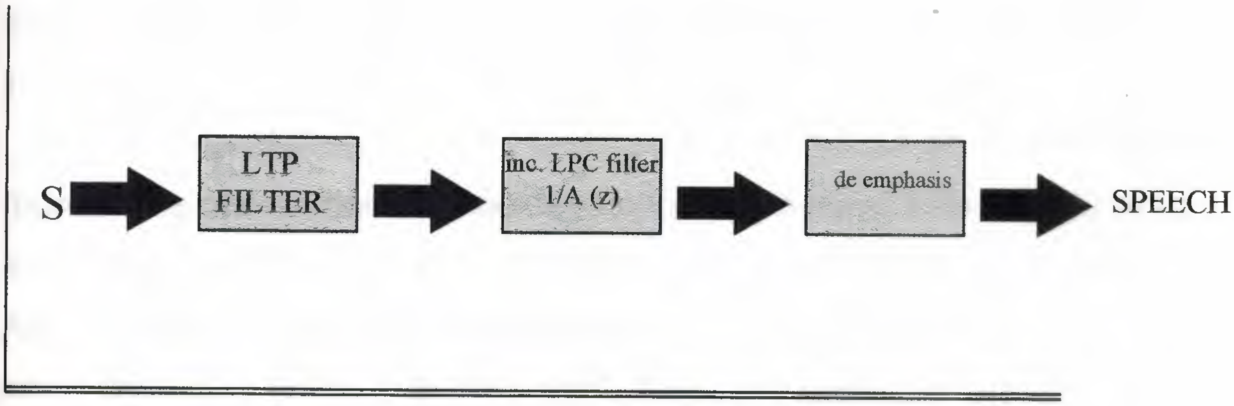


FIGURE 3.16 MODELLING OF THE 13 KBIT/S SPEECH SIGNAL

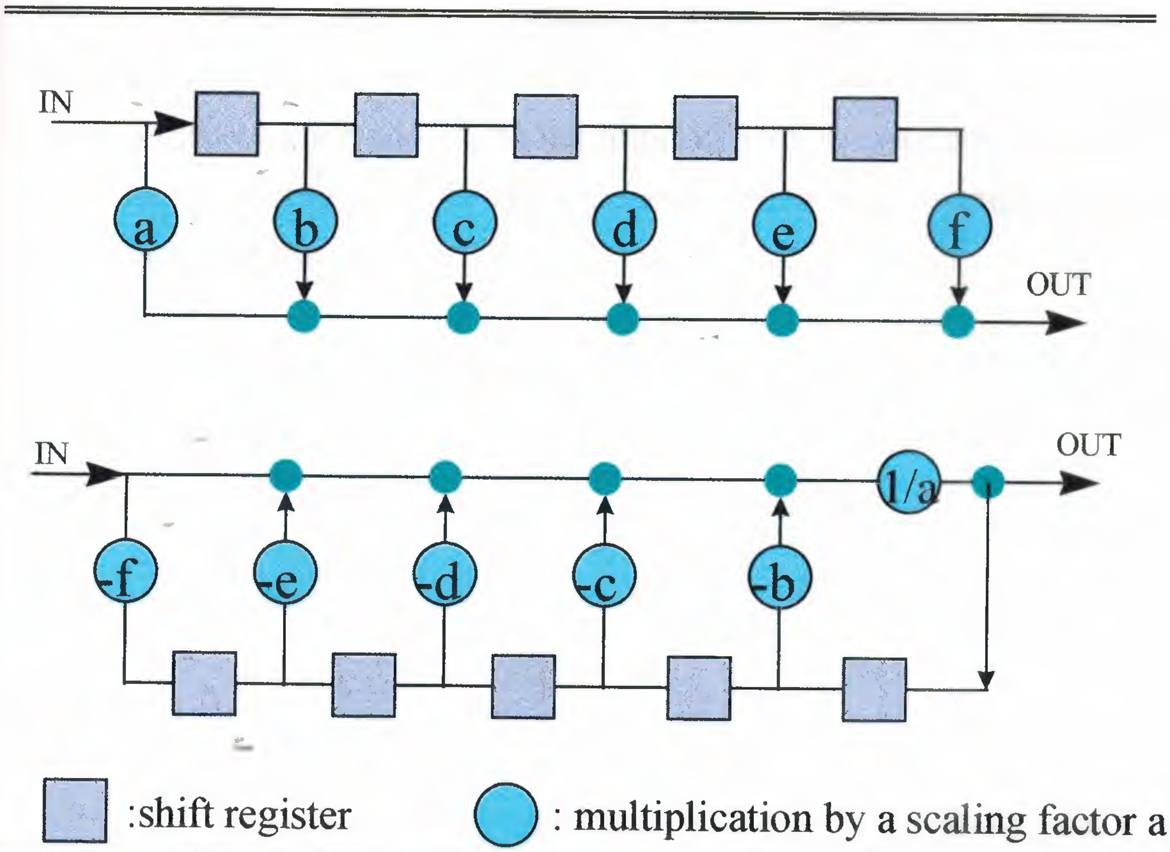


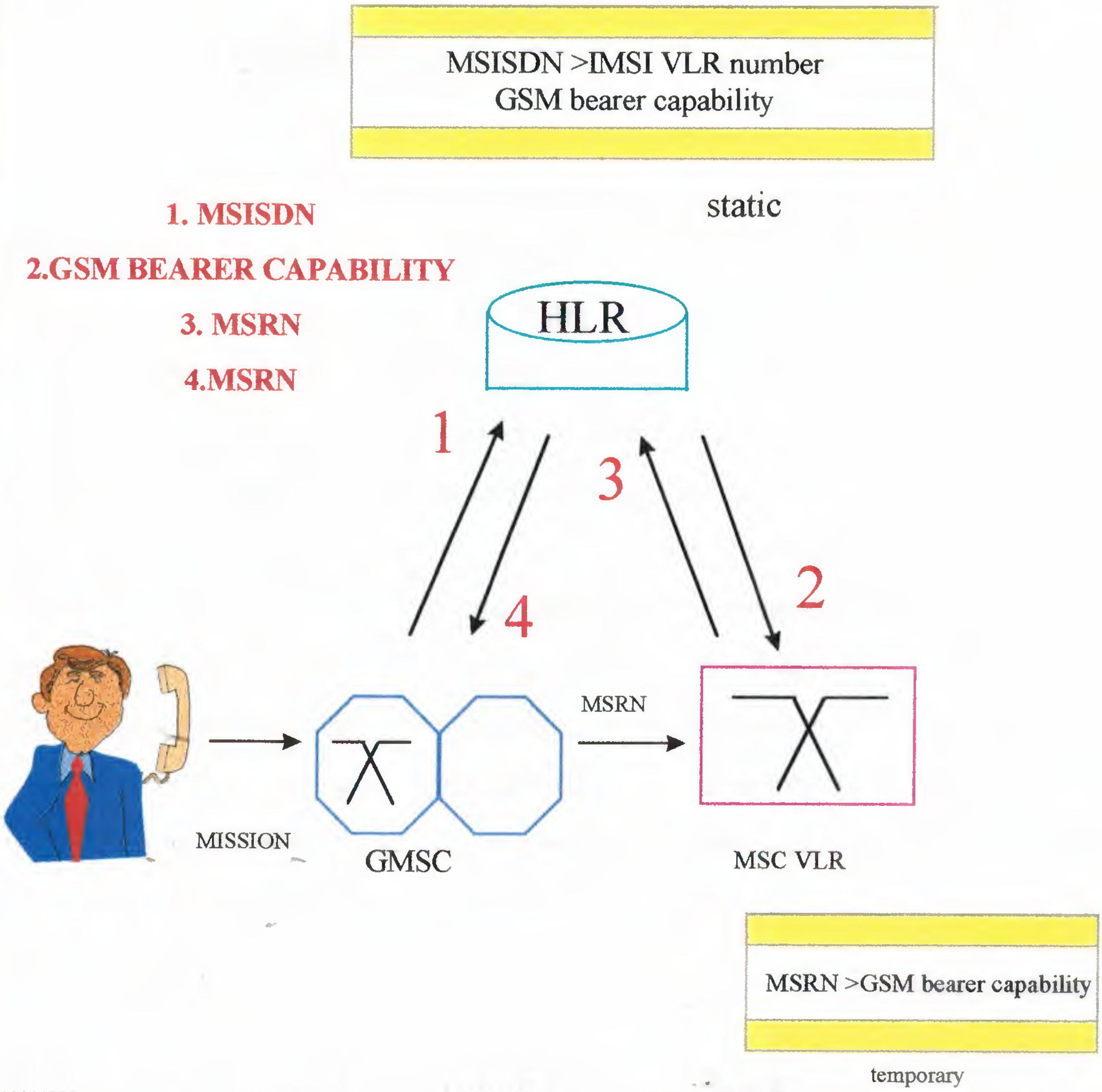
FIGURE 3.17 A LINEAR FILTER AND THE CORRESPONDING INVERSE FILTER

of this filter vary from one block to another, and are transmitted in the speech frame. The structure of linear filter and the inverse filter can be found in figure 3.17.

The excitation signal S itself is coded so that the set of all pramaters - those of the above filters plus the description of the signal S-fits into 260 bits. S is sampled regularly ("RPE, Regular Pulse Excitation") at a rate of anly 8/3 kHz. According to the signal processing theory, this allows to know accurately the information concerning only the lower 1.3 kHz bandwidth of S. The excitation signal at the input of the filters is reconstructed by inserting null value samples, so as to obtain a signal sampled at 8 kHz. From a spectral point of view, this result in a signal with spectral components above 1.3. kHz which are drived (second and third harmonics) from those below. This phase of the 8/3 kHz samples with regard to the 8 kHz samples can vary, and is transmitted once for every 5 ms slot.

The samples are coded using Adaptive Pulse Code Modulation (APCM). This coding is called "adaptive" because the maximum amplitude and the ratio of each sample to this maximum value are coded separately. This differs from the usual 64 kbit/s coding, where each sample is directly coded using a fixed scale.

Table 3.2. summarises all the parameters transmitted in each 260 bits frame (i.e: every 20 ms).



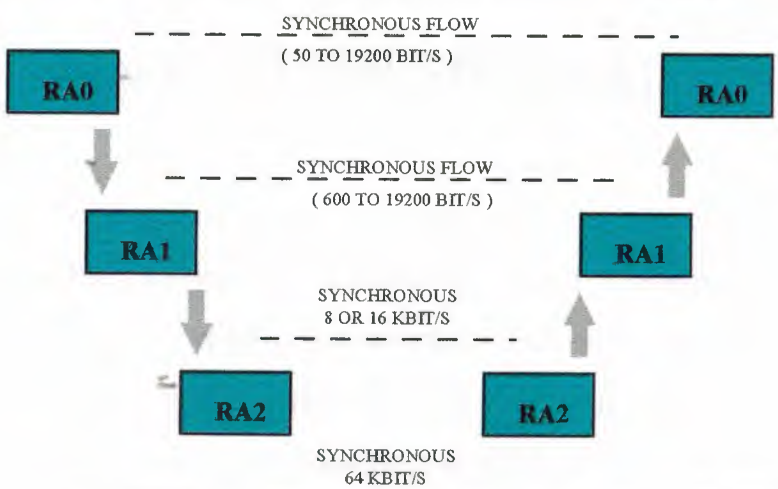


FIGURE 3.20 RATE ADAPTATION IN ISDN ;THE THREE STEPS

6.5.6- NETWORK INDEPENDENT CLOCKING:

The exact transmission rate through digital networks such as ISDN is imposed by a network clock. Now, the exact end-to-end rate may be different when one end is in the PSTN. An audio modem used with a PSTN line may synchronise its transmission on its own clock. In this case, the frequency tolerance is 100 ppm, a very high value corresponding to one bit in excess or in default each second at 9600 bit/s.

To cope with such cases, V.110 includes mechanism whicle enable a rate adaptation unit to indicate frequency corrections to the other. This mechanism also allow an indication of when a bit has to be skipped, or on the contrary to be added and in this case the bit value.

These mechanisms make use of a set of 8 commands, such as " no change" , accelerate your clock toward the terminal by 20% "," skip one bit "," "add a bit of value 1 " , and so on.

6.5.7- THE ISDN GENERAL RATE ADAPTATION SCHEME:

When considering the end-to-end synchronous case, or when looking at the portion of the transmission path between two RAO functions in the case of the end-to-end asynchronous case, the data flow consist in a bi-directional synchronous flow at the naminal rate, accompained by some auxiliary information representing a rate of a few kbit/s. Rate adaptation proceeds in two steps, called respectively RA1 and RA2. The whole process is shown in figure 3.20.

The RA1 functions provides a bit flow at the intermediate rate of 8 kbit/s or 16 kbit/s, according to the nominal rate to transport. RA1 is not only a rate adaptation function. It also includes the multiplexing and demultiplexing between the auxiliary information (modem control plus other signals) and the main flow, as shown in figure 3.2.1. This is done by time multiplexing, that is to say that the multiplexed bit flow is a regular alternation of bits of the main flow and of auxiliary information bits. This requires synchronisation between the multiplexer and the demultiplexer which is maintained by transporting additional bits. The period of recurrence for the multiplex structure is either 5 ms (for the 16 kbit/s intermediate rate) or 10 ms (for the 8 kbit/s intermediate rate), and defines successive multiplex frames. By the way, these values of 5 and 10 ms are one of the reason for the choice of 20 ms as the frame period for speech in GSM.

6.6- THE GSM T CONNECTIONS:

The transmission path between the TAF on the mobile user side and the IWF is functionally totaly equivalent to what appears between the terminal to " modem " interface and the 64 kbit/s circuit in the case of an ISDN connection using V. 110. So the RA0, RA1, RA2 functions will appear somewhere between the TAF (included) and the IWF (excluded). However the transmission over the radio interface must be introduced somewhere in the picture.

Data transmission on the radio interface is not done at 64 kbit/s, and V. 110 obviously cannot be used in its pure from. A first idea could be to keep V.110 as it is with the exception of the RA2 function, which is very simple, and has clearly to do only with transportation over 64 kbit/s circuits. Between RA1 functions, the transmission is done at an intermediate rate, 16 kbit/s or 8 kbit/s, which could have been fitted onto the transmission over

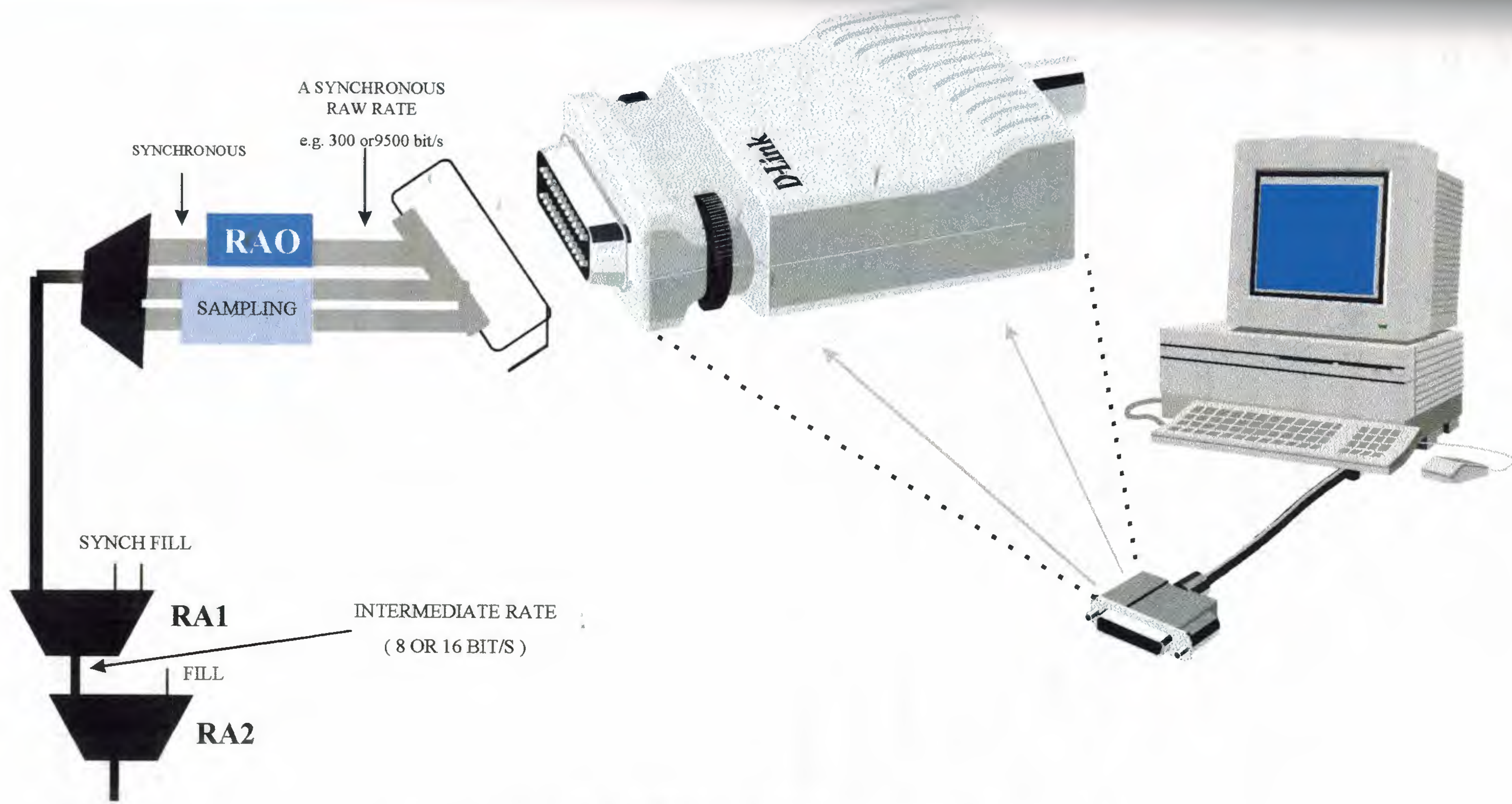


FIGURE 3.21 RATE ADAPTATION IN ISDN : RA1

the radio interface. Yet, the problem on the radio interface is to limit as much as possible the information to be transmitted, so that the maximum part of the raw through output can be devoted to optimised redundancy, in order to maximise the transmission quality.

When the V.110 bit stream at the intermediate rate is looked at, it becomes apperant that an important part of the exchanged bits can be removed in GSM. The first of these are the synchronisation bits. Out of the 80 bits of a V.110 frame, 17 are used for synchronisation. GSM radio transmission is based on a complex synchronisation scheme, and there is no difficulty in deriving the V.110 frame boundaries from the GSM synchronisation (thanks in particular to the choice of 20 ms as a fundamental GSM synchronisation period, which is a multiple of 5 and 10 ms). In fact, another important aspect of synchronisation comes from the forward error correcting scheme used over the radio interface. With such schemes, residual errors are grouped into bursts, corresponding to an ill-fated radio coding block. This is possible because the coding block recurrence has been chosen to be 20 ms. The rule is then simple; a radio coding block corrsponds exactly to 2 or 4 V. 110 frames.

But 63 bits still remain per V.110 frame. Out of those, three are not transmitted over the radio interface, because they can be reconstructed by the receiver. These are bits E1, E2 and E3 which indicate the true end-to-end data rate. This is not correspond to new information between the mobile station and the infrastructure, since the rate is transmitted separately by signalling means for setting-up purposes, and thus can be dispensed with. What remains consists finally of 60 bit frames, which can be seen as a simple subset of the original V.110 frame.

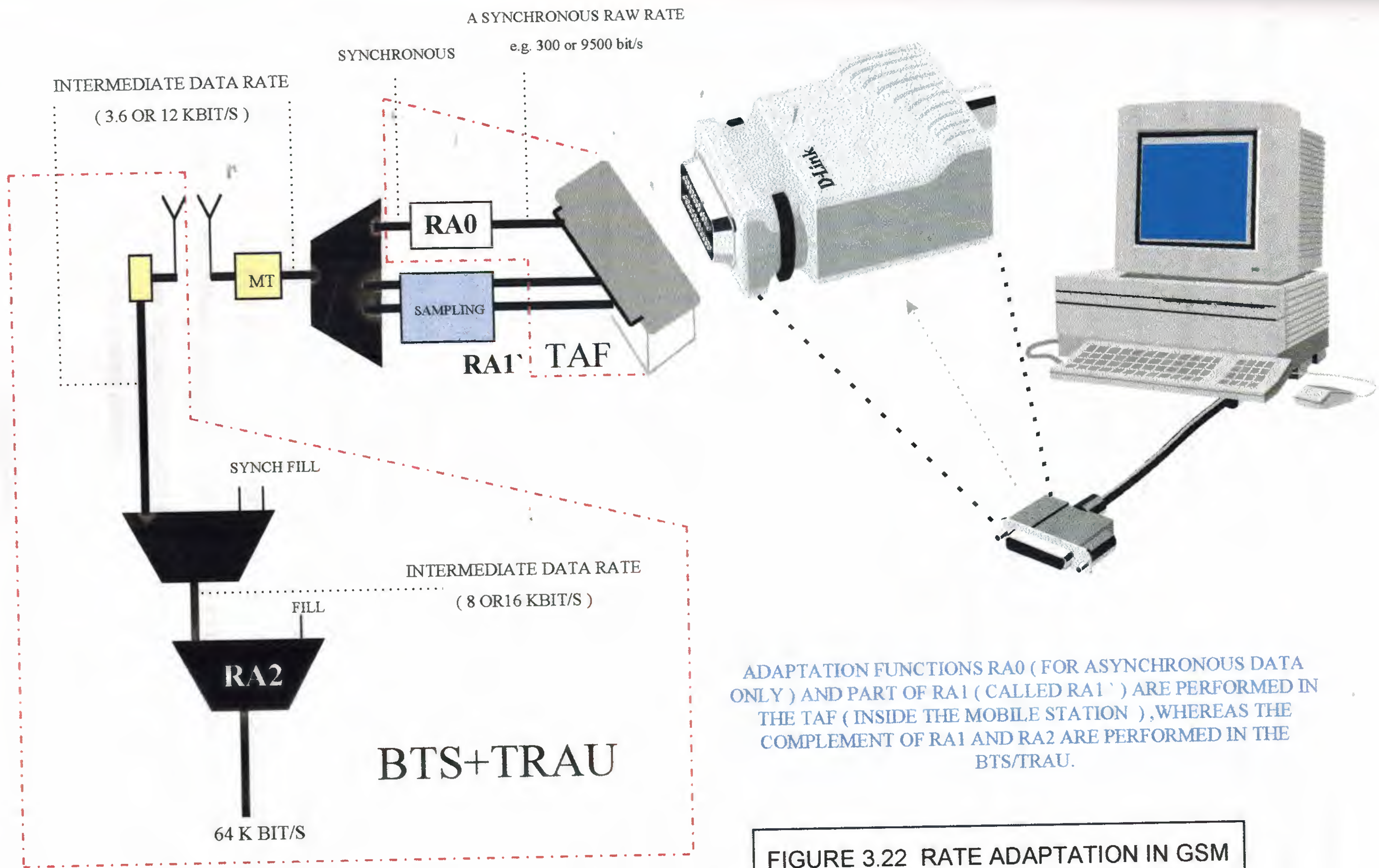
The resulting " intermediate rate" for GSM is then 12 kbit/s (derived from the 16 kbit/s) or 6 kbit/s (derived from the 8 kbit/s). In fact, a third and lower rate has been introduced for user bit rates below 2400 bit/s, once again to optimise the redundancy. We have seen that in ISDN, rates below 480 bit/s are rate-adapted to 4800 bit/s by simple bit repetition. In GSM, this simple repetition is done only up to 2400 bit/s. Because the same amount of auxiliary information is kept, the intermediate rate corresponding to ser data rates of 2400 bit/s or less is then 3.6 kbit/s (2.4 + 1.2) The "V.110 - like" frame in this case is not 60 bits long any more, but 36 bits long. The transformation from ISDN frames at 4800 bit/s to GSM frames is done simply by taking every other user bit. The reverse transformation consists in duplicating each user bit.

Figure 3.22 shows how the radio interface is introduced in the rate adaptaion chain, to be compared to the ISDN case of figure 3.21. The RA0 function is performed on the mobile side, as well as the rate adaptation inspired by the ISDN RA1, called RA1'. This includes in the synchronous cases the network independent clocking control as defined in V.110.

On the infrastructure side the RA1'/RA1 function perform the translation between the radio interface format and the ISDN format, and an RA2 function completes the ISDN adaptation, so that the data flow reaching the IWF is an a full ISDN format. The difference between a V.110 frame and a radio rate adaptation frame is simple, and the translation between the two is easy. It is just a matter of adding (respectively moving) the synchronisation bits, synchronising the V. 110 frames with coding blocks; and adding (respecti-vely removing) bits E1, E2, E3, whose contents are known thanks to signalling inside the GSM infrastructure.

6.6.1- THE GSM NT CONNECTIONS:

Since in any case the IWF has a lot to do for an NT connection, there no reason why GSM NT connections need to strictly follow ISDN specifications as in the T case. The needs are basically to transport a flow of 240 bit frames between the TAF and the IWF, using a maximum total rate of



ADAPTATION FUNCTIONS RA0 (FOR ASYNCHRONOUS DATA ONLY) AND PART OF RA1 (CALLED RA1 `) ARE PERFORMED IN THE TAF (INSIDE THE MOBILE STATION) ,WHEREAS THE COMPLEMENT OF RA1 AND RA2 ARE PERFORMED IN THE BTS/TRAU.

FIGURE 3.22 RATE ADAPTATION IN GSM

12 kbit/s. The adaptation to ISDN is done, if need be, at the ends of the connection (TAF and IWF).

However there are some advantages in using as close as possible transmission methods for different modes, and NT transmission has been designed to have a common core with the 9.6 kbit/s T connections. Indeed, the data rate effectively carried between the TAF and the IWF for 9600 bit/s T connections is also 12 kbit/s, as explanied above. In addition, the distinction between user data and auxiliary information is irrelevant for the RA1' function or for the transmission over the radio interface.

A simple solution for the transmission of the auxiliary information on NT connections would have been to do the same as in the T cases. There would have been no major obstacle to this choice, and the result would have been elegant. A difference choice was made. For T connections auxiliary information adds 12 bits for each period of 5 or 10 ms. This would have resulted in respectively 48 or 24 bits per RLP frame. This aws felt to be a high overhead, of which at least a third is useless (bits E4 to E7), since network independent clocking signals are not used in NT connections. Moreover the rest of the bits, which correspond to side signals to and from the terminal, rarely vary. A more complex approach has been chosen to reduce the load incurred by the auxiliary information in most cases. The key fact is that at most, three side signals from the terminal are sampled in each direction (table 3.6). The idea was to transmit the values of these signals once per RLF frame, plus to indicate the transitions, if any, during the period corresponding to the user bits in the frame. The formatting is such that a minimum of 8 bits is consumed pet frame, plus a further 8 bits to indicate a transition. So, if the signals are stable, which is the case when the connection is operational, only one octet per RLP frame is used for auxiliary information instead of 6. If the side signals are often changing, the auxiliary information may use much more than 6 actets per frame, but such cases cannot occur when effective transmission of user data takes place.

Another important point for NT connections is the need for frame delimitation. As usual, frame delimitation can be easily obtained on the radio path as a side product of the comprehensive synchronisation arrangement. Then, despite the difference of frame length (RLP frames are four times as long as V.110 frames for the 9.6 kbit/s T connection), it was possible to use exactly the same transmission scheme for the T and NT cases over the radio interface, and so it was done.

Unfortunately the radio path is not the only segment in the way, and the RLP frame delimination must also be transported between the BTS and the IWF. The V.110 frame delimination is available, but it is not sufficient, the RLP frames being larger. And there like a devilout of its box we find the three E bits; they are free for any use in the case of an NT connection. They are used to convey the frame delimitation.

For T connections, the three E bits are dealt with in the RA1'/RA1 translation. NT connections use a slightly modified RA1'/RA1 translation, which manages the correspondance between the RLP frame delimination over the radio interface and one the given by the V.110 frame delimination and the E bits. In the NT case, these bits are used the indicate the position of a. V. 110 frame in a group of four constituing ar RLP block. In the downlink direction, the RA1'RA1 uses this information to put the four frames of same RLP block in a same radio interface block. The size of the RLP block has been chosen according to the radio block size, such that errors affecting one coding block affect only one RLP frame. Conversely, the E bits are set in the uplink direction according to the position of the contents of the V.110 frame in the corresponding radio block.

The next point to look at is the protocol conversion. It has already been mentioned that the RLP replaces the start/stop protocol or the packet protocol used by the terminal. The RLP between the TAF and the IWF provides the same functionality as the original protocol, but adapted to the GSM transmission. The conversion is done by relay functions in the TAF and the IWF. These functions depend of course on the terminal protocols. There are three cases; the start/stop protocol, LAPB (X.25-2) and the protocol used for fax. The fax protocol is in essence the same as X. 25-2 but with additional signalling so that transmission is basically identical and the three cases are effectively only two. The Specifications distinguish two relay protocols, namely the L2R-COP (Layer 2 Relay Character Oriented Protocol) and the L2R - BOP (Layer 2 relay Bit Oriented Protocol).

In the asynchronouc case, all the functionalities of the start/stop protocol are fulfiilled by the NT mode functions. The frame synchronisation enables bits inside the frame to be constructed into octets, thus removing the need for start and stop signals. Only the 7 or 8 bits of user information are transported (1 fiil bit is added in the case of 7-bits characters, so that in all cases there are 8 bits per character in the frame). At the receiver's end, the start and stop signals can be reinserted at their correct position. Note that the duration of the stop signal is lost, and as a consequence the relative timing of the characters is lost, but their order is kept. This just makes asynchronous transmission even more asynchronous, and no application is known which suffers from this. Flow control is fulfilled in the start/stop protocol either by using special characters, or by toggling modem control signals. The RLP provides its own flow control, in particular because it needs to regulate the flow when for instance too many repetitions are needed at one moment, which can happen in the case of bad luck. So the start/stop flow control protocol can be relayed by the similar functionality of RLP, once the type of flow control in use is konwn. A last interesting function of the start/stop protocol is the "break" signal, which is basically a violation of the start/stop rules (a break signal is a start signal longer than a character, i.e: such that the stop signal arrives too late). The break signal is used basically as a reset mechanism at the disposal of the user, to be used when things are going strangely. A special method is provided in the relay protocol to convey an indication of the reception of a break signal, which is regenerated by the receiver.

At last point for this presentation of the start/stop relay protocol concerns the efficiency of the method. If we take the worst case, which is a 9600 bit/s asynchronous connection with 7-bit characters and 1-bit long start and stop signals, we have a maximum throughput of 9600/9=1067 characters per second, that is 21 and one third characters per RLP frame period. A frame contains as a whole 240 bits, 40 of which being used for error detection, for frame numbering and for the acknowledgement and flow control protocol. A minimum of 8 is used for the modem control signals. There remains 192 bits, that is to say 24 octets (which corresponds to a data rate of 9.6 kbit/s exactly). A benefit of 2 2/3 octets is then obtined, allowing on average, one frame out of 8 to be repeated frame. For user rates of 4800 bit/s or less, things are obviously much better.

The relay method for protocols which deal with frames, such as HDLC, is rather different. Because the RLP replace the protocol, the digital stream coming from the external world can bi stripped from the overhead introduced by the replaced protocol. In the case of HDLC, this corresponds to the link layer header (2 octets per HDLC frame), the synchrounisation overhead (a minimum of 1 actet) and the error detection overhead (2 octets). The remaining data consists of chunks of variable length, which in general do not fit into the fixed length RLP frames. Frame delimination is than needed, and this is done using a special status octet after the last octet of the content of an HDLC frame. The final gain is then 4 octets per HDLC frame. The relative gain depends on the HDLC frame length The frame rate in RLP frames being exactly equal to the original rate, the breathing space obtained for repitition is never null, but is greater when the HDLC frames are smaller.

6.6.2- THE PTS-TRAU INTERFACE:

We have now presented the complete connection for data. Or so it seems. A last complication has to be studied, which is the transmission at 16 kbit/s. We have seen that in order for the operator to make economies on the cost of internal links, a scheme has been contrived to transport speech on 16 kbit/s links. The same thing had to be done for data. At first view, it seems that a modified RA2 function would do. But this would not take into account the constraint that only in-band information can be used by the BTS to control the TRAU. The problem is then to distinguish speech from data with in-band information, and there is no room available in V.110 frames for this.

What has been chosen in order to solve this issue is to specify specific rate adaptation modes on the BTS-TRAU interface, rather different from V.11, but compatible with the BTS-TRAU interface as specified for speech. This is schematically represented in figure 3.23. The bit stream is structured in 320 bit blocks, recurring each 20 ms. A notable difference is that the time alignment control, introduced for speech to prevent an additional transmission delay due to lack of synchronism with the radio path, is not user for data.

The simply the TRAU, the 63 bits resulting from the stripping of the 17 synchronisation bits from a V.110 frame are kept unmodified. Hence bits El, E2 and E3 are sent 2 or 4 times in the blocks (whether the intermediate rate is 8 or 16 kbit/s).

This results in the block contents as summarised in table 3.8. The structure is the same in both directions (uplink or downlink).

It is worth noting that, because the specification indicates that the fill bits (in the 8 kbit/s intermediate rate case) are set to "1" as in V.110, the distinction between the 8 kbit/s and the 16 kbit/s cases is totally irrelevant for the transcoder, which can perform the same operations in both cases while still respecting the specification.

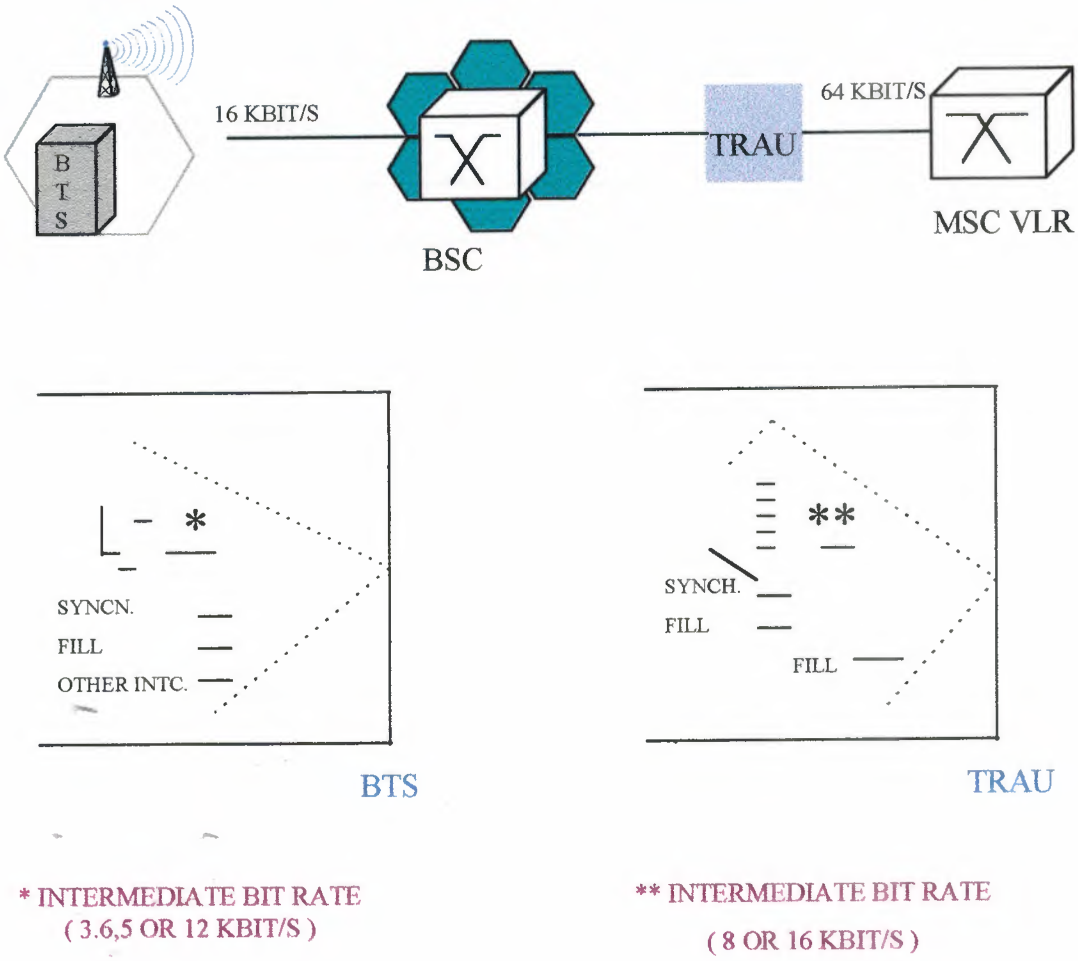


FIGURE 3.23 THE SPLIT BETWEEN BTS AND TRAU FOR RATE ADAPTATION.

Role of the bits	Number of bits per frame (uplink = downlink)
Frame synchronisation	35
Descrimination between speech and data, full rate and half rate	5
Intermediate rate on the TRAU/IWF	1

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