



NEAR EAST UNIVERSITY

Faculty of Engineering

**Department of Electrical and
Electronic Engineering**

CELLULAR COMMUNICATION SYSTEMS

**Graduation Project
EE- 400**

Submitted By: Amer Seder (20011812)

Supervisor: Assoc. Prof. DR. Sameer Ikhdair

Nicosia-2006

ACKNOWLEDGMENTS

At the beginning I would like to thank ALAH and my family, specially my parents there continuous support and endless love, brought me to this position. I would like to dedicate this work as a humble thanking for them. I wish them a place in the heaven after a long healthy and happy life.

Special thanks to my supervisor Assoc. Prof. Dr Sameer Ikhdair for being my advisor in this work. Under his supervision I was able to pass through many difficult problems in my project, I learned a lot from him about the communication and the telecommunications, he always answered my questions generously, and his answers were more than enough for me. I really appreciate his efforts in supporting me scientifically and immaterially.

Thanks to faculty of engineering specially and to Near East University generally for providing such an interesting educational environment.

Finally, I also want to thank my life friends: Omar Yasin, Hamzeh shatnawi, Ahmad Abu Shehab, Al Najjar, Adnan, Thaer and Haitham Abu Awwad, Being with them made 4 years of my life full of exciting, wonderful and fascinating moments, which I will never forgot.

ABSTACT

Recently, the demand for wireless communication has grown tremendously, and consequently cell sizes have decreased to meet this demand. Small cells are now used to increase the capacity of the system by reusing the resources more intensively in high traffic demand areas (Guerrero and Aghvami, 1999). Indeed, as small cells are needed to achieve higher capacities, increasing handoff rates are expected, leading to the undesirable consequence of an increase in the switching load of the network

This project is mainly explain the cellular communication system so that it gives a general information about the basic cellular system and also the operation of the cellular system, the required bandwidth and also the frequencies, so that it will give a full view for the reader about the cellular communication systems.

Results have formed the initial core base of the users' requirements. In addition, a technical analysis of the state-of-the-art of the mobile technologies has been conducted to identify key issues for the migration of existing services toward the UMTS, taking into account these users requirements.

TABLE OF CONTENTS

ACKNOWLEDGMENT	I
ABSTRACT.....	II
CONTENTS	III
INTRODUCTION	V
1. THEORY OF CELLULAR COMMUNICATION SYSYEMS	1
1.1 Introduction	1
1.2 Some Historical Notes	2
1.2.1 Binary Code	2
1.2.2 Telegraphy	2
1.2.3 Telephony	2
1.2.4 Radio	3
1.2.5 Satellite Communications	3
1.2.6 Optical Communications	3
1.2.7 Computer communications	3
1.3 Concept of Cellular System	4
1.4 Concept of Frequency Reuse.....	4
1.5 Cell Splitting.....	5
1.6 Air Interface Structure.....	6
1.7 Logical Channels.....	6
1.7.1 The Control Channels	6
1.7.2 Mapping on the Physical Channels.....	8
1.8 Handoff.....	13
1.8.1 Intra-BSC Handoff.....	15
1.8.2 Inter-BSC Handoff.....	17
1.8.3 Inter-MSC Handoff.....	21
2. CHANNEL CODING.....	27
2.1 The channel coding Theorem	27
2.2 Linear Block Codes	27
2.2.1 Syndrome Decoding	29
2.2.2 Minimum Distance Considerations	30

2.3 Cyclic Codes.....	30
2.3.1 Encoder for Cyclic Codes	31
2.4 Convolutional Codes	33
2.5 Code Tree, Trellis, and State Diagram	34
2.6 The Communications Channel	37
2.7 Electromagnetic Waves.....	38
2.8 Frequency and Wavelength.....	39
2.9 The Electromagnetic Spectrum	40
2.10 Bandwidth.....	41
2.11 Bandwidth and Channel Capacity	42
3. SPREAD SPECTRUM TECHNIQUES.....	43
3.1 General Concepts	43
3.2 Direct Sequence (DS) or PseudoNoise (PN).....	45
3.3 Biphase modulation.....	47
3.4 Quadriphase Modulation	48
3.5 PN Signal Characteristics.....	49
3.6 Frequency Hopping	50
3.6.1 The Frequency-Hopping Transmitter	51
3.6.2 The Frequency-Hopping Receiver	52
3.7 Hybrid Spread-Spectrum Systems	53
4. INTRODUCTION TO CELLULAR MOBILE SYSTEMS.....	54
4.1 Limitations of Conventional mobile telephone systems	54
4.1.1 Spectrum efficiency considerations	54
4.2 Basic Cellular System	55
4.3 Mobile fading characteristics	56
4.4 Operation of Cellular Systems	56
CONCLUSION	59
REFERENCES	60

INTRODUCTION

The purpose of a communication system is to transport an information bearing signal from a source to a user destination via a communication channel. Basically, a communication system is of an analog communication system, the information - bearing signal is continuously varying in both amplitude and time, and it is used directly to modify some characteristic of a sinusoidal carrier wave, such as amplitude, phase, or frequency bearing signal is processed so that it can be represented by a sequence of discrete message . While in a digital contain system , the information bearing signal is basically a stream of binary sequence modulated via phase, amplitude or frequency to form the well know modulation techniques PSK, ASK, and RSP.

So this project consists of four chapters, in chapter one we described the *theory of cellular communication systems*; the concept of the cellular system, the mapping of the physical channels also explained,

Chapter two includes the *channel coding*, the codes types (linear block coding, cyclic codes), the communication channels and the bandwidth,

Chapter three investigates the *spread spectrum technology*, the DS (direct sequence), and some of the modulation techniques.

Finally, chapter four discusses the cellular mobile system as an introduction, also the basic and the operation of the cellular system is described.

1. THEORY OF CELLULAR COMMUNICATION SYSTEMS

1.1 Introduction

Recently, the demand for wireless communication has grown tremendously, and consequently cell sizes have decreased to meet this demand. Small cells are now used to increase the capacity of the system by reusing the resources more intensively in high traffic demand areas (Guerrero and Aghvami, 1999). Indeed, as small cells are needed to achieve higher capacities, increasing handoff rates are expected, leading to the undesirable consequence of an increase in the switching load of the network

The handoff procedure is a means to continue a call even when a mobile station crosses the border of one cell into another. Figure (1.1) shows handoff process. Properly designed handoff procedure is essential in maintaining the quality of a call in progress and in keeping as low as

Possible both the probability of forced termination of the call itself and the signaling and switching load to the network.

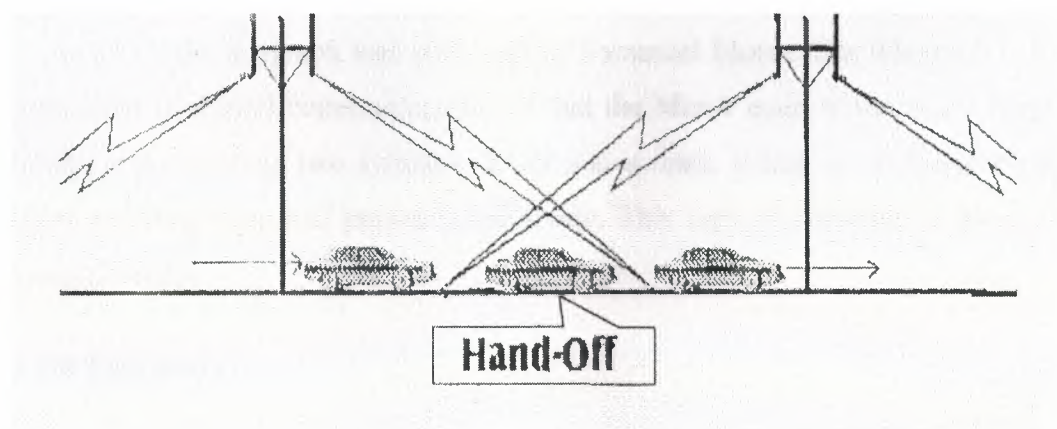


Figure 1.1 *Handoff process*

In the following sections: the concept of cellular system, frequency reuse, cell splitting, an overview of the air interface structure and the handoff procedure will be discussed.

1.2 Some Historical Notes

In this section, we present some historical notes on communications, with emphasis on digital communications and related issues.

The material is organized under separate categories:

1.2.1 Binary Code

The origins of the binary code, basic to the operation of digital communications, may be traced back to the earls' work of Frances Bacon at the beginning of the seventeenth century.

In 1703, Gottfried Wilhelm Leibnitz gave a lecture to Royal Academy of Sciences in Paris, entitled 'Explication de l'arithmetique Binaire' The text of his lecture was published in the proceedings of the Academy in 1750. Leibnitz used the numbers 0 and 1 for his binary code.

It appears that leibnitz's binary code was developed independent from Bacon and Wilkins.

1.2.2 Telegraphy

In 1837, the telegraph was perfected by Sammuel Morse. The telegraph is the forerunner of digital communications in that the Morse code is variable - length binary code utilizing two symbols, a dot and a dash, which are represented by short and long electrical pulses, respectively. This type of signaling is ideal for manual keying.

1.2.3 Telephony

In 1874, the telephone was conceived by Alexander Graham Beill in Brantford, Ontario, and it was born in Boston, Massachusetts in 1875. The telephone made real - time transmission of speech by electrical encoding and replication of sound a practical reality.

1.2.4 Radio

In 1864, James Clerk Maxwell formulated the electromagnetic theory of Light and predicted the existence of radio waves. The existence of radio waves was established by Heinrich Hertz in 1887.

It appears that digital modulation techniques were first employed for microwave radio transmission in France in the 1930s. Then, after a long pause, digital radio (i.e., digital communications by radio) experienced a renaissance in the early 1970s.

1.2.5 Satellite Communications

In 1945, Arthur C. Clarke proposed the idea of using an earth - orbiting satellite as a relay point for communication between two earth stations. In 1957, the Soviet Union Launches Sputnik, I which transmitted telemeter signals for 21 days. This was followed shortly by the launching of Explorer I by the United States in 1958, which transmitted telemetry signals for about five months. A major experimental step in communications satellite technology was taken with the launching of Telstar from Cape Canaveral on July 10, 1962.

1.2.6 Optical Communications

The use of optical means (e.g., smoke and fire signals) for the transmission of information dates back to prehistoric times. However, no major breakthrough in optical communications was made until 1966, when Kao and Hockham proposed the use of a clad glass fiber as a dielectric waveguide.

1.2.7 Computer communications

Computers and terminals started communicating with each other over long distances in the early 1950s. The links were initially voice-grade telephone channels operating at low speeds (30 to 1200 b/s). Today, telephone channels are routinely used to support data transmission at rates of 9.6 kb/s or even as high as 16.8 kb/s.

The processing techniques of communications signal has arisen during the past two decades. The material is developed in the context of a structure used to trace the processing steps from the information source to the information sink. Transformations are organized according to functional classes: Formatting and source coding, modulation, channel coding, multiplexing and multiple accesses, frequency spreading, encryption, and synchronization.

1.3 The Concept of Cellular System

Cellular is a system concept that has come into being because radio spectrum (the frequencies that carry the radio messages) is a limited resource.

The concept of cellular systems is the use of low power transmitters in order to enable the efficient reuse of frequencies. In fact, if the transmitters which are used are very powerful, the frequencies can not be reused for hundreds of kilometers as they are limited to the covering area of the transmitter. So, in a cellular system, the covering area of an operator is divided into cells. A cell corresponds to the covering area of one transmitter or a small collection of transmitters. The size of a cell is determined by the traffic generated in the area and /or the time advanced.

The frequency band allocated to a cellular mobile radio system is distributed over a group of cells and this distribution is repeated in all the covering area of an operator. The whole number of radio channels available can then be used in each group of cells that form the covering area of an operator. Frequencies used in a cell will be reused several cells away. The distance between the cells using the same frequency must be sufficient to avoid interference. The frequency reuse will considerably increase the capacity in number of users.

1.4 Concept of Frequency Reuse

Frequency reuse is the core concept of the cellular mobile radio system. A particular radio channel, say f_1 , used in one geographic one to call a cell with a coverage radius r can be used ii) another cell with the same coverage radius at a distance d away. Figure (1 .2) shows frequency reuse concept.

In this frequency reuse system, users in different geographic locations (different cells) may simultaneously use the same frequency channel. This can drastically increase the spectrum efficiency; however, serious interference known as cochannel interference may occur if the system is not properly designed. (Lee, 1996).

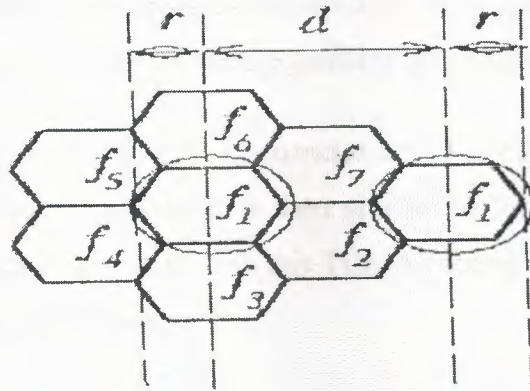


Figure 1.2 frequency reuse concept

1.5 Cell Splitting

In addition to frequency reuse, cell splitting may be implemented to improve the utilization of spectrum efficiency. When traffic density starts to build up and the frequency channels in each cell cannot provide enough mobile calls, the original cell can be split into smaller cells. Figure (1 .3) shows cell splitting concept (Lee, 1996).

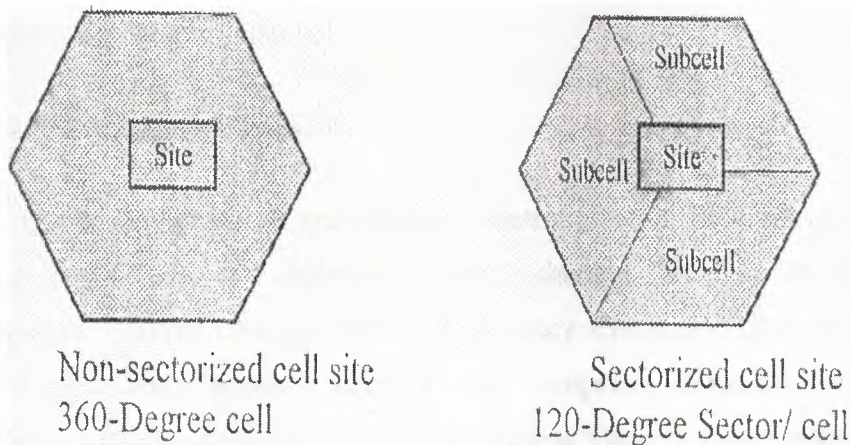


Figure 1.3 cell splitting concept

1.6 Air Interface Structure

Since radio spectrum is a limited resource shared by all users, a method must be devised to divide up the bandwidth among as many users as possible. The method chosen by (Global System for Mobile Communication) GSM is a combination of Time- and Frequency-Division Multiple Access (TDMA/FDMA). The FDMA part involves the division by frequency of the maximum 25 MHz bandwidth into 124 carrier frequencies spaced 200 kHz apart.

One or more carrier frequencies are 11 assigned to each base station. Each of these carrier frequencies is then divided in time, using a TDMA scheme. The fundamental unit of time in this TDMA scheme is called a burst period with time duration of 0.577 ms.

Eight burst periods are grouped into a TDMA frame which forms the basic unit for the definition of Logical channels. The logical channel is specific type of information carried by a physical channel, where the physical channel is the medium over which the information is carried.

1.7 Logical Channels

In order to exchange the information needed to maintain the communication links within the cellular network, several radio channels are reserved for the signaling information, so the logical channel carries a user's data, or signaling data. In other words, there are two main groups of logical channels, traffic channels and control channel.

1.7.1 The Control Channels

The control channels are broadcast control channel (BCCH), common control channel (CCCH), and dedicated control channel (DCCH). BCCH comprises Broadcast Control Channel (BCH), Frequency Correction Channel (FCCH) and Synchronization Channel (SCH). CCCH comprises Random Access Channels (RACH), Paging Channel (PCH) and Access Grant Channel (AGCH). DCCH comprises Stand-Alone Dedicated

Control Channel (SDCCH, Slow Associated Control Channel (SACCH) and Fast Associated Control Channel (FACCH). Figure (I .4) shows the different logical channels. The details of BCH, SACH and FACH are given only, since these channels are associated with handoff procedure.

- **Broadcast control Channel (BCH):** The broadcast control channel is transmitted by the Base Transceiver Station (BTS) at all times to inform Mobile Station about specific system parameters including location area identity (LAI), list of neighboring cells, list of frequencies used in the cell and cell identity. So the Mobile Station (MS) should monitor [periodically (at least 30 sec), when it is switched on and not in a call] downlink information that is transmitted on broadcasts channel.

The BCH is transmitted at constant power at all times, and its signal strength is measured by all MS. “Dummy” bursts are transmitted to ensure continuity when there is no BCH carrier traffic. BCH is transmitted downlink, point-to-multipoint.

- **Slow Associated control Channel (SACCH):** is used to transfer signaling data while an ongoing conversation on a TCH is in progress or while the SDCCH is being used. This channel can carry about two messages per second in each direction. It conveys power control and time information in the downlink direction and receives signal strength indicator (RSI), and link quality report in the uplink direction. SACCH is transmitted both up-and downlink, point-to-point.

- **Fast Associated control channel (FACCH):** is used when there is a need for higher capacity signaling in parallel with ongoing traffic. FACCH works in “stealing mode”, meaning that FACCH “steal “the TCH burst and insert its own information so the FACCH is transmitted instead of a TCH. When doing so, the transmitting side must set the “stolen bit indicator” to 1. When noting, on the receiving side, that the stolen bit indicator sends 1, the bursts will be handled as signaling information. To lessen the disturbance of the speech, the last speech segment will be

repeated. The FACCH is mainly used for Handoff commands. FACCH is transmitted both up-and down link, point-to-point.

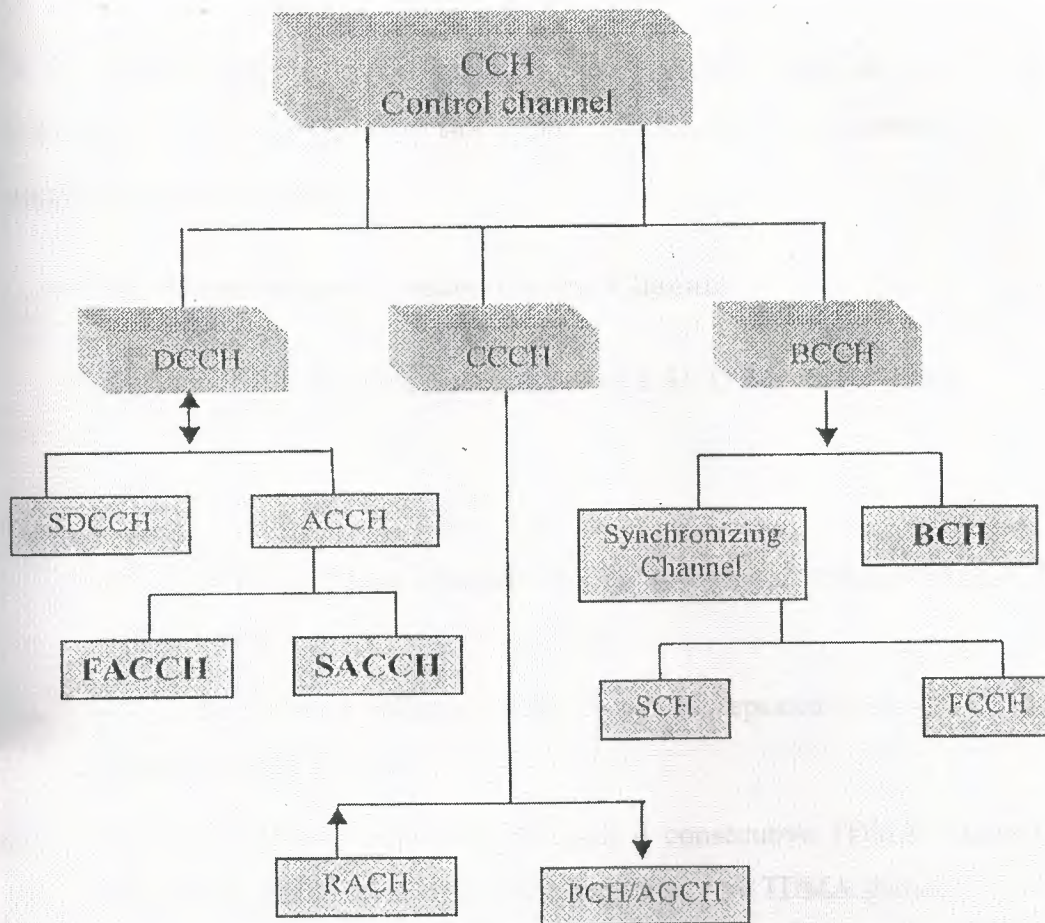


Figure 1.4 *logic channels in GSM*

1.7.2 Mapping on the Physical Channels

The logical channels are mapped, or multiplexed on the physical channels which mean that the control channels mentioned above are transmitted according to certain rules concerning what physical channel (frequency and time slot) to use and how often they are to be repeated.

The TDMA-frames are grouped together into multi-frames that are then repeated cyclically. There are basically two types of multi-frame; the 26 TDMA multi-frame used for traffic and the 5 I TDMA multi-frames used for control signaling. One super frame consists of 51 traffic multi-frames or 26 control multi-frames and consists of 51x26 TDMA frames with a total duration of 6.12 sec. The highest order frame is called a hyper frame and consists of 2048 super frames or

2715648 frames. The time duration of the hyper frame is 3 hours, 28 min, and 52.76 sec (Mehrotra, Asha 1996).

At a base station with n carriers, each with eight time slots, the carriers are called $C_0, C_1, C_2, \dots, C_n$. On time slot 0 on C_0 a channel combination of only control channels are mapped.

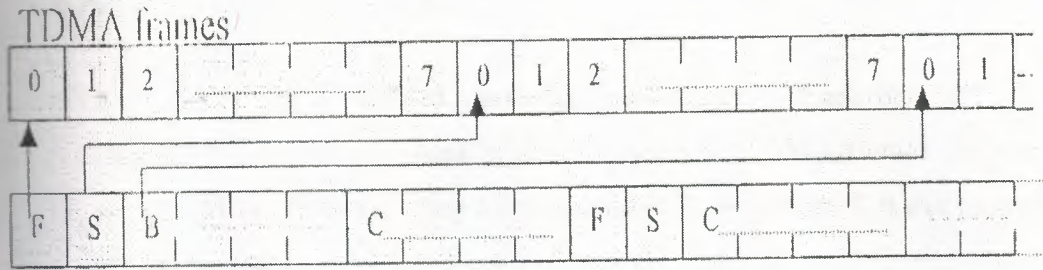
- **The Broadcast and Common Control Channel**

TS0 on C_0 are grouping the information into a 51 TDMA multi-frame.

It contains:

- BCH, Broadcast channels FCCH, always start the multi-frame. It will be repeated every 10 TDMA-frames.
- SCH always follows FCCH. It will be repeated every 10 TDMA frames, just like FCCH.
- BCCH will come next. It needs 4 consecutive TDMA frames to transmit the information and it will repeat every 50 TDMA frames.
- CCCH (Common Control Channels). CCCH downlink could be either PCH or AGCH. It will use a block of four consecutive TDMA frames. Nine CCCH-blocks can be fitted in one 51 TDMA multi-frame.
- I stand for Idle, even though in this case it is really a dummy burst being transmitted. Since other MSs might be measuring signal strength by monitoring this physical channel, something must always be transmitted. Therefore, in TDMA frame 51, when we have nothing to send, a dummy burst will nevertheless be sent.

Figure (1.5) shows mapping of logical channel on TS0 on C_0 downlink and uplink.



F:-FCCH

S:-SCH

B: -BCCH

C: - CCCH (PCH or AGCH)

Figure 1.5 (a) Mapping of logical channels on TSO on C_0 downlink.

In uplink, the only logical channel to be mapped is the access channel (RACH)

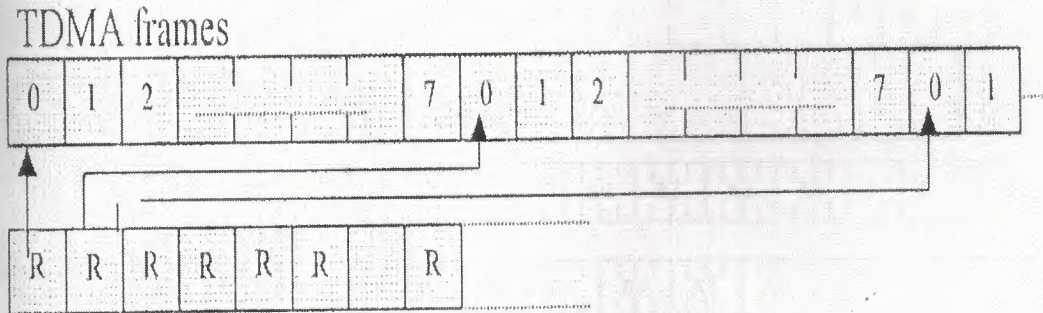


Figure 1.5 (b) mapping of logical channels on TSO on C_0 uplink.

• The Dedicated Control Channels

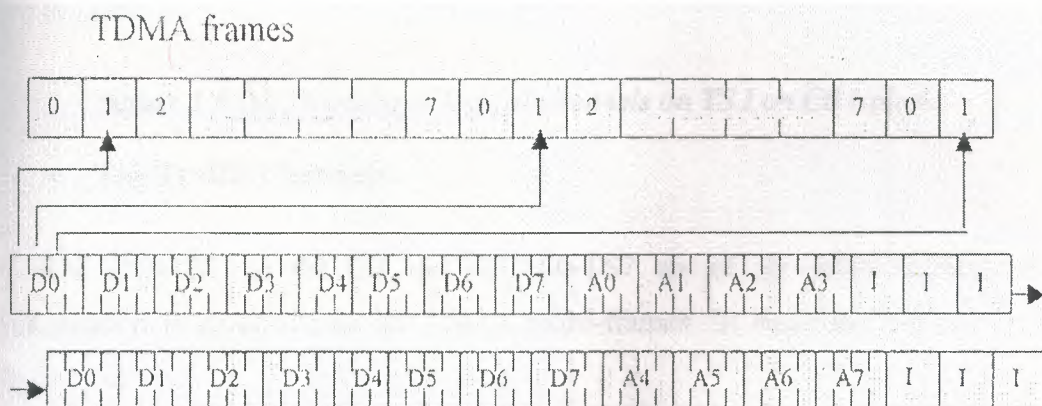
The Dedicated Control Channel is usually mapped on TSI on C_0 up-as well as downlink the information is grouped into 102 TDMA multi- frames.

In this multi-frame, it is found that:

- SDCCH is divided into eight sub-channels. Each SDCCH sub-channel is occupying a block of four consecutive TDMA frames. As soon as the MS has finished using a certain SDCCH sub-channel, it can be used by another MS.
- SACCH, for each SDCCH sub-channel there is a corresponding SACCH. This channel is used to transfer signaling information concerning measurements during call set-up.

The uplink looks similar to the downlink for TS1; C_0 the only difference is that the uplink is a number of TDMA frames delayed in relation to downlink.

Figure (1.6) shows mapping of logical channel on TS1 on C_0 downlink and uplink



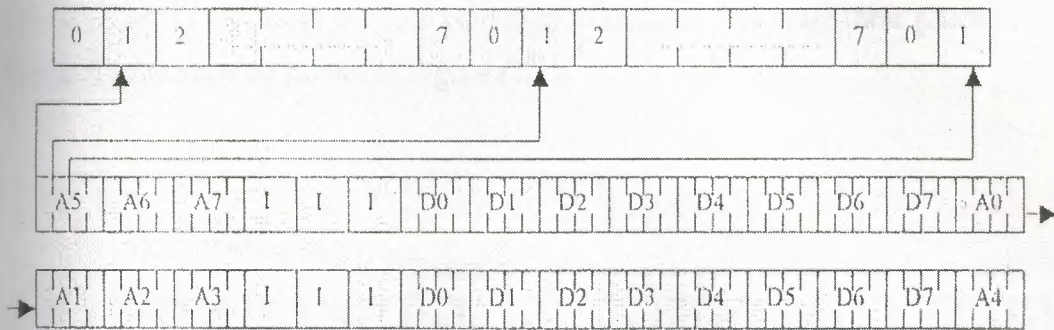
SDCCH + SACCH

Dx: - SDCCH

Ax: - SACCH

I: - Idle

Figure 1.6 (a) Mapping of logical channels on TS 1 on C_0 downlink.



SDCCH + SACCH

Dx: - SDCCH

Ax: - SACCH

I: - Idle

Figure 1.6 (b) Mapping of logical channels on TS 1 on C0 Uplink

• The Traffic Channels

On TS2-TS7 on the C0 and on TS0-TS7 on all the other carriers, the information is grouped into 26 TDMA multi-frames. In these multi-frames it is that;

- TCH, containing data or speech.
- SACCH, carrying the control signaling necessary during traffic, for instance measurement data, power order, or timing advance order.
- Idle frames, this is not a logical channel, rather it is used to indicate that the transmitter is off during this particular multi-frame. Figure (1.7) shows 26 TDMA—multi-frame.

1.8 Handoff

Handoff is the switching of an on-going call to a different cell, which happens when a user moves from one cell coverage to another. There are three phases of a handoff procedure as shown in Figure (1.8).

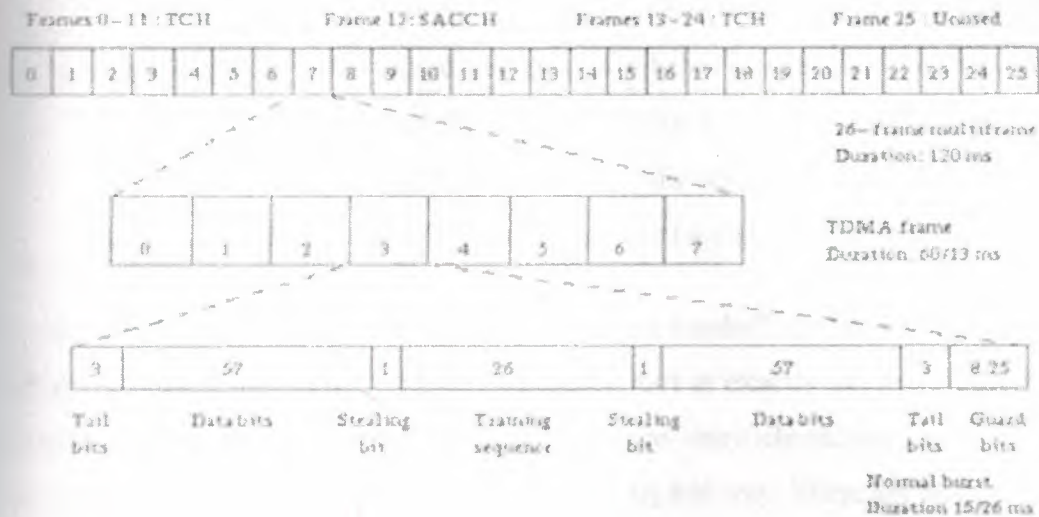


Figure 1.7 26-TDMA multi-frames

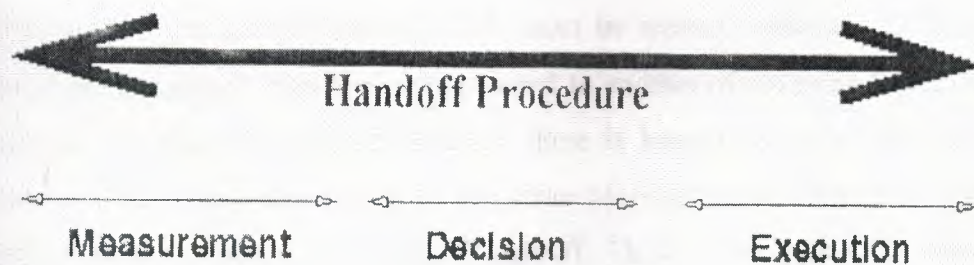


Figure 1.8 Phases of a handoff procedure

These phases are:-

- **Measurements:** The mobile terminals as well as the access point (Base Transceiver Station) do several measurements continuously. For e.g.

the signal strength is one parameter which might be measured by both the terminal and the access point (Graziosi *et al.* 1999). In GSM, the mobile station transmits report on up to 6 neighboring cells in addition to the measurements relative to the serving cell, this reporting is carried by messages on the small signaling channel associated with each traffic channel and called the SACCH.

- Decision: based on the measurements taken, a decision is made as to whether a handoff is required. For e.g. a decision to perform a handoff might be taken if the signal strength goes below a specified threshold. In GSM, the decision is taken by the Base Station Controller (BSC).
- Execution: the actual handoff of the terminal from one cell to another is performed in this phase. There are two modes of handoff:

Synchronous and asynchronous. In synchronous handoff, the old and new cells are synchronized so that their TDMA timeslots start at exactly the same time. In asynchronous handoff, the old and new cells are unsynchronized, so the MS cannot independently correct the timing advance in this way. There are essentially two sub-phases in the execution of the handoff:

- New Link establishment.
- Release of old link.

There are three types of handoff in GSM based on the position of the switching point at handoff, all of which must be treated somewhat differently. First, there is handoff from one radio channel to another of the same BSC, which is known as intra-HSC handoff. Second, there is handoff between channels of different BSCs under the control of the same Mobile-Service Switching Center (MSC), which is known as inter-BSC handoff. Third, there is handoff between channels under the control of different MSCs in the same Public Land Mobile Network (PLMN), which is known as inter-MSC handoff. Figure (1 .9) shows these three types of handoff. The detailed protocols for these three types will be provided in the following section.

1.8.1 Intra-BSC Handoff

Figure (1.10) shows a handoff process between channels of the same BSC. The MS is shown at both ends, indicating its connection to the old and new BTSs. With a call in progress, the BSC may determine if a change of channel is necessary. The BSC is aware of all the relevant information since it already manages the current context of the connection. The BSC allocates TCH in a new cell, chooses handoff reference number which it uses to determine whether the correct mobile gains access to the air-interface channel which it allocates, then the BSC orders BTS-new to activate it with a "Radio Subsystem Management (RSM) Channel activation" message. BTS-new responds with an "RSM Channel Activation Acknowledge" message to the BSC. The BSC then sends a "Radio interface layer 3-Radio Resources (RIL 3-RR) Handoff Command" message to the MS on the FACCH, via BTS-old, assigning the new channel, its characteristics, new SACCH, and whether to use synchronous or asynchronous handoff. Upon receiving this message, the MS suspends all transmission of signaling messages except those RR messages concerning the Handoff until resuming is indicated by set asynchronous balanced mode (SABM) message, initiates the release of the old channel and connection to the new one.

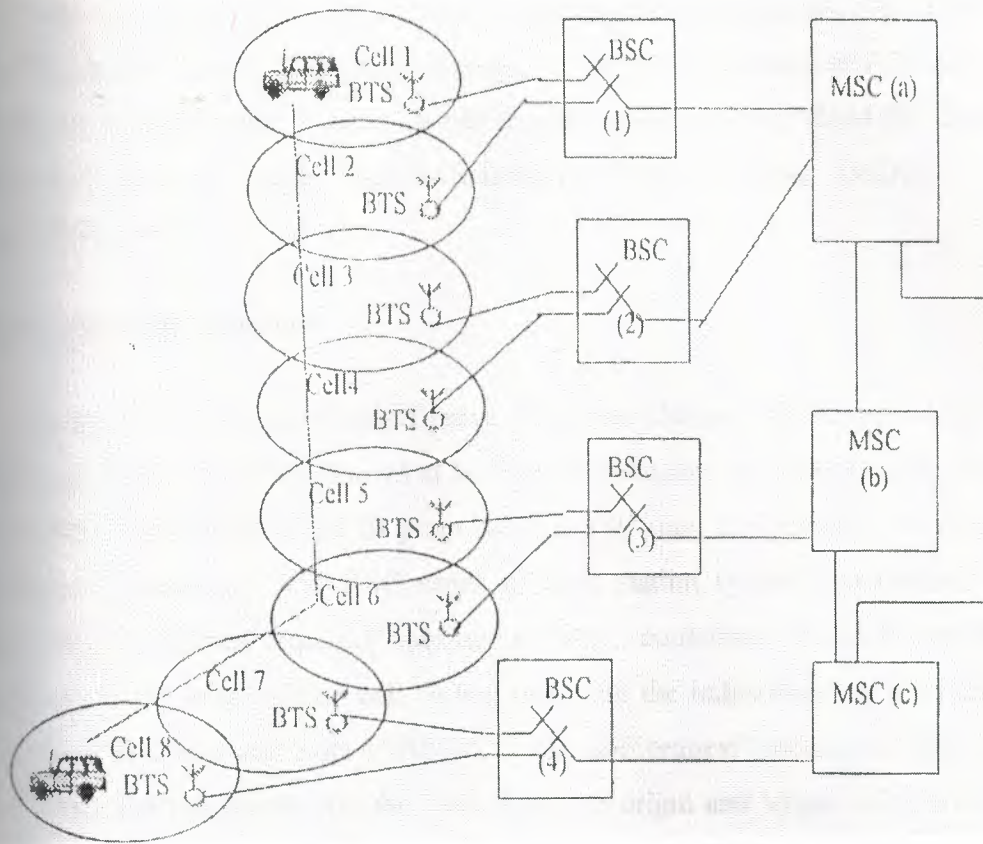


Figure 1.9 *Mobile handoff*

Two procedures are possible depending on whether the on and new cells are synchronized or not. In the synchronous mode, after switching to the new channel, the MS sends to the new BTS, in successive assigned multi-frame slots on the FACCH, four “RIL3-RR Handoff Access” messages. It then activates the new channel in both directions. When it has received sufficient “Handoff Access” messages, the new BTS may also send an “RSM Handoff Detection” message to the BSC.

In asynchronous mode, the MS starts sending a continuous stream of “RIL3-RR Handoff Access” messages to the new BTS until it receives in response an “RIL3-RR Physical information” message giving the timing advance to apply. For efficiency reasons, the “RIL3-RR physical information” message may be sent several times in a row, until the reception of “set asynchronous balances mode (SABM)” frame from the MS makes it clear to BTS-new that it has received the message, this message answered by an “unnumbered ACK (UA)” frame.

After the lower layer connections are successfully establishes, the MS sends an "RIL3-RR Handoff Complete" message to the BSC over the new FACCH. The BSC directs BTS-old to release the old channel by sending an "RSM RF Channel Release" message with Acknowledgement from BTS-old (MOULY and PAUTET, 1992).

1.8.2 Inter-BSC Handoff

Figure (1.11) shows a handoff process between channels of the same MSC but different BSC. The MS is shown at both ends, indicating its connection to the old and new BTSs. With a call in progress, the BSC may determine if a change of channel is necessary. The BSC sends a "base station system management part (BSSMAP) Handoff required" message to MSC, containing the identities of the target cell and of the origin cell. When receiving the indication that a handoff is required, the MSC transmits a "BSSMAP Handoff request" message to BSC-new, including the information on the cells (both the origin and target cells), the class mark and the cipher mode. The BSC-new allocates TCH in new cell, choose handoff reference number then order BTS-new to activate it by a "Radio Subsystem Management (RSM) Channel activation" message. BTS-new responds with an "RSM Channel Activation Acknowledge" message to the BSC-new. The BSC-new encapsulates the "RIL3-RR Handoff Command" message in a "BSSMAP Handoff Request Acknowledge" message.

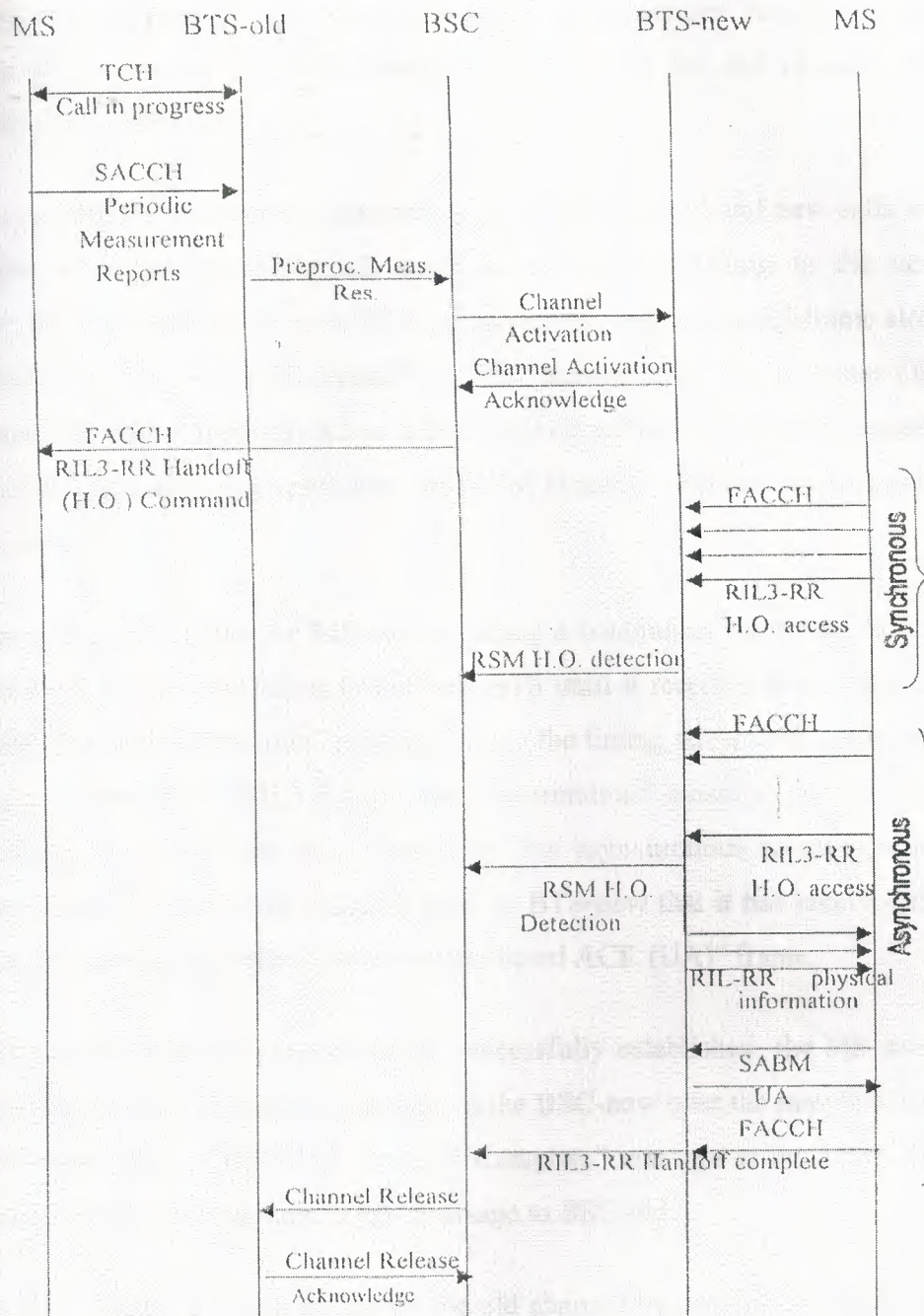


Figure 1.10 *intra-BSC Handoff*

The MSC transmit “BSSMAP Handoff Command” message which contain everything the MS may need to access the new channel (such as handoff reference number, assignment of a new SACCH, whether to use synchronous handoff or asynchronous handoff). The BSC—old then sends an “Radio interface layer 3-Radio Resources (RIL3-RR) Handoff Command” message to the MS on the FACCH, via BTS0-old, assigning the new channel, its characteristics, new

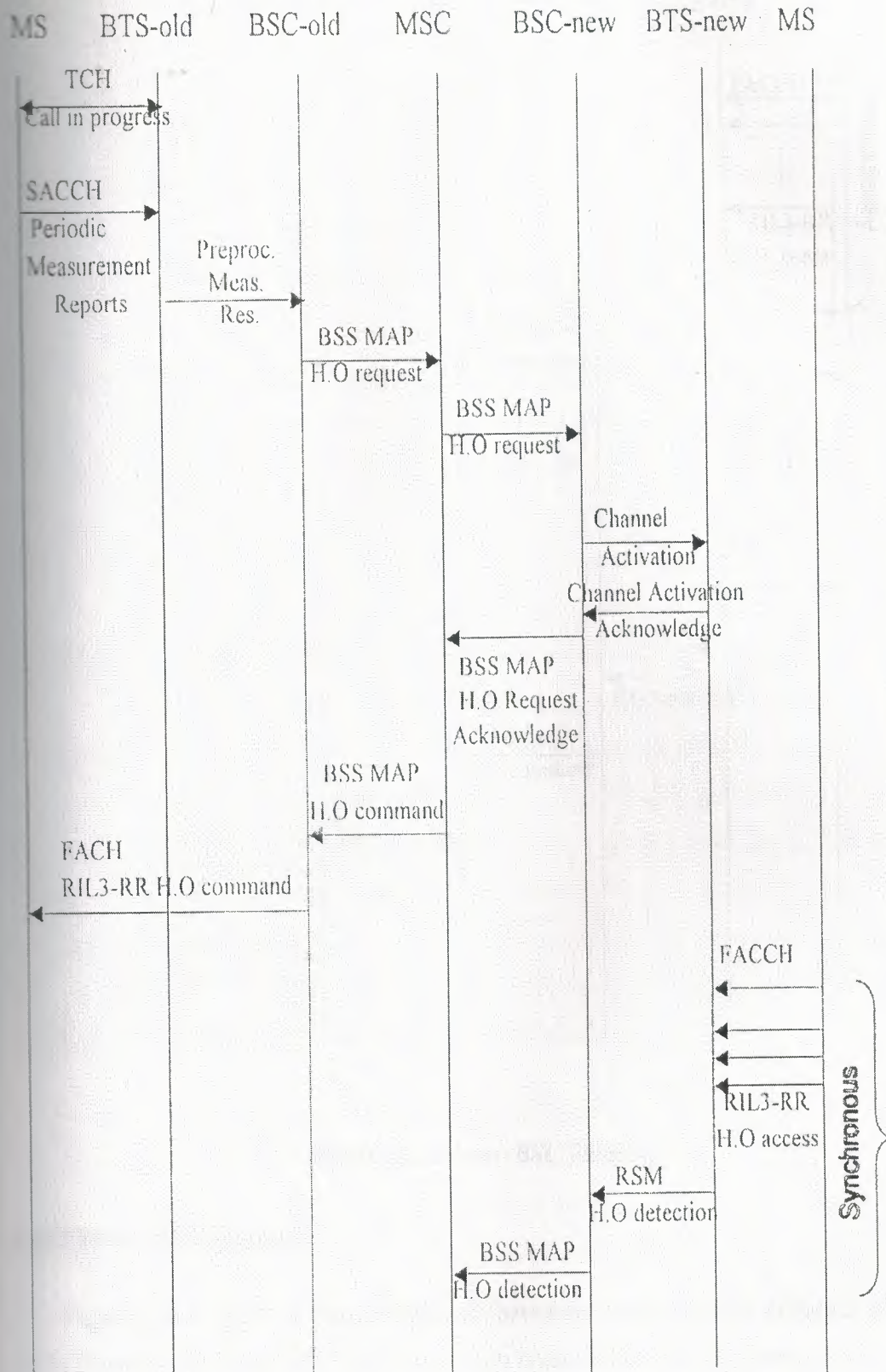
SACCH, and whether to use synchronous or asynchronous handoff. Upon receiving this message, the MS initiates the release of the old channel and connection to the new one.

Two procedures are possible depending on whether the old and new cells are synchronized or not. In the synchronous mode, after switching to the new channels, the MS sends to the new BTS, in successive assigned multi-frame slots on the FACCH, four "RIL3-RR Handoff Access" messages. It then activates the new channel in both directions. When it has received sufficient "Handoff Access" messages, the new BTS may also send an "RSM Handoff Detection" message to the BSC-new.

In asynchronous mode, the MS starts sending a continuous stream of "RIL3-RR Handoff Access" messages to the new BTS until it receives in response an "RIL3-RR Physical information" message giving the timing advance to apply. For efficiency reasons, the "RIL3-RR physical information" message may be sent several times in a row, until the reception of "set asynchronous balances mode (SABM)" frame from the MS makes it clear to BTS-new that it has received the message, this message answered by a "unnumbered ACK (UA)" frame.

After the lower layer connections are successfully established, the MS sends an "RIL3RR Handoff Complete" message to the BSC-new over the new FACCH. The BSC-new sends a "BSSMAP Handoff Complete" message to the MSC. The MSC send "BSSMAP clear command" message to BSC-old.

The BSC directs BTS-old to release the old channel by sending an "RSM RF Channel Release" message with Acknowledgment from BTS- old. (MOULY and PAUTET, 1992, and GSM 03.09)



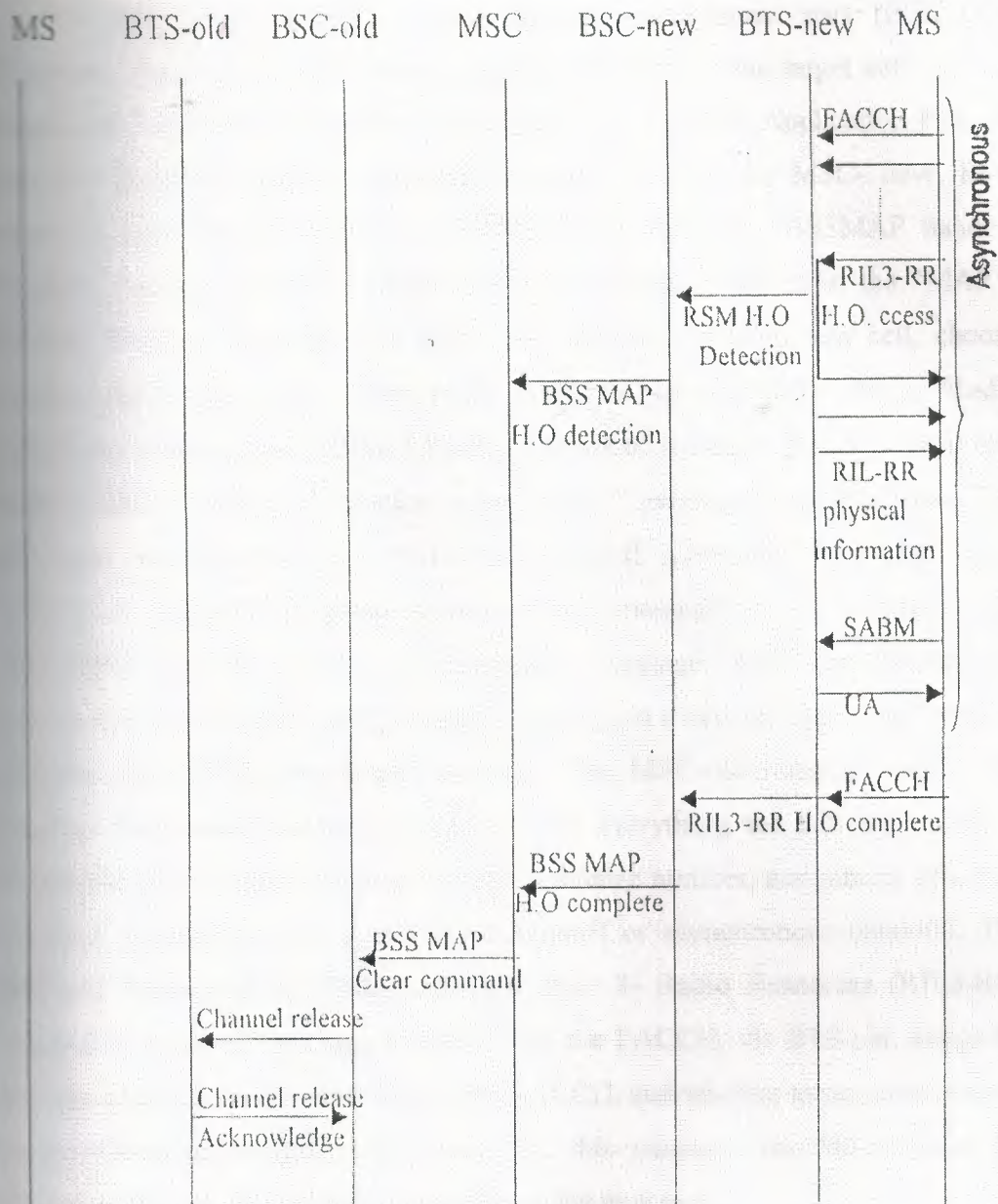


Figure 1.11 *Inter-BSC Handoff*

1.8.3 Inter-MS Handoff

Figure (1.12) shows a Handoff process between channels of the different BSC and different MSC. The MS is shown at both ends, indicating its connection to the old and new BTSs. With a call in progress, the BSC may determine if a change of channel is necessary.

The BSC sends a "base station system management part (BSSMAP) Required" message to MSC, containing the identities of the target cell and the origin cell. The MSC translates the message in a "Mobile Application Part, E-interface (MAP/F) Perform Handoff" message towards the MSC-new. Both messages have similar contents. The MSC-new transmits "BSSMAP handoff Request" message to BSC containing the information received in the "MAP/E Perform Handoff" message. The BSC-new allocates TCH in new cell, choose handoff reference number then order BTS-new to activate it by a "Radio Subsystem Management (RSM) Channel activation" message. BTS-new responds with an "RSM Channel Activation Acknowledge" message to the BSC-new. The BSC-new encapsulates the "RIL3-RR Handoff Command" message in a "BSSMAP Handoff Request Acknowledge" message. When receiving the "BSSMAP handoff request Acknowledge" message, MSC-new inserts the included "RIL3-RR Handoff Command" message in a new envelope, the "MAP/E perform Handoff Acknowledge" message. The MSC-old transmit "RIL3- RR Handoff Command" message which contain everything the MS may need to access the new channel (such as handoff reference number, assignment of a new SACCH, whether to use synchronous handoff or asynchronous handoff). The BSC-old then sends a "Radio interface layer 3- Radio Resources (RIL3-RR) Handoff Command" message to the MS on the FACCH, via BTS-old, assigning the new channel, its characteristics, new SACCH, and whether to use synchronous or asynchronous handoff. Upon receiving this message, the MS initiates the release of the old channel and connection to the new one.

Two procedures are possible depending on whether the old and new cells are synchronized or not. In the synchronous mode, after switching to the new channels, the MS sends to the new BTS, in successive assigned multiframe slots on the FACCH, four "RIL3-RR Handoff Access" messages. It then activates the new channel in both directions. When it has received sufficient "Handoff Access" messages, the new BTS may also send an "RSM Handoff Detection" message to the BSC-new.

In asynchronous mode, the MS starts sending a continuous stream of "RIL3-RR Handoff Access" messages to the new BTS until it receives in response an

"RIL3-RR Physical information" message giving the timing advance to apply. For efficiency reasons, the "RIL3-RR physical information" message may be sent several times in a

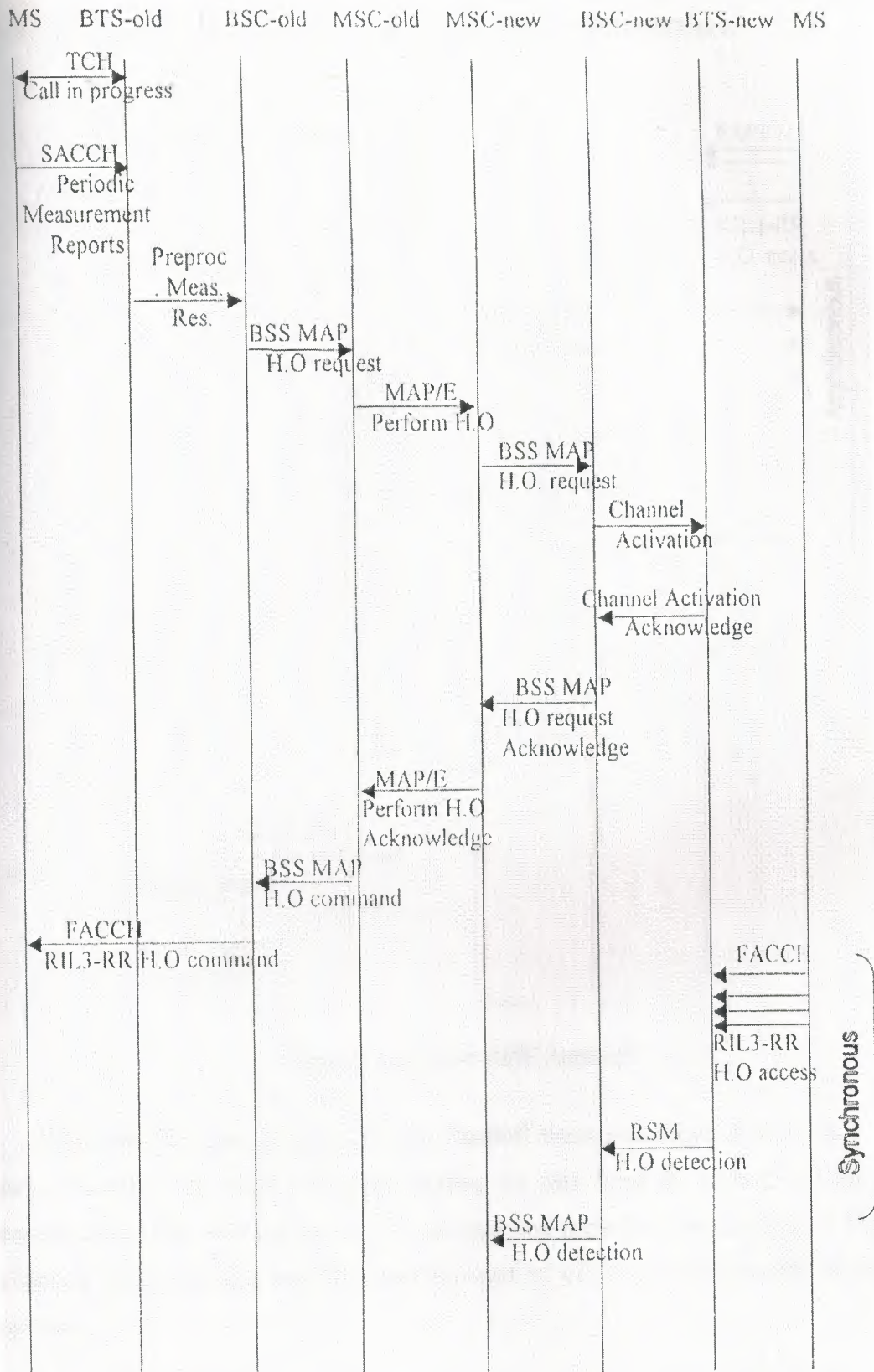
row, until the reception of "set asynchronous balances mode (SABM)" frame from the MS makes it clear to BTS-new that it has received the message, this message answered by a "unnumbered ACK (UA)" frame.

After the lower layer connections are successfully established, the MS sends an "RIL3-RR Handoff Complete" message to the BSC-new over the new FACCH. The BSC-new sends a "BSSMAP Handoff Complete" message to the MSC-new. The MSC-new sends "MAP/E send end signal" message to MSC-old. The MSC-old sends "BSSMAP clear command" message to BSC-old. The BSC directs BTS-old to release the old channel by sending an "RSM RF Channel Release" message with Acknowledgment from BTS-old (MOULY and PAUTET, 1992, and OSM 03.09).

It can be seen from the Figure (1.10) for intra-BSC handoff that when the BSC determines if a change of channel is necessary, it allocates TCH in new cell then orders BTS-new to activate it, while the MS is still connected to the old channel and will not release it until receiving Handoff Command" message on the FACCH via BTS-old, assigning the new channel, then MS is connected to this new one, but the old channel will not be released by the BTS-old until it receives "RSM RF Channel Release" message from BSC.

Similarly, from Figure (1.11) for Inter-BSC handoff and Figure (1.12) for inter-MSC handoff, when the BSC determines if a change of channel is necessary BSC sends a "base station system management part (BSSMAP) Handoff Required" message to MSC, containing the identities of the target cell and the origin cell then MSC decides if this handoff is between different BSCs under its control or Handoff between BSC under its control and BSC under the control of another Mobile-Service Switching Center (MSC), in both cases, the BSC-new allocates TCH in new cell, then orders BTS-new to activate it, while the MS is still connected to the old channel and will not release it until receiving Handoff Command" message on the FACCH, via BTS-old, assigning the new channel then

MS is connected to this new one, but the old channel will not be released by the BTS-old until it receives "RSM RF Channel Release" message from BSC-old.



Whatever the type of handoff, each handoff execution requires to initiate a new channel in the target cell while holding the path from the current cell for a certain time. This will reduce the overall systems capacity. The holding of two channels time during a handoff execution and its effect on the capacity of the system.

2. CHANNEL CODING

2.1 The channel coding Theorem

The channel coding theorem states that if a discrete memory less channel has capacity C and source generated information at a rate less than C , then there exists a coding technique such that the output of the source may be transmitted over the channel with an arbitrarily low probability of symbol error.

The theorem thus specifies the channel capacity C as a fundamental limit on the rate which the transmission of reliable messages can take place over a discrete memory less channel.

The most unsatisfactory feature of the channel coding theorem, however, is the non constructive nature. The theorem only asserts the existence of good codes. The error-control coding techniques provide different methods of achieving this important system requirement. We consider block codes first, followed by convolution codes, and then trellis codes.

2.2 Linear Block Codes

Consider an (n, k) linear block code in which the first portion of k bits is always identical to the message sequence to be transmitted. The $n-k$ bits in the second portion are referred to as generalized parity check bits or simply parity bits. Block codes in which the message bits are transmitted in unaltered form are called systematic codes. For applications requiring both error detection and error correction, the use of systematic block codes simplifies implementation of the decoder.

Let m_0, m_1, \dots, m_{k-1} constitute a block of k arbitrary message bits. Thus we have 2^k distinct message blocks. Let this sequence of message bits be applied to a linear block encoder, producing an n -bit code word whose elements are denoted by x_0, x_1, \dots, x_{n-1} . Let b_0, b_1, \dots, b_{n-k} denote the $(n-k)$ parity bits in the code word.

Clearly, we have the option of sending the message bits of a code word before the parity bits, or vice versa. The former option is illustrated in Figure 2.1 the $n-k$ left-most bits of a code word are identical to the corresponding parity bits, and k right-most bits of the code word are identical to the corresponding message bits.



Figure 2.1 Structure of code word.

We define the 1-by- k message vector m , the 1-by- $(n-k)$ parity vector b , and the 1-by- n code vector x as follows, respectively:

$$m = [m_0, m_1, \dots, m_{k-1}] \quad (2.2.1)$$

$$b = [b_0, b_1, \dots, b_{n-1}] \quad (2.2.2)$$

and

$$x = [x_0, x_1, \dots, x_{n-1}] \quad (2.2.3)$$

$$b = mP \quad (2.2.4)$$

Where P is the k -by- $(n-k)$ coefficient matrix defined by:

$$P = \begin{bmatrix} P_{00} & P_{10} & \dots & P_{n-k-1,0} \\ P_{01} & P_{11} & \dots & P_{n-k-1,1} \\ \vdots & \vdots & \ddots & \vdots \\ P_{0,k-1} & P_{1,k-1} & \dots & P_{n-k,k-1} \end{bmatrix} \quad (2.2.5)$$

$$x = [b : m] \quad (2.2.5)$$

We get

$$x = m[P : I_k] \quad (2.2.6)$$

Where I_k is the k -by- k identity matrix:

$$I_k = \begin{bmatrix} 1 & 0 & \dots & 0 \\ 0 & 1 & \dots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \dots & 1 \end{bmatrix}$$

Define the k -by- n generator matrix*

$$G = [P : I_k] \quad (2.2.7)$$

Then,

$$x = mG \quad (2.2.8)$$

Let H denote an $(n-k)$ -by- n matrix, defined as

$$H = [I_{n-k} : P^T] \quad (2.2.9)$$

$$xH^T = mGH^T = 0 \quad (2.2.10)$$

The matrix H is called parity-check of the code,

2.2.1 Syndrome Decoding

The generator matrix G is used in the encoding operation at the transmitter. On the other hand, the parity-check matrix H is used in the decoding operation at the receiver. Let y denote the 1 -by- n received vector that results from sending the code vector x over a noisy channel. We express the vector y as the sum of the original code vector x and a vector e , as shown by

$$y = x + e \quad (2.2.11)$$

The vector e is called the error vector or error pattern.

2.2.2 Minimum Distance Considerations

Consider pair of code vector x and y that has the same number of elements, the Harming distance $d(x, y)$ between such a pair of code vectors is defined as the number of locations in which their respective elements differ.

The Harming weight $w(x)$ of a code vector x is defined as the number of nonzero elements in the code vector.

The minimum distance d_{\min} of a linear block code is defined as the smallest Harming distance between any pair of code vectors in the code.

We may state that the minimum distance of a linear block code is the smallest Harming weight of the nonzero code vectors in the code.

2.3 Cyclic Codes

Cyclic codes form a subclass of linear block codes. An advantage of cyclic codes over most other types of codes is that they are easy to encode.

A binary code is said to be a cyclic code if exhibits two fundamental properties.

1. Linear property: The sum of two words is also a code word.
2. Cyclic property: Any cyclic shift of a code word is also word.

Property 1 restates the fact that a cyclic code is a linear block code. To restate Property 2 in mathematical terms, let the n -tuple $(x_0, x_1, \dots, x_{n-1})$ denote a code word of an (n, k) linear block code. The code is a cyclic code if the n -tuple.

$$(x_{n-1}, x_0, \dots, x_{n-2}),$$

$$(x_{n-1}, x_{n-1}, \dots, x_{n-3}),$$

$$(x_1, x_2, \dots, x_0),$$

Are 11 code words.

The code word with elements x_0, x_1, \dots, x_{n-1} may be represented in the form of a code word polynomial as follows:

$$x(D) = x_0 + x_1 D + \dots + x_{n-1} D^{n-1} \quad (2.3.1)$$

$$x(D) = x_0 + x_1 D + \dots + x_{n-1} D^{n-1} \quad (2.3.2)$$

Where D is an arbitrary real variable. Naturally, for binary codes, the coefficients are 1s or 0s. Each power D in the polynomial $x(D)$ represents a one-bit cyclic shift in time. Hence, multiplication of the polynomial $x(D)$ by D may be viewed as a cyclic shift or rotation to the right, subject to the constraint $D^n = 1$.

For a single cyclic shift, we may thus write

$$Dx(D) \bmod (D^n - 1) = x_{n-1} + x_0 D + \dots + x_{n-2} D^{n-1} \quad (2.3.3)$$

Where mod is the abbreviation for "modulo"

For two cyclic shifts, we may write

$$D^2 x(D) \bmod (D^n - 1) = x_{n-2} + x_{n-1} D + \dots + x_{n-3} D^{n-1} \quad (2.3.4)$$

This is a polynomial representation of the code word

$$(x_{n-2}, x_{n-1}, \dots, x_{n-3})$$

2.3.1 Encoder for Cyclic Codes

Earlier we showed that the encoding procedure for an (n, k) cyclic code. These three steps can be implemented by means of the encoder shown in Figure 2.2 Consisting of a linear feedback shift register with $(n-k)$ stages.

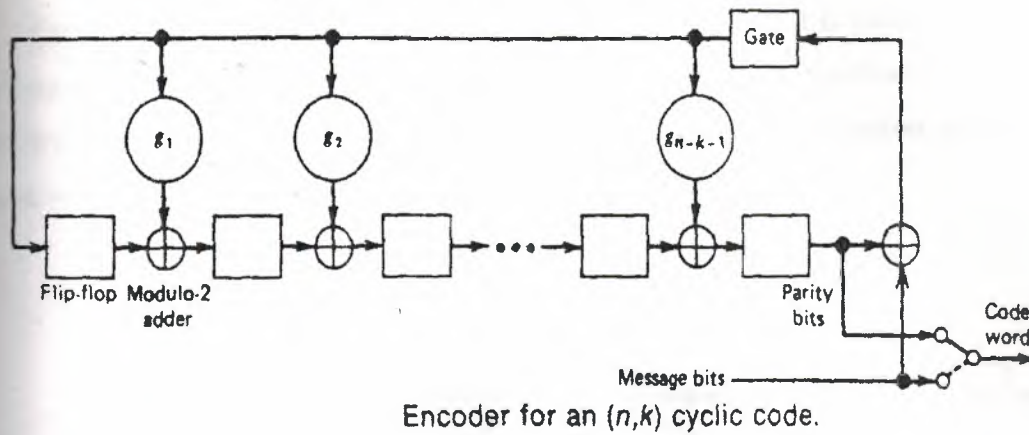


Figure 2.2 *Encoder for Cyclic Codes*

The operation of the encoder shown proceeds as follows:

- 1- The gate is switched on. Hence, the k message bits are shifted into the channel. As soon as the k message bits have entered the shift register, the resulting $(n-k)$ bits in the register form the parity bits (recall that the parity bits are the same as the coefficients of the remainder $b(D)$).
- 2- The gate is switched off, thereby breaking the feedback connections.
- 3- The contents of the shift register are shifted out into the channel.

Calculation of the Syndrome: Suppose the code word $(x_0, x_1, \dots, x_{n-1})$ is transmitted over a noisy channel resulting in the received word $(y_0, y_1, \dots, y_{n-1})$.

Let the received word be represented by a polynomial of degree $n-1$ or less, as shown by:

$$y(D) = y_0 + y_1 D + \dots + y_{n-1} D^{n-1} \quad (2.3.5)$$

Let $a(D)$ denote the quotient and $s(D)$ denote the remainder, which are the results of dividing $y(D)$ by the generator polynomial $g(D)$. Therefore

$$y(D) = a(D)g(D) + s(D) \quad (2.3.6)$$

The remainder $s(D)$ is a polynomial of degree $n-k$ or less. It is called the syndrome polynomial in that its coefficients make up the $(n-k)$ -by-1 syndrome s . When the syndrome polynomial $s(D)$ is nonzero, the presence of transmission errors in the received word is detected.

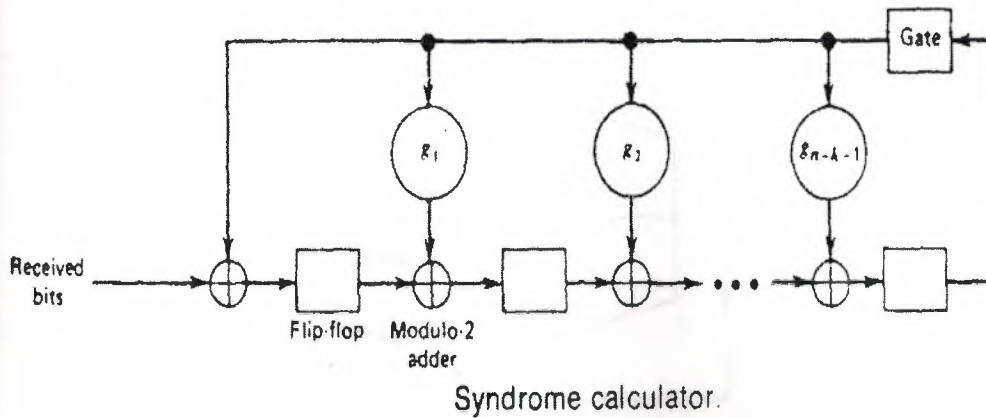


Figure 2.3 Syndrome calculator

The figure 2.3 shows a syndrome calculator that is identical to the encoder except for the fact that the received bits are fed into the $(n-k)$ stages of the feedback shift register from the left.

2.4 Convolutional Codes

There are applications where the message bits come in serially. In such situations, the use of convolution coding may be the preferred method. A convolution encoder operates on the incoming message sequence continuously in a serial manner.

The encoder of a binary convolution code with rate $1/n$, measured in bits per symbol, may be viewed as a finite-state machine that consists of an M -stage shift register with prescribed connections to n modulo-2 adders, and a multiplexer that serializes the outputs of the adders. An L -bit message sequence produces a coded output sequence of length $n(L+M)$ bits. The code rate is therefore given by

$$r = \frac{L}{n(L+M)} \text{ Bits / symbol} \quad (2.3.7)$$

Typically, we have $L > M$. Hence the code rate simplifies as

$$r = \frac{1}{n} \text{ Bit/symbol} \quad (2.3.8)$$

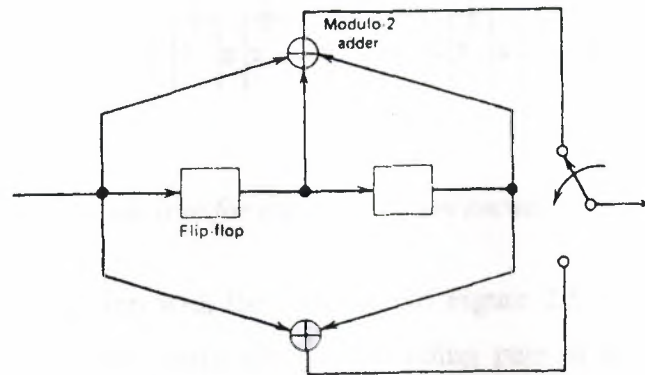


Figure 2.4 convolution encoder

The constraint length of a convolution code, expressed in terms of message is defined as the number of shifts over which a single message bit can influence the encoder output $K=M+1$, the constraint length of the encoder is Figure 2.4 shows a convolution encoder with $n=2$ and $k=3$. Hence the code rate of encoder $= 1/2$. The encoder operates on the incoming message sequence, one bit at a time.

2.5 Code Tree, Trellis, and State Diagram

Traditionally, the structural properties of a convolution encoder are portrayed in graphical form by using any one of three equivalent diagrams:

Code, tree, trellis, and state diagram.

We will use the convolution encoder of Figure 2.4

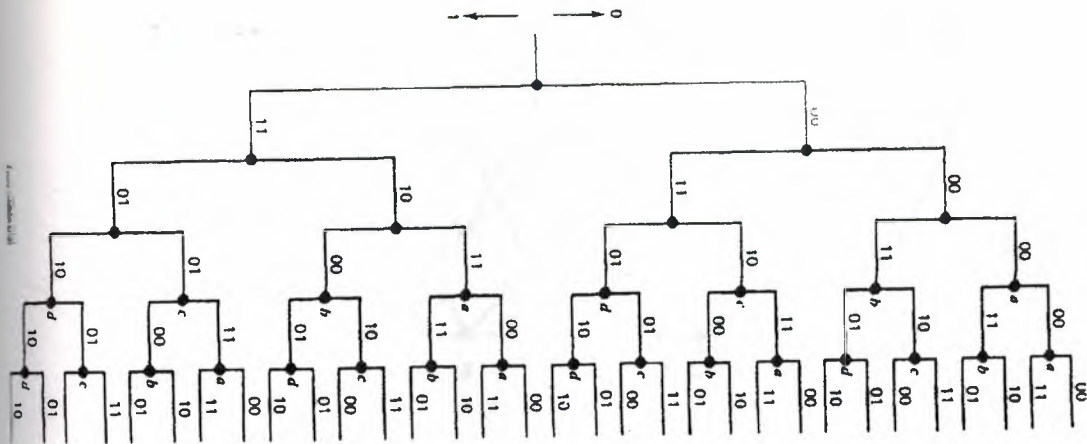


Figure 2.5 Code tree for the convolution encoder of Figure 2.4

We begin the discussion with the code tree of Figure 2.5. Each branch of the tree represents an input symbol, with the corresponding pair of output binary symbols indicated on the branch. The convention used to distinguish the input binary symbols 0 and 1 is as follows. An input 0 specifies the upper branch of a bifurcation, while input 1 specifies the lower branch. A specific path in the tree is traced from left to right in accordance with input (message) sequence. The corresponding coded symbols on the branches of that path constitute the sequence supplied by the encoder to the discrete channel input.

From diagram of Figure 2.5 we observe that the tree becomes repetitive after the first branches.

We may collapse the code tree of Figure 2.5 into the new form shown in Figure 2.6, called a trellis. It is so called since a trellis is a tree-like structure with remerging branches. The convention used in Figure 2.5 to distinguish between input symbols 0 and 1 is as follows. A code branch produced by an input 0 is drawn as a solid line, while a code branch produced by an input 1 is drawn as a dashed line. As before each input (message) sequence corresponds to a specific path through the trellis.

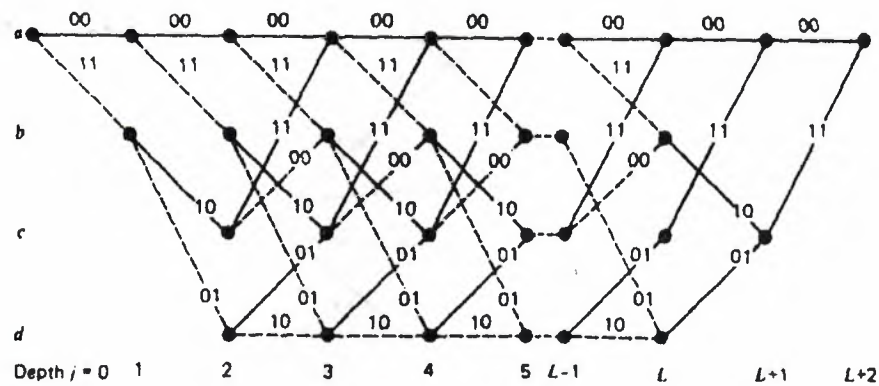


Figure 2.6 Trellis for the convolution encoder

A trellis is more instructive than a tree in that it brings out explicitly the fact that the associated convolution encoder is a finite-state machine. We define the state of a convolution encoder of rate $1/n$ as the most recent $(K-1)$ message bits moved into the encoder's shift register.

In case of the simple convolution encoder of Figure 2.4, we have $(K-1) = 2$. Hence, the state of this encoder, can assume any one of four possible values, as described in Table 2.1 the trellis contains $(L + K)$ levels, where L is the length of the incoming message sequence, and K is the constraint length of the code.

state	Binary description
a	00
b	10
c	01
d	11

Table 2.1 state table for the convolution Encoder

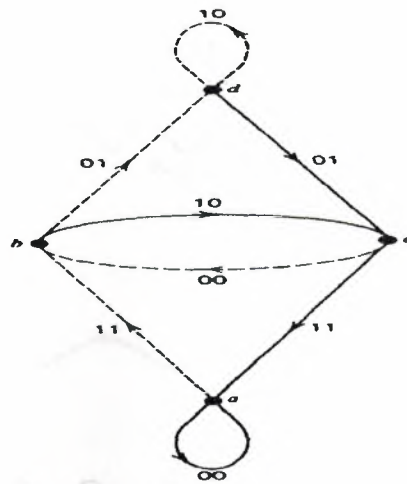


Figure 2.7 *state diagram of the convolution encoder*

We follow a solid branch if the input is a "0", and a dashed branch if it is a "1".

Thus, the input relation of a convolution encoder is completely described by its state diagram.

2.6 The Communications Channel

All communications systems and methods require a channel. This is because sending a message from one point to another involves the transmission of energy. All communications depend on the transfer of energy. The energy may be in various forms, such as light, electromagnetic waves, heat, sound, or mechanical motion. The channel is the path, or conduit, for this energy.

The term "channel" as used in the communications industry includes both the path energy and the path for the energy, but it may also encompass other aspects of the overall link.

A channel may carry signal, multiple signals in the same direction, or multiple signals in opposite directions.

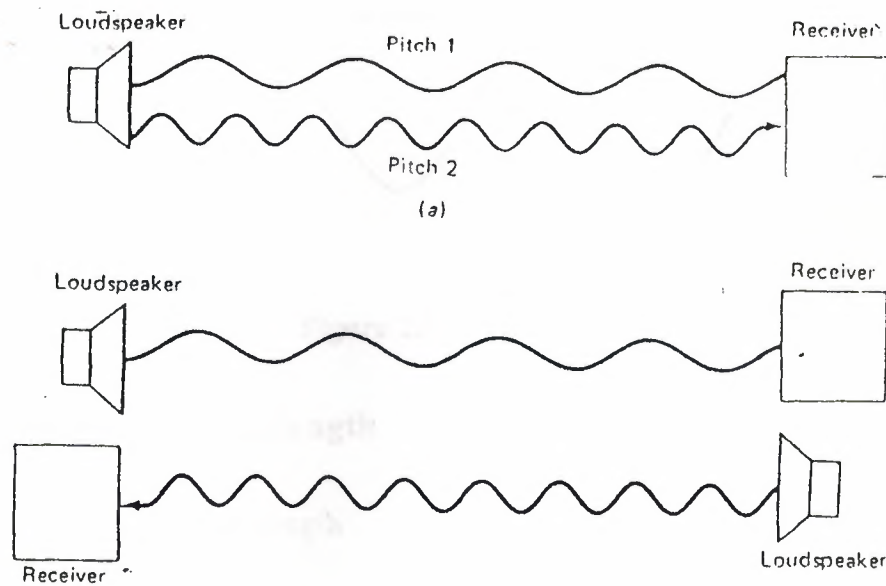


Figure 2.8 Different tones allow the same channel path to carry two messages at the same time in either (a) the same direction or (b) opposite directions. The different tones do not interfere with each other, even over the identical path.

2.7 Electromagnetic Waves

Electromagnetic waves carry energy via the electric field and magnetic field that form the wave. From a physics perspective, the energy can be thought of both as a wave and as particles, or bundles, of energy called photons.

A single equation describes the most important property of electromagnetic waves, which is the relationship of the frequency, wavelength, and velocity of the wave.

$$\frac{\text{Velocity}}{\text{Wavelength}} = \text{frequency} \quad (2.7.1)$$

The wavelength is the distance between successive crests of the wave (Figure 2.9). In a vacuum, such as in space, the value of velocity is 3×10^8 meters/second.

The definition of “wavelength” is the distance between the same relative point on successive cycles, such as the crest or valley.

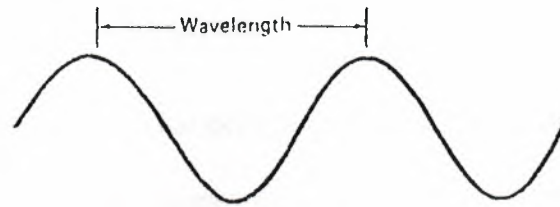


Figure 2.9 *Wavelength*

2.8 Frequency and Wavelength

$$\text{Velocity} = \text{Frequency} \times \text{wavelength} \quad (2.8.1)$$

Therefore, in a given channel, as the frequency goes up the wavelength goes down. Frequency is measured in cycles per second, or hertz (Hz).

The range of frequencies and wavelengths used for communications is enormous. Frequencies from 10Hz through several hundred billion hertz are used, depending on various requirements of the channel. The corresponding wavelengths, in a vacuum, would be 30 million meters to less than centimeters. The total range of frequencies that can be used is called the electromagnetic spectrum. The spectrum has been divided into many groupings, or bands and different bands are assigned for different uses. If the electromagnetic wave is traveling through the air or space, having many users within the same bands can cause interference with each other. An international commission meets to decide and assign which frequencies should be used by various countries and operations. For example, the range of frequencies from 540 to 1600 kilohertz (KHz) is assigned to the regular amplitude - modulated (AM) broadcast into band of each country.

2.9 The Electromagnetic Spectrum

The electromagnetic spectrum has been divided into general bands, for convenience.

Frequency Band	Name
3-10 KHz	Extremely low frequency (ELF)
10-30 KHz	Very low frequency
30-3000 KHz	Low frequency (LF)
300-300 KHz	Medium frequency (MF)
3-30 MHz	High frequency (HF) also called "short wave"
30-3000 MHz	Very high frequency (VHF)
300-300 MHz	Ultra high frequency (UHF) also called "microwaves"
3-30 GHz	Super high frequency (SHF)

Table 2.2 the Electromagnetic Spectrum

The spectrum of visible light is at even higher frequencies than the SHF band. Visible light has frequencies from 4300 to 7500 GHz. Light can be used for communications, but because of the extraordinarily high frequencies, systems using light must employ a completely different set of design of design schemes, even though light is an electromagnetic wave.

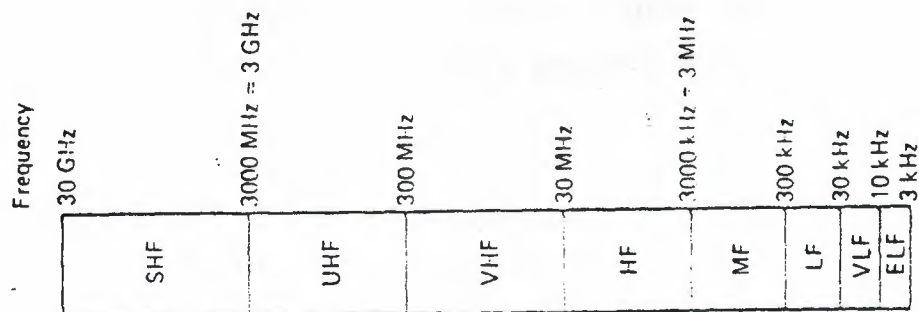


Figure 2.10 The major divisions of the electromagnetic spectrum (not shown to scale).

All the frequencies in the electromagnetic spectrum follow the same basic laws of physics. However, because of additional practical and real-world considerations, such as water vapor in the air, the energy of waves, their ability to penetrate solid objects, and the way they bounce and reflect the performance of communication channel is greatly affected by the frequency which is used.

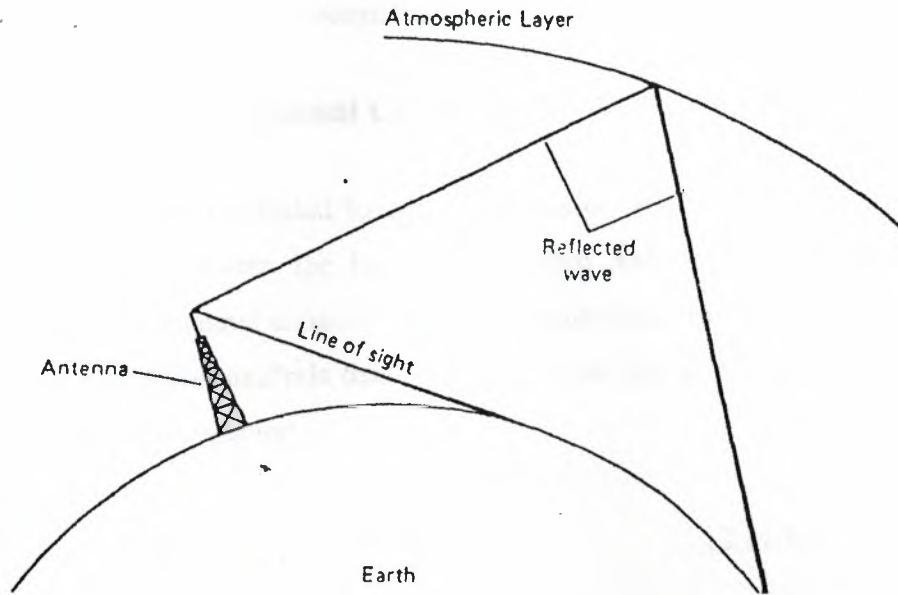


Figure 2.11 Transmitted signals can travel by direct line of sight or by reflection from layers of the atmosphere.

Noise is an undesired electrical signal that is superimposed on the desired signal. The atmosphere of the earth and the vacuum of space may other sources of electromagnetic signals. The ones deliberately generated by the transmitter for the channel.

2.10 Bandwidth

Bandwidth is an extremely important concept in data communications. The communications channel must have sufficient bandwidth to handle the amount of data information that must be passed over it. If the bandwidth of the channel is too low, the rate of data transfer may be less than required. If the channel is to handle more than one signal, then the bandwidth of the channel must be equal to the sum of the bandwidths of

each signal. Bandwidth is a simple case of "you can't get something for nothing". The price paid for transmitting data at the desired rate is the bandwidth needed.

Some typical examples of bandwidth will illustrate the relationship between bandwidth and information rate. A voice signal, transmitted over the telephone, uses a bandwidth of 3 KHz. A standard TV channel uses 6-MHz bandwidth, by contrast, of which 4.3 MHz is for the video information.

2.11 Bandwidth and Channel Capacity

A wider bandwidth is needed to carry information at a higher rate. What is the specific relationship between the bandwidth needed and the data rate that can be achieved (called the channel capacity) with that bandwidth? In 1984, Claude Shannon showed by mathematical analysis that there was a specific MHz, simple formula that related bandwidth and capacity:

$$\text{Capacity} = \text{bandwidth} \times \log_2 \left(1 + \frac{\text{signalpower}}{\text{noise power}} \right) \quad (2.11.1)$$

Where the capacity is measured in bits/second (bits/s), bandwidth in hertz and signal and noise powers must be in the same units.

Note: \log_2 is log to the base 2, and for any number X

$$\log_2(x) = \frac{\log_{10}(x)}{\log_{10}(2)} = \frac{\log_{10}(x)}{0.3} \quad (2.11.2)$$

3. SPREAD SPECTRUM TECHNIQUES

3.1 General Concepts

The discussions of communication systems in previous chapter have been concerned with the efficiency with these systems utilize signal energy and bandwidth.

These are situations, however, in which it is necessary for the system to resist external interference, to operate with a low-energy or to make it difficult for unauthorized receivers to observe the message. In such a situation, it may be appropriate to sacrifice the efficiency aspects of the system in order to enhance these other features. Spread – spectrum techniques offer one way to accomplish this objective.

The use of spread-spectrum techniques originated in answer to the unique needs of military communications, and it is reasonable to assume that these techniques will soon penetrate the civilian sector. Therefore, a discussion of modern communications would not be complete without a look at the fundamentals and the applications of spread spectrum.

For a communication system to be considered a spread-spectrum system, it is necessary that the transmitted signal satisfy two criteria. First, the bandwidth of the transmitted signal must be much greater than the message bandwidth.

This by itself, however, is not sufficient because there are many modulation methods that achieve it. For example, frequency modulation, pulse code modulation, and delta modulation may have bandwidths that are much greater than the message bandwidth. Hence the second criterion is that the transmitted bandwidth must be determined by some function that is independent of the message and is known to the receiver.

Since the spread-spectrum system is not useful in combating white noise, it must have other applications that make it worth considering. These applications include:

- 1- Antijam capability - particularly for narrow-band jamming.
- 2- Interference rejection.

- 3- Multiple-access capability.
- 4- Multipath protection.
- 5- Covert operation or low probability of intercept (LPI).
- 6- Secure communications.
- 7- Improved spectral efficiency - in special circumstances.
- 8- Ranging.

There are many different types of spread-spectrum systems and one way of classifying them is by concept. On this basis spread-spectrum systems may be considered to be either averaging systems or avoidance systems. An averaging system is one in which the reduction of interference take place because the interference can be averaged over a large time interval. An avoidance system, on the other hand, is one in which the reduction of interference occurs because the signal is made to avoid the *interference a large fraction of the time.*

A second method of classifying spread-spectrum systems is by modulation. The most common modulation techniques employed are the following.

- 1- Direct sequence (pseudonyms)
- 2- Frequency hopping
- 3- Time hopping
- 4- Chirp
- 5- Hybrid methods

The relation between these two methods of classification may be made clearer by noting that a direct -sequence system is an averaging system, whereas frequency-hopping, time hopping and chirp systems are avoidance systems. On the other hand, a hybrid modulation method may be either averaging or avoidance, or both.

3.2 Direct Sequence (DS) or PseudoNoise (PN)

The terms direct sequence and pseudnoise are used interchangeably here and no distinction is made between them. A typical direct-sequence transmitter is illustrated in Figure 3.1. Note that it contains a PN code generator that generates the pseudonoise sequence. The binary output of this code generator is added, modulo 2, to the binary message, and the sum is then used to modulate a carrier. The modulation in this case is diphase or phase reversal modulation so that the output is simply a phase shift keyed signal. The PN code is generated in a maximal length shift register such as shown in Figure 3.2.

Pseudnoise code generators are periodic in that sequence that is produced repeats itself after some period of time. Such a periodic sequence is portrayed in Figure 3.3. The smallest time increment in the sequence is of duration t_1 , and is known as a time chip. The total period consists of N time chips.

When the code is generated by maximal linear PN code generator, the value of N is $2^n - 1$, where n is the number of stages in the code generator. An important reason for using shift register codes is that they have very desirable autocorrelation properties.

The autocorrelation function of a typical PN sequence is shown in Figure 3.4. Note that on a normalized basis, it has a maximum value of one that repeats itself every period, but in between these peaks, the level is at a constant value of $-(1/N)$. If N is a very large number, the autocorrelation function will be very small in this region.

Another reason for using shift register codes is that the period of the PN sequence can easily be made very

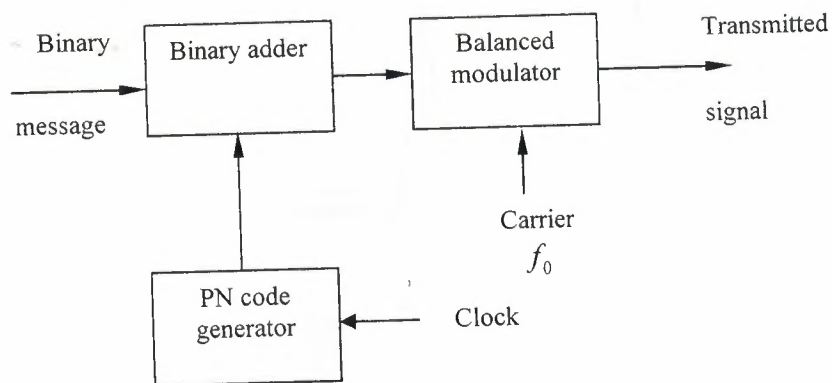


Figure 3.1 Direct-sequence transmitters

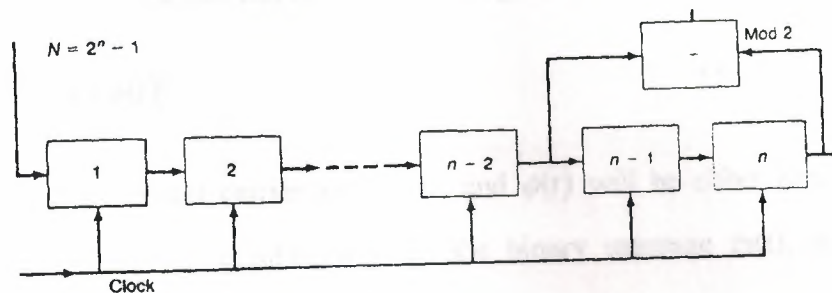


Figure 3.2 maximal linear PN code generators

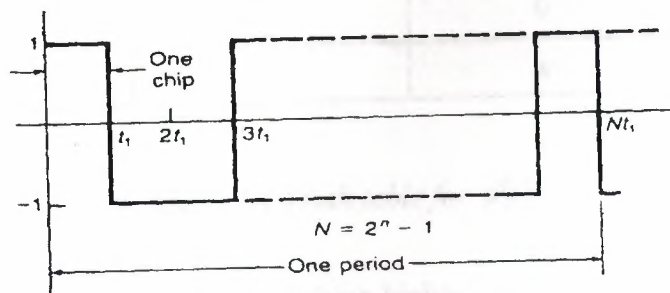


Figure 3.3 Periodic binary PN sequence

The modulation of the PN sequence on the spread-spectrum carrier can be either biphase or quardriphase. It is of interest to consider both of these methods.

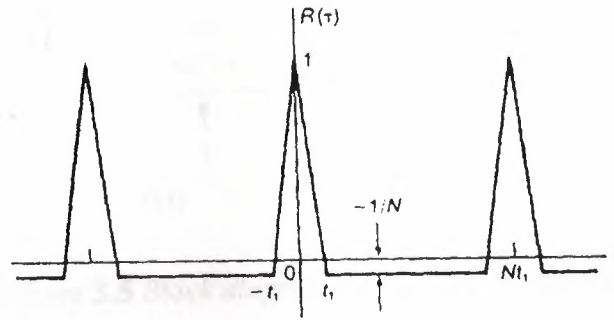


Figure 3.4 Autocorrelation function of PN sequence

3.3 Biphase modulation

A phase-modulation carrier can be expressed in general as

$$s(t) = A \sin[\omega_0 t + \phi(t)] \quad (3.3.1)$$

Where A is the constant carrier amplitude, and $\phi(t)$ will be either zero or π . The values of $\phi(t)$ for various combinations of the binary message $m(t)$, and the PN sequence, $b(t)$, are shown in Table 3.1.

		$m(t)$	
		1	-1
$b(t)$	1	0	π
	-1	π	0

Table 3.1 Truth table for $\phi(t)$

A block diagram of a system accomplishing biphase modulation is shown in Figure 3.5. This system employs a balanced modulator that ideally produces the desired phase shift keying without any residual carrier at the output. It is necessary that the message bit duration t_m be an integral multiple of the chip duration t_1 as shown in Figure 3.6.

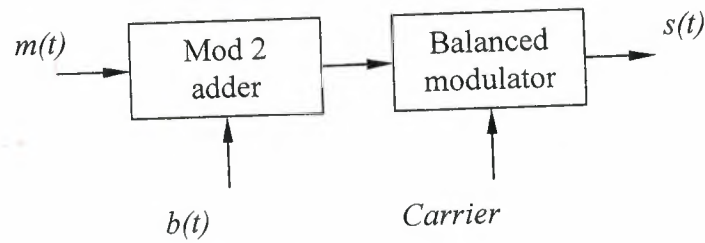


Figure 3.5 Block diagram for bi phase modulation

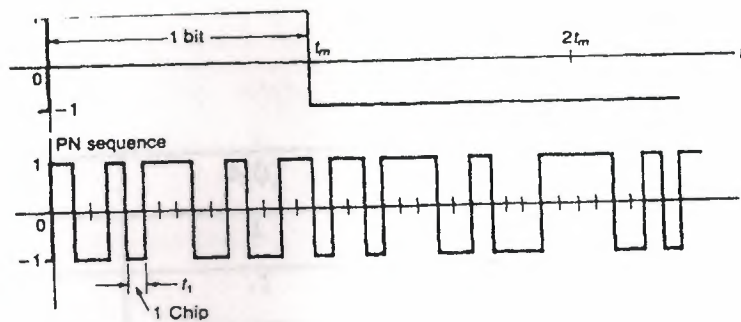


Figure 3.6 Relation between the code sequence & the binary message

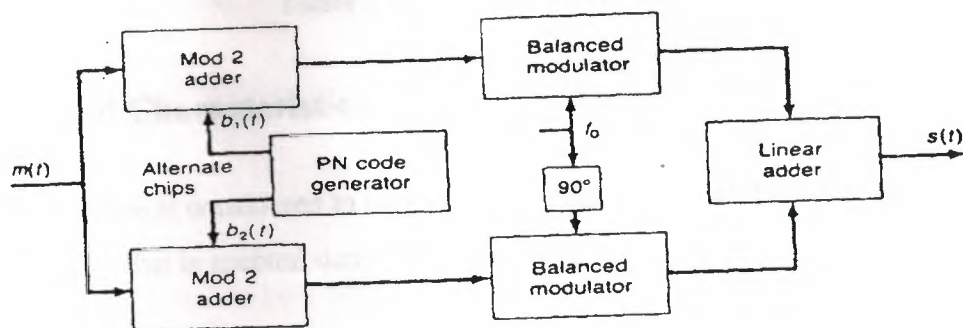


Figure 3.7 Block diagram for quadriphase modulation.

3.4 Quadriphase Modulation

A block diagram of a system producing quadriphase modulation is shown in Figure 3.7. In this case two balanced modulators are used and the carriers to these two modulators are 90 degrees apart in phase. There are also two modulo-2 adders that add the message binary sequence to the PN code sequence, using alternate chips from the code sequence to do so. This means that each chip of the PN code is stretched to

duration of $2t_1$ before being added to the binary message. The quadriphase signal can again be represented as

$$s(t) = A \sin[\omega_0 t + \phi(t)] \quad (3.4.1)$$

In which A is the carrier amplitude and $\phi(t)$ is the phase modulation. The relation of $\phi(t)$ to the state of the message and the states of the PN code sequence is shown in Table 3.2.

$b_1(t)$	$b_2(t)$	$m(t)$	$m(t)$
		1	-1
1	1	$\pi/4$	$5\pi/4$
1	-1	$7\pi/4$	$3\pi/4$
-1	1	$3\pi/4$	$7\pi/4$
-1	-1	$5\pi/4$	$\pi/4$

Table 3.2 Truth table of $\phi(t)$

3.5 PN Signal Characteristics

If PN sequence is considered to be purely random, rather than periodic, it is straightforward to show that its spectral density has the form

$$S(f) = \frac{t_1}{2} \left\{ \left[\frac{\sin \pi(f - f_0)t_1}{\pi(f - f_0)t_1} \right]^2 + \left[\frac{\sin \pi(f + f_0)t_1}{\pi(f + f_0)t_1} \right]^2 \right\} \quad (3.5.1)$$

In which the expression has been normalized to represent a signal having unit average power. This spectral density is displayed for positive frequencies in Figure 3.8. It is customary to define the bandwidth of a PN signal as the frequency increment between the two zeros of the spectral density that are closest to the center frequency. It is clear from Figure 3.8 that this signal bandwidth is $2/t_1$.

Since the message is also binary, it will have a similar spectral density but centered on zero. Thus the message spectral density is:

$$S_m(f) = t_m \left[\frac{\sin \pi f t_m}{\pi f t_m} \right]^2 \quad (3.5.2)$$

The bandwidth of the message B_m is simply $1/t_m$ because it is customary to use only the positive frequency portion of the spectrum in defining bandwidth.

An important parameter that is sometimes useful in specifying the performance of a spread-spectrum signal in the presence is known as processing gain, PG, is frequently defined as the ratio of the signal bandwidth to the message bandwidth. Thus:

Some authors define the processing gain as the ratio of the chip rate to the message bit rate.

3.6 Frequency Hopping

In a frequency-hopping signal, the frequency is constraint in each time chip, but changes from chip to chip. This type of signal is illustrated in Figure 3.9.

It is frequently convenient to categorize frequency-hopping systems as either "fast hop" or "slow hop".

A fast -hop system is usually considered to be one in which the frequency hopping takes place at a rate that is greater than the message bit rate; in a slow-hop system, the hop rate is less than the message bit rate. There is, of course, an intermediate situation in which the hop rate and the message bit rate are of the same order of magnitude.

$$PG = \frac{B_s}{B_m} = \frac{2t_m}{t_1} \quad (3.6.1)$$

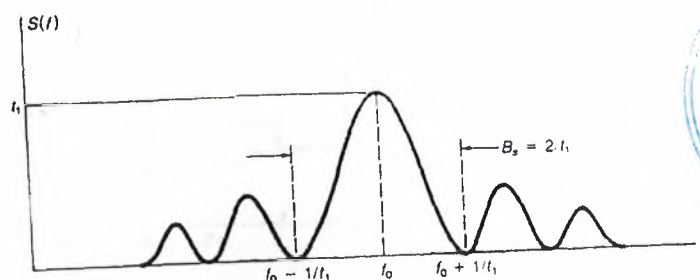


Figure 3.8 Spectral density of a random binary sequence.

For purposes of illustration, a fast-hop system is considered here in which there are k frequency hops in every message bit. Thus the chip duration is:

$$t_1 = t_m k \quad (3.6.2)$$

where $k=1, 2, 3, \dots$. The number of frequencies over which the signal may hop is usually a power of 2, although not all these frequencies are necessarily used in a given system.

3.6.1 The Frequency-Hopping Transmitter

The block diagram of a frequency-hopping transmitter is shown in Figure 3.10; the frequency hopping is accomplished by means of a digital frequency synthesizer. This in turn is driven by a PN code generator. The frequency synthesizer is controlled by m binary digits and produces one of $M = 2^m$ frequencies for each distinct combination of these digits. One of these m controlling digits comes from the message and the other $m-1$ digits come from the PN code generator. If the digit from the message produced the smallest frequency change, then by itself it would produce a binary FSK signal. The $m-1$ digits from the PN code generator then hop this FSK signal over the range of possible frequencies.

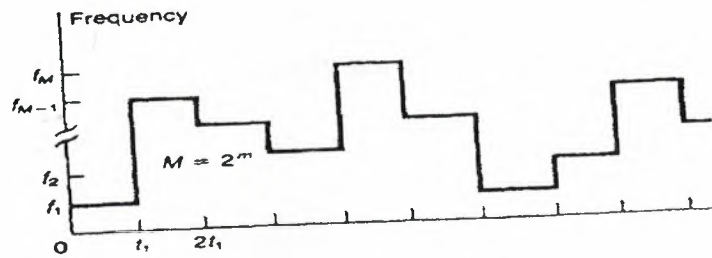


Figure 3.9 Frequency-hopping signals

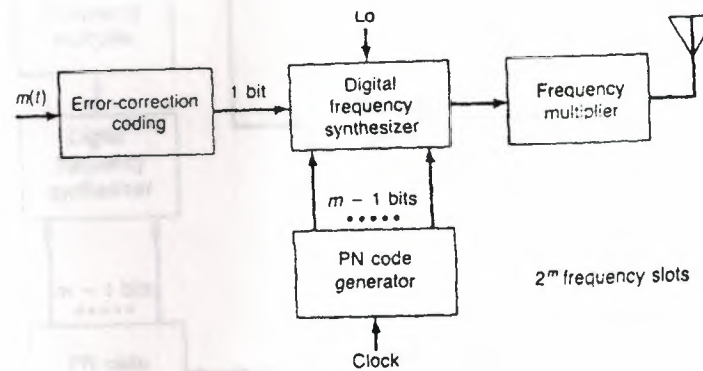


Figure 3.10 Frequency-hopping transmitters

The message, prior to modulating the frequency synthesizer, normally will have error-correction coding applied to it. If any one hop is interfered with, all of the bits in that particular hop may be destroyed, and therefore, it is necessary to be able to reconstruct the message by using error-correction techniques. It may also be noted that there is a frequency multiplier at the output of the system, to increase the bandwidth & PG. It also changes the shape of the spectrum.

3.6.2 The Frequency-Hopping Receiver

Usually the reception of a frequency-hopping signal is done on a noncoherent basis. Coherent reception is possible, but it is more difficult. A typical noncoherent, frequency-hopping receiver is shown in Figure 3.11. Note that this consists of a digital frequency synthesizer driven by a PN code generator and followed by frequency multiplier. This locally generated frequency-hop signal is multiplied by the incoming signal in a mixer, and if the two are in step, the result will be a normal binary FSK signal. Error correction is then applied to produce the eventual message. The output of

the mixer is also applied to early and late gates that produce an error signal to control the clock frequency. This keeps the locally generated frequency-hop signal in step with the incoming signal.

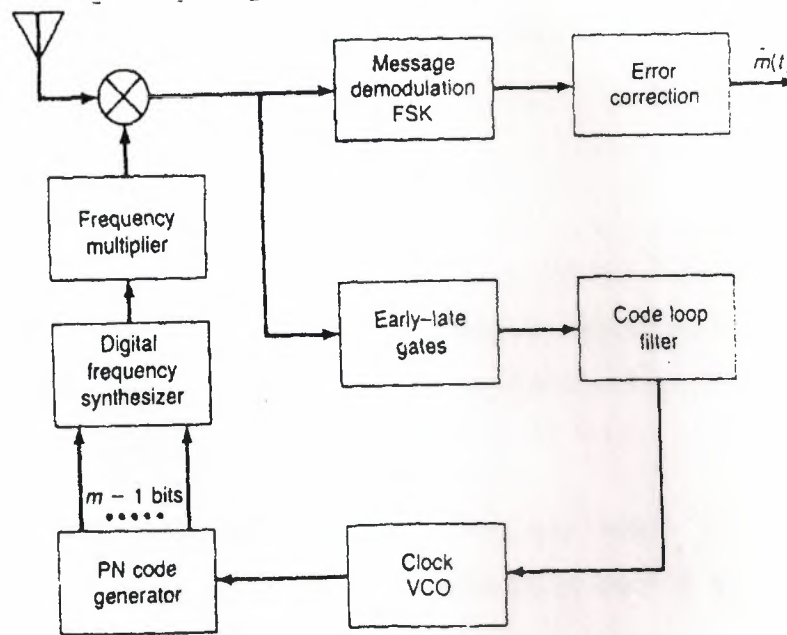


Figure 3.11 *Noncoherent Frequency - hopping receiver*

3.7 Hybrid Spread-Spectrum Systems

The use of a hybrid system attempts to capitalize upon the advantage of a particular method while avoiding the disadvantages. Many different hybrid combinations are possible. Some of these are:

PN/TH, FH/TH, PN/FH/TH

To illustrate how a hybrid system might operate, consider the case of a PN/FH hybrid system. This system might use a PN code to spread the signal to an extent limited by either code generator speed acquisition time. Then frequency hopping would be used to increase the frequency spread. The difference between the frequencies in the frequency-hopping portion of the system would normally be equal to the bandwidth of the PN code modulation. Usually some form of noncoherent message modulation is used because of the frequency hopping, and differential phase shift keying is a typical

example. Since there are fewer frequencies to be implemented, the frequency synthesizer is simpler for a given overall bandwidth. Thus this system gains some of the advantages of direct-sequence systems and of frequency-hop systems, and avoids some of the disadvantages of both.

4. INTRODUCTION TO CELLULAR MOBILE SYSTEMS

4.1 Limitations of Conventional mobile telephone systems

One of many reasons for developing a cellular mobile telephone system and deploying it in many cities is the operational limitations of conventional mobile telephone systems: limited service capability, poor service performance, and inefficient frequency spectrum utilization.

4.1.1 Spectrum efficiency considerations

A major problem facing the radio communication industry is the limitation of the available radio frequency spectrum. In setting allocation policy, the Federal Communications Commission (FCC) seeks systems which need minimal bandwidth but high usage and consumer satisfaction.

The ideal mobile telephone system would operate within a limited assigned frequency band and would serve an almost unlimited number of users in unlimited areas. Three major approaches to achieve the ideal are:

- 1- Single-sideband (SSB), which divides the allocated frequency band into maximum numbers of channels.
- 2- Cellular, which reuses the allocated frequency band in different geographic locations.
- 3- Spread spectrum, frequency-hopping, which generates many codes over a wide frequency band.

In 1971, the cellular approach was shown to be spectrally efficient system.

The FCC's decision to choose 800 MHz was made because of severe spectrum limitations at lower frequency bands.

4.2 Basic Cellular System

A basic Cellular system consists of three parts: a mobile unit, a cell site, and a mobile telephone switching office (MTSO) as Figure 4.1 shows - with connections to link the three subsystems.

1- Mobile units: A mobile telephone unit contains a control unit, a transceiver, and an antenna system.

2- Cell site: The cell site provides interface between the MTSO and the mobile units. It has a control unit, radio cabinets, antennas, a power plant, and data terminals.

3- MTSO: The switching office, the central coordinating element for all cell sites, contains the cellular processor and cellular switch. It interfaces with telephone company zone offices, controls call processing, and handles activities.

4- Connections: The radio and high-speed data links connect the three subsystems. Each mobile unit can only use one channel at a time for its communication link. But the channel is not fixed; it can be any one in the entire band assigned by the serving area, with each site having multi channel capabilities that can connect simultaneously too many mobile units.

The MTSO is the heart of the cellular mobile system. Its processor provides central coordination and cellular administration.

The cellular switch, which can be either analog or digital, switches calls to connect mobile subscribers to other mobile subscribers and to the nationwide telephone network. It uses voice trunks similar to telephone company interoffice voice trunks. It also contains data links providing supervision links between the processor. The radio link carries the voice and signaling between the mobile unit and the cell site. The high speed data links cannot be transmitted over the standard telephone trunks and therefore must use either microwave links or T-carriers (wire lines). Microwave radio links or T-carriers carry both voice and data between the cell site and the MTSO.

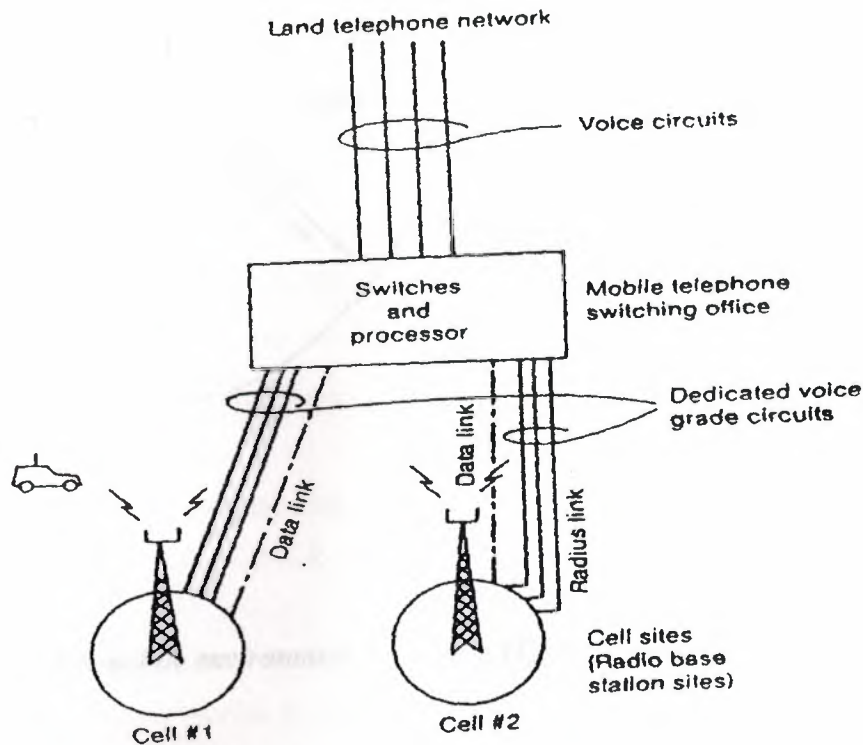


Figure 4.1 basic cellular systems

4.3 Mobile fading characteristics

Rayleigh fading is also called multipart fading in the mobile radio environment. When these multipart waves bounce back and forth due to the buildings and houses, they form many standing-wave pairs in space, as shown in Figure 4.2. Those standing-wave pairs are summed together and become an irregular wave-fading structure. When a mobile unit is standing still, its receiver only receives signal strength at that spot, so a constant signal is observed. When the mobile unit is moving, the fading structure of the wave in the space is received. It is a multipart fading. The recorded fading becomes fast as the vehicle moves faster.

4.4 Operation of Cellular Systems

This section briefly describes the operations of the cellular mobile system from a customer's perception without touching on the design parameters. The operation can be divided into four parts and a handoff procedure.

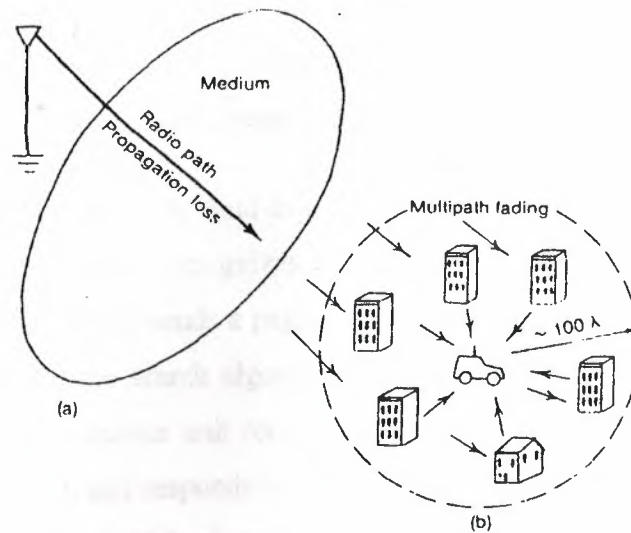


Figure 4.2 a mobile environment -two parts (1) Propagation loss (2) Multipart fading

Mobile unit initialization: when a user sitting in a car activates the receiver of the mobile unit, the receiver scans 21 set-up channels which are designated among the 333 channels. It then selects the strongest and locks on for a certain time. Since each site is assigned a different set-up channel, locking onto the strongest set-up channel usually means selecting the nearest cell site. This self-location scheme is used in the idle stage and is user- independent. It has a great advantage because it eliminates the load on the transmission at the cell site for locating the mobile unit. The disadvantage of the self-location scheme is that no location information of idle mobile units appears at each cell site. Therefore, when the call initiates from the land line to a mobile unit, the paging process is longer. Since a large percentage of calls originate at the mobile unit, the use of self-location schemes is justified. After 60s, the self -location procedure is repeated. In the future, when land-line originated calls increase, a feature called “registration” can be used.

Mobile originated call: The user places the called number into an originating register in the mobile unit, checks to see that the number is correct, and pushes the “send” button. A request for service is sent on a selected set-up channel obtained from a self-location scheme. The cell site receives it, and in directional cell sites, selects the best directive antenna for the voice channel to use. At the same time the cell site sends a request to the mobile telephone switching office (MTSO) via a high-speed data link.

The MTSO selects an appropriate voice channel for the cell, and the cell site acts on it through the best directive antenna to link the mobile unit. The MTSO also connects the wire-line part through the telephone company zone office.

Network originated call: A land-line party dials a mobile unit number. The telephone company zone office recognizes that the number is mobile and forwards the call to the MTSO. The MTSO sends a paging message to certain cell sites based on the mobile unit number and the search algorithm. Each cell site transmits the page on its own-set-up channel. The mobile unit recognizes its own identification on a strong set-up channel, locks onto it, and responds to the cell site. The mobile unit also follows the instruction to tune to an assigned voice channel and initiate user alert.

Call-termination: When the mobile user turns off the transmitter, a particular signal (signaling tone) transmits to the cell site, and both sides free the voice channel. The mobile unit resumes monitoring pages through the strongest set-up channel. The system switches the call to a new frequency channel in a new cell site without either interrupting the call or alerting the user. The call continues as long as the user is talking. The user does not notice the handoff occurrences.

Hand off Procedure: During the call, two parties are on a voice channel. When the mobile unit moves out of the coverage area of a particular cell site, the reception becomes weak. The present cell site requests a hand off; the system switches the call to a new frequency channel in a new cell site without either interrupting the call or alerting the user. The call continues as long as the user is talking. The user does not notice the hand off occurrences.

CONCLUSION

The concept of cellular systems is the use of low power transmitters in order to enable the efficient reuse of frequencies. In fact, if the transmitters which are used are very powerful, the frequencies can not be reused for hundreds of kilometers as they are limited to the covering area of the transmitter. So, in a cellular system, the covering area of an operator is divided into cells. A cell corresponds to the covering area of one transmitter or a small collection of transmitters. The size of a cell is determined by the traffic generated in the area and /or the time advanced.

The most unsatisfactory feature of the channel coding theorem, however, is the no constructive nature. The theorem only asserts the existence of good codes. The error-control coding techniques provide different methods of achieving this important system requirement. We consider block codes first, followed by convolution codes, and then trellis codes.

The term "channel" as used in the communications industry includes both the path energy and the path for the energy, but it may also encompass other aspects of the overall link. A channel may carry signal, multiple signals in the same direction, or multiple signals in opposite directions.

The use of spread-spectrum techniques originated in answer to the unique needs of military communications, and it is reasonable to assume that these techniques will soon penetrate the civilian sector. Therefore, a discussion of modern communications would not be complete without a look at the fundamentals and the applications of spread spectrum.

The MTSO is the heart of the cellular mobile system. Its processor provides central coordination and cellular administration.

REFERENCES

- [1] Simon Haykin, An Introduction to Analog and Digital Communications, John Wiley & Sons, Inc, 1989.
- [2] R.E Zierner & W.H Tranter, Principles of Communication System Modulation, and Noise, Houghten Muffin Company, 1990.
- [3] Leon W. Couch II, Digital and Analog Communication Systems Macmillan Publishing Company 1993.
- [4] K. Sam Shanmugan, Digital and Analog Communication Systems, John Wiley & Sons, Inc Copyright 1985.
- [5] Jerry D. Gibson, Principles of Digital and Analog Communications, Macmillan Publishing Company, 1987.
- [6] George R.Cooper, Clare D.Mc Gillem, Modern Communications and spread spectrum, Mc Graw - Hill, 1986.
- [7] William C.Y.Lee, Mobile Cellular Telecommunications Systems, Mc Graw - Hill copyright 1989.
- [8] Kim, D.K. and Sung, D.K., 1999, Characterized of Soft Handoff in CDMA Systems, IEEE Trans. Vech. Technol, 48(4): 1195-1202.