NEAR EAST UNIVERSITY

Faculty of Engineering

Department of Electrical and Electronic Engineering

ACCESS TECHNOLOGIES IN WIRELESS COMMUNICATION

Graduation Project
EE- 400

Student : Nabil Ahmed Siddique (981227)

Supervisor : Prof. Dr. Fakhreddin Mamedov

Nicosia - 2003

ACKNOWLEDGMENT

I wish to thank my parents who supported and encouraged me at every stage of my education and who will being generous for me as they are ever.

My sincere thanks and appreciation for my supervisor Prof. Dr. Fakhreddin Mamedov upon his attitude towards my project and who was generous with his help, valuable advices and comments to accomplish this research and who will be always my respectful teacher. He provides me a great knowledge about the communication field.

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ABSTRACT

Communication is an important part of our lives, because we are almost always involved in some form of it. In order to send a message from one point to another, three system components must be present. We need a source which generates a message and places it on, a transmission medium which carries the message, to the receiver.

These elements are the minimum requirements in any communication process, and the absence of any one of them makes the communication process incomplete, and in many cases useless. For communication to be effective, the message must be understood. In certain types of communication, information flows only in one direction, for example, radio/ TV stations and receivers at home.

Classification of communication modes helps us describe a particular communication system. How ever, certain concepts must be understood for one to discern the operation and use of modern communication systems. Analog and digital messages, transmission of messages, encoding of messages, routing and switching of messages, the network of highways and road ways that message travel over, and regulation of the flow of messages within that network are a few of the themes that r necessary to an understanding of a particulars of a communication systems.

INTRODUCTION

In this project the access technologies in wireless communication is studied with intensive care. Wireless communication is a new technology, which will have a large impact on telecommunications as well as in the wireless networking, WLANs (Wireless local area network) and in the satellite communications. The project consists of four chapters.

Chapter 1 gives a briefly introduction to wireless communications. In this chapter wireless communication is explained with special emphasis on cellular telephony. It also explained the generation system FDMA, TDMA and CDMA.

Chapter 2 explained the first generation of the communication system, FDMA. It also gives a short introduction to AMPS, which is based on frequency division multiplexing.

Chapter 3 gives a basic explanation about the Time division multiple access and time division multiplexing. It also discussed about the TDMA technical details. The TDMA system is assigned to the same set of frequencies as AMPS ; however, because of time multiplexing into slots, each carrier handles three physical channels. Thus the TDMA, has three times the capacity of AMPS.

Chapter 4 covers the discussion about the code division CDMA by considering the historical development of the spread sectrum. It also gives a short introduction about the concept of spread spectrum and spread spectrum techniques ; frequency hopping and binary spreading sequence.

TABLE OF CONTENTS

CHAPTER 1

INTRODUCTION TO WIRELESS COMMUNICATION

1.1 Introduction

No area of telecommunications has experienced the recent explosive growth like that of wireless communications. Of course, wireless communications is essentially radio communications, which is not all that new. However, the maturation of various technologies (such as very large scale integration of circuitry and advances in signal processing), cost reductions in technologies, new frequency availability, and business pressures for mobility and networking in the past few years have finally made wireless communications efficient and cost effective.

Wireless telecommunications is, unfortunately, a simple and all encompassing phrase. As the name implies, wireless telecommunications means communicating without the use of wires or other physical guides or conduits (such as fiber) through some distance. There are numerous examples of wireless communications, including remote controls for TVs, stereos, garage door openers, pagers cordless phones, and cellular phones to name a few.

There are several advantages to wireless communications, with the most significant being that the callers need not be tied to a particular location; that is, wo can have mobile terminals. Also, the medium, unlike wires, fiber or coaxial cable/ is simply air (for satellite communications it is space) and is free. The primary disadvantage is that, because of the possibility of interference among users, the electromagnetic spectrum must be subdivided, referred to as frequency allocation.

Many topics associated with wireless are the same as with other areas of telecommunications, but some functions, such as power and bandwidth, have a different significance. For instance, we wish to minimize power consumption in the mobile units because they are usually run off batteries. We also wish to minimize the broadcast power of the base station to reduce the effect of interference with other base stations. Bandwidth has one set of problems with terrestrial communications, for instance in a twisted pair, and a different set of problems with cellular technology where air is the transmission medium. One interesting problem that has to be handled by a cellular phone system is locations of the communicating parties; that is, the calling party must identify itself and then the system must find the party being called.

In this chapter we intend to explain the technology associated with the major form of wireless communications: cellular telephony.

1.2 Pagers and GPS Services

Pagers and paging have been around for a number of years. Paging is a comparatively simple and inexpensive means of communications. 1t is a classic example of simplex transmission in that it is entirely one way without any response from the receiver; however, two-way paging is likely to be the next wave. There are narrow and wide area pagers; some simple pagers operate only within a building, others in a radius of a couple miles, and still others can provide worldwide coverage. The concept of a pager is straightforward: broadcast a message into the air via a transmitter or transmitters and assume that the correct receiver will get the message. Pagers typically require large transmitters (high power) and many transmitters (in wide area paging systems) over which the message is simulcast. Also/ since the messages are typically short, the data rates need n ot be high.

A similar system to paging is position-locating systems, which also require only a receiver. The modern position-locating systems rely on the global positioning system (GPS) satellites that are in medium earth orbit. A GPS receiver will receive signals from multiple satellites and, from the received data, calculate the location within a radius of a few meters. These systems have revolutionized navigation for ships and airplanes and they are now available in auto mobiles.

1.3 Introduction to Cellular Telephone

The fastest growing and most commonly practiced and recognized version of wireless telecommunications is the cellular phone. The cellular concept is remarkable in its apparent simplicity but is rather complicated to implement because of the vast number of calls to be serviced. Think of a large flat plane divided into hexagonal cells as shown in Figure 1-1. Assume that a base station exists at the center of each cell. Mobile units within a particular cell are served

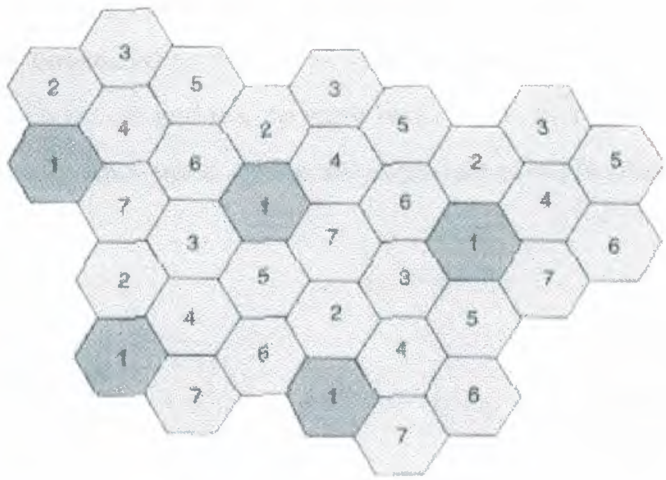


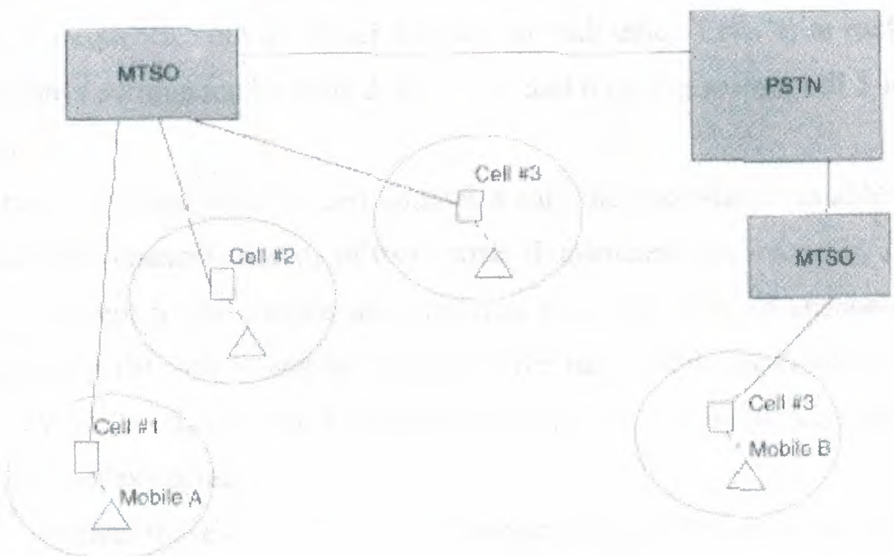
Figure 1.1 The cellular concept

by that cell's base station, and if a mobile unit moves from one cell to another a handoff occurs (to the new cell's base station).

If each cell had its own unique set of frequencies, we would quickly use up the available bandwidth, but that is not necessary. If the power is adjusted properly and the spacing is appropriate, we can have sets of cells with the same frequencies, with virtually no interference, allowing for frequency reuse. Notice the groups of shaded cells in Figure 1-1. They constitute the set of frequencies used for this system. Within the group of seven cells the frequencies are different, but then these seven cells are reused over and over again. This concept allows the manufacturers of standard telephones (mobile units) to use the same set of channels. These mobiles can then be used throughout the country. Not all systems use sets of seven cells; additionally, some do not reuse frequencies at all.

If we were dealing with devices such as walkie-talkies, when you talked on one cellular phone the signal would go through the air and be received directly by your friend on another cell phone. But of course that is not practical, especially if you are any distance away from each other and other people are using walkie-talkies in the vicinity. The other people using walkie-talkies would have to have their own frequencies to avoid interference with your conversation—a totally impractical situation. Therefore, each cell phone connects to a base station that makes the connection to another cell phone or a regular phone in the public switched telephone network (PSTN).

The base stations are connected to switches which are connected to other switches, which are ultimately connected to the PSTN. How are these connections made? There is no need to do this through the air and use up more band width, so typically it is done with land lines, which are usually fiber optic cables although line-of-sight microwave links are also often used.



MTSO = Mobile Telephone Switching Office

□ = Base Station for Cell

Δ = Mobile within a Particular

Figure 1.2 The architecture of a cellular phone connection

Figure 1-2 illustrates the interconnection of two cellular phones through base stations, switches, and the PSTN. We will discuss the details of how this phone call is set up shortly, but for now the important point is what the path looks like. Notice that there is no direct connection between two wireless phones through the air. Even if they happened to be in the same cell the call would be interconnected through at least a switch and a channel set up for each mobile unit,

1.3.1 Cellular Operation

The cellular concept is straightforward and ingenious. The terrain is divided into cells represented as hexagons or circles (hexagons look neater, but ragged-looking circles are closer to reality), as shown in Figure 1-1. In most systems, seven sets of cells (reuse factor of 7) are repeated. Within the pattern of seven cells, each cell has a unique set of frequencies to use, thereby avoiding interference among cells. There are no adjacent cells where the same frequencies are used. Notice the shaded cells (number 1), which have the same set of frequencies but are never adjacent to each other. Cell "1, in each set of seven cells, is always surrounded by cells 2, 4, 7, 5 3, and 6 (going around cell 1 in a clockwise direction).

When a terminal within a cell initiates a call, the base station establishes a channel for the call. The channel consists of two carrier frequencies: one frequency is for the base station to transmit to the mobile unit (referred to as the forward channel) and another frequency is for the mobile unit to transmit to the base station (referred to as the reverse channel). Within a channel, the forward and reverse sets of frequencies are separated to provide full-duplex operation.

If a terminal moves from one cell to another, a handoff must occur. The base station currently serving a mobile continuously measures the received signal. If that signal falls below a specific level, then a handoff sequence is initiated. The first step is for the base station to send a message to its switch (mobile telephone switching office, MTSO, or mobile switching center, MSC). The switch asks nearby base stations to measure the signal from the mobile unit, and the base station having the strongest signal wilt then handle the call. This usually requires switching the mobile to another channel (referred to as a hard handoff).If all goes well, the switching to another base station is virtually unnoticed by those involved in the conversation.

If the unit happens to be adjacent to an area served by another system, then an intersystem handoff is initiated. This type of handoff is more complicated, but the principle is the same. In the end, the mobile will be connected to a new base station which is part of a different MTSO.

1.3.2 Cell Size and Expansion of Systems

In areas of low population density and thus little cellular traffic, the cells can be up to several kilometers in diameter, depending on the lay of the land hills, valleys, large buildings, and other obstructions. All of these configurations affect the design and layout of cells. Of course power restrictions also determine how large a cell can be. As we move into more densely populated areas, including suburbs and cities, the phone traffic expands proportionately as do building obstructions. Another area of increased mobile phone usage is along major highways and at the intersection of two or more major highways. These conditions necessitate having smaller cells.

There is a lot of research and developmental software on the subject of designing cells. Most of the fundamental mathematics and computer science theories have been developed, but many researchers are still studying the nature of the optimum definitions of cells. Many telecommunications engineers and technologists are involved in cell design and antenna design and positioning.

In the recent past, an explosive growth of cellular telephone usage has occurred. Of course, manufacturers and service providers are delighted with the increased business, but there are consequent problems. The biggest problem occurs when all the channels are in use and there is no room for another call. This leads to call blocking not allowing a subscriber to initiate a call. Incidentally, it is a dilemma for service operators to choose between allowing a new call versus allowing a handoff of a call in progress into a busy cell. Typically, companies have decided that customers find it more annoying for a call to be terminated than to have to wait to start a call. The service provider will leave some channels unused even though a caller is waiting to initiate a call. The main problem is that the increased traffic can only be accommodated in the following ways:

- Expand bandwidth and reallocate frequencies this is a problem because only so much spectrum is available (the allocation was expanded from 40 MHz to 50 MHz) and the channels cannot be made narrower without serious interference problems.

- Subdivide the cells so there are fewer users per cell this is the main way users have been added, but there are practical limitations on how small cells can be.
- Develop and implement new technologies this is also being done

(TDMA and CDMA).

1.3.3 Signal Degeneration in Cellular Systems

A signal moving down a copper wire is attenuated so many dB per foot due to loss in the wire. Also, a light signal in a fiber optic cable is attenuated. We can use amplifiers and regenerators at the appropriate spots to bolster these signals. However, we do not have such a luxury for radio signals in the air, and these signals also degenerate with distance. Of course, this degeneration is another limitation on cell size.

A common unit of power is the dB where dB is defined as P(dB) = lOlog [P(W)l. However, this is a fairly large unit for cellular systems and more often the unit dBm is used, which is defined as: P(dBm) = 10log[P(mW)l = 10log[watts/lmW], For example, a transmitter transmitting 2 watts of power could be expressed as:

(a) P(dB) = 10 log[P(W)] = 10 log(2) = 3.01 dB or,

(b) P(dBm) = 10 log[P(mW)] = 10 log(2W/lmW) = 10log(2000) = 33.0 dBm.

There are four major impediments regarding signal integrity in cellular telephony:

1. The power loss due to distance, assuming free space (no obstructions), can be determined. The received powers, P_R, can be determined from $P_R = kP_T/(4\Pi)^2 d^2$ where P_T is the transmitted power, d is the distance, and k is a constant that is determined from such variables as transmitter and receiver antenna gains (G_t. and G_r)/ wavelength of the signal (λ), and losses in the system hardware (L). The formula for k is: $k = G_tG_r \lambda^2/L$
2. Objects such as buildings and hills in the path of the signal cause both signal loss due to absorption and additional signal paths due to reflection. These multiple paths can cause a single signal to arrive at the receiver at different times, producing phase shifts.

3. Motion of the terminal across the terrain causes what is called slow fading: that is, at different spots the reception will be stronger or weaker than in other spots. These signal changes are mostly random because of the general unpredictability of the terminal's motion. However, in rural areas the main effect is decreasing power due to distance (approximately a linear decline with the log of the distance from the transmitter). In cities and suburbs the random effect is more noticeable.
4. Rayleigh fading (or fast fading) is due to the terminal moving quickly through a cell. The various rays reaching the receiver undergo a Doppler shift (the same as the familiar examples of train whistle frequency changes or the red shift of stars). Sometimes the signals add and sometimes they subtract, adversely affecting the quality of the received signal even when the average received signal is strong.

There are various competing factors in the quality of service relative to power. On the one hand, we would like to boost the power from the base stations and from the mobiles (forward and reverse), but on the other hand increasing power increases the likelihood of interference among cells. Also, since mobiles are typically battery powered, increasing the transmit power of a mobile unit proportionately reduces its battery life. Thus, system designers look for other methods to increase the quality of service.

1.3.4 Roaming, Intersystem Operations, and IS-41

Roaming occurs when a subscriber for a particular service provider moves into an area administered by another service provider. Typically, this has meant an extra charge tacked on to the user's bill; however, competition among the service providers is, if not totally eliminating these expensive roaming charges, at least making the regions where there is no roaming charge larger and larger.

Aside from charging customers money for roaming, technological problems need to be overcome to actually allow roaming. In the early days of cellular telephony, large and/or nearby cellular carriers made agreements and common procedures to set up and document calls for billing purposes. This system was problematic, however, in that there was no national standard for this rapidly growing industry. In the late 1980s and early 1990s the Telecommunications Industry Association (TIA) developed interim standard 41 (IS-41) so a national standard (U.S. and Canada) would provide services to roaming subscribers.

IS-41 is actually a whole seven-layer protocol. The standard specifies devices, interfaces, switches, and even databases. IS-41 borrows from and is consistent with the global system for mobile communications (GSM). GSM is the digital cellular system standard used in Europe and is a very comprehensive standard in comparison to most U.S. standards that are mostly concerned with the interface between the base and mobile terminals, referred to as the air interface.

A whole vocabulary of messages is defined by IS-41. These messages are mainly used to manage intersystem handoffs. Roaming is coordinated by two major databases: a home location register (HLR) and a visitor location register (VLR). The critical element in making roaming work properly is the system identifier (SID) number. Every operating company receives an S1D when it receives an FCC license. The S1D associates any mobile phone with its service provider and thereby indicates whether it is roaming or not; that is, if the mobile unit is outside its service area its SID will not match the S1D of the base station where it is, and it is considered to be a "roamer."

Most of the large wireless service providers arc phasing out roaming and have instituted one-rate type plans. From a bureaucratic and financial point of view/ this is probably the end of roaming, but from a physical, software, and technical point of view, roaming will still exist because the connections have to be made to the various systems. There are other industry plans to be considered such as local number portability (LNP) and calling party pays (CPP). The industry is so new and is growing so fast that it is difficult to predict which systems and which standards will survive.

11.3.5 Multiple Access

Let us consider a single cell with numerous mobile units within that cell. All of these phones are trying to access the base station for the cell. How can the base station service these various phones? There must be a way to separate them and have them operate somewhat, or apparently, simultaneously; that is, we need multiple access. Recall that multiple access was also a problem that had to be resolved in LANs.

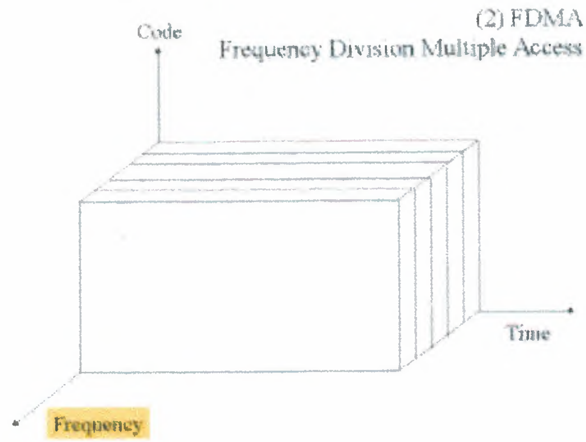
There are three basic techniques for multiple access. The first generation of phones used a totally analog approach that is referred to *as* frequency-division multiple access or FDMA. Similar to radio stations on the AM or FM dial, the channels within a cell are separated by frequency. Each mobile unit in a cell is assigned a set of two frequencies for forward and reverse transmissions, or duplex operation. The available bandwidth determines how many phones can operate within the cell.

The second generation of cellular telephony uses time-division multiple access or TDMA. This system follows interim standards 54 and 136, sometimes referred to as North American TDMA or NA-TDMA. The TDMA system actually uses a combination of frequency- and time-division multiplexing. Each frequency used is separated into time slots and channels are assigned to time slots within the frequencies. This system increases the amount of available channels compared to standard FDMA and thus provides higher spectrum efficiency.

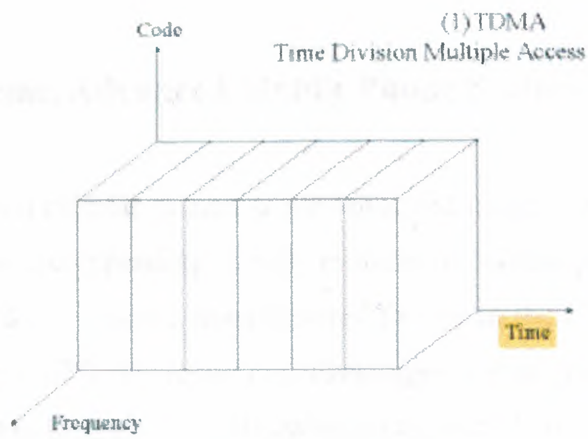
Another second-generation technology uses code-division multiple access or CDMA. In this scheme of multiple access, all the cellular terminals use the full frequency spectrum but are separated by an individual code for each mobile unit. This code becomes part of the modulation technique before transmission and it is up to the base station to extract the signal for a particular mobile unit from all the other signals by demodulating with the code for that mobile unit.

Figure 1-3 diagrams the three different multiple access techniques. Notice in Figure l-3(a) that in the FDMA method the channels are defined by their frequencies, and that a full-duplex channel consists of two frequencies with one for forward (base station to mobile) and one for reverse (mobile to base station), for example, "al" and "a," respectively. Figure l-3(b) shows the TDMA method in which each frequency is divided into time slots, thus providing more channels per frequency. In Figure l-3(c) the PN codes define and separate the channels.

(a) FDMA


Code
(2) FDMA
Frequency Division Multiple Access
Time
Frequency

(b) TDMA


Code
(1) TDMA
Time Division Multiple Access
Time
Frequency

(c) CDMA

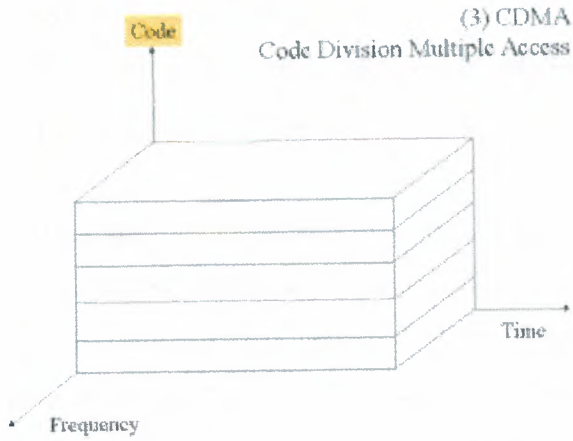


Figure 1.3 FDMA, TDMA, CDMA, methods of multiple access

1.3.6 Analog Systems, Advanced Mobile Phone System (AMPS)

The first-generation cellular system is the advanced mobile phone system (AMPS). Although digital systems are expanding quickly millions of cellular phones still employ the analog system. In fact, digital phones manufactured for use in the U.S. must also have the dual capability of using AMPS. There are two advantages to dual-mode phones: (1) digital phones have the hardware to roam into all analog areas, and (2) it prevents manufacturers and service providers from jumping on the digital bandwagon totally and forcing analog phone customers to throw out their analog phones and sign a digital contract.

If an infinite amount of frequency spectrum were available, we could divide the frequencies and allocate each cell phone its own set of forward and reverse frequencies. There is a limited spectrum however, so each cell has a unique set of frequencies and seven different sets are reused far enough apart so that those cells using the same frequencies do not interfere with each other. Thus we have frequency-division multiple access.

In any given area, the AMPS frequency allocation is as follows: To promote competition the frequencies are divided so that within each area there are always two spectrum allocations, A and B. One operating company uses the A frequencies and another uses the B frequencies. The band of frequencies for forward transmission (base station to mobile) is 869-894 MHz and the band for reverse transmission is 824-849 MHz. Each channel occupies 30 kHz. Dividing up the frequencies creates 416 channels in the forward band and 416 channels in the reverse band for a total of 416 pairs of 30 kHz channels or 416 - 7 = 59 channel pairs per cell. Some of the channels are required for signaling and the remaining channels for traffic.

What type of signal is actually transmitted by the mobile unit? The obvious signal is the carrier frequency that is within the bands just discussed. The carrier is modulated by the voice signal, but actually there is more (see Figure 1-4). Each base station has its own supervisory audio tone (SAT). When the call is set up, a base station and its SAT are established. During the call both the base station and the mobile inject the SAT into the modulator and both continually monitor the SAT to ensure that the mobile unit is receiving the right signal and not that of another base station. The analog signal (usually

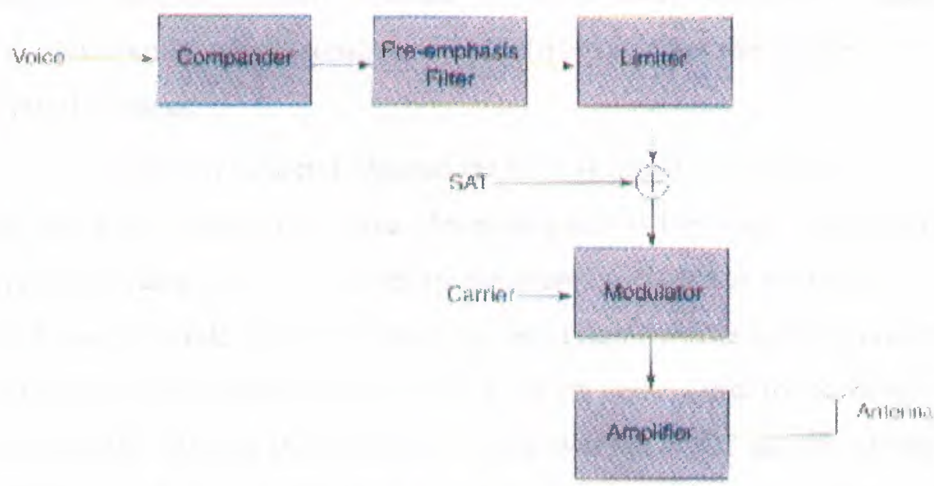


Figure 1.4 Block diagram of the transmit section of an AMPS unit

voice) is also processed before modulation. Because of the high dynamic range of the speech, the signal is also compressed or companded.

Two channels handle the actual conversations: the forward voice channel (FVC), and the reverse voice channel (RVC). The mobile switching center sets up the call by directing the appropriate base station to dedicate an FVC and RVC pair of channels for the duration of the call. The base station monitors the strength of the RVC and SAT signals to help in handoff decisions. If these signals fall below a certain threshold, then the MSC initiates a handoff.

AMPS Signaling and Messages : Channels 313-354 are used exclusively for system control information. When a call is in progress, AMPS uses a system called blank-ami'-burst. As the name implies, the system interrupts the conversation by blanking the sound and inserts a control message. This process takes only 100 ms or less and is virtually unnoticed by the callers because it only sounds like a click and, as long as the clicks are infrequent/ it is not bothersome.

Four AMI'S logical signaling channels are used to carry messages. The formats for each of the control channels are similar in that they all begin with an alternating binary sequence for synchronization. This sequence is produced in AMPS by frequency shifts using Manchester-coded frequency shift keying. These signals are created using the carrier ±8 kHz. Also, every control channel format will follow the synchronisation bits with an 11-bit Barker code, which consists of 11100010010. After the Barker code, comes the actual control message,

1. The forward control channel (FOCC) is broadcast by the base station and received by the mobile units in its area. As mentioned earlier, each area is subdivided into two systems, A and B. These phones are numbered such that A phones have even numbers and B phones have odd numbers. Therefore, each frame carries a 28-bit code word for terminals with even phone numbers (A) and a 28-bit code word for terminals with odd phone numbers (B). Among the messages carried over the FOCC are the mobile identifier number (MIN) page for a called mobile and the message for a mobile to move to a specific voice channel. The FOCC also broadcasts global messages (pertinent to all mobiles in the area), such as the frequency terminals should use to transmit registration messages and the power level for messages transmitted by the terminals. The base channel continuously transmits on the FOCC. When there are no actual messages, the system sends filler messages that contain a 3-bit number specifying the transmit power level for messages transmitted by the mobile units.

2. The reverse control channel (RECC) is random access from the mobile unit to the base station. It is random access because any mobile could be transmitting on the channel and there must be some means of dealing with a number of mobiles contending for the base station's attention. The mobile observes the busy/idle bit on the FOCC for an available window to transmit its message. After the Barker code is a 7-bit digital color code, which exists for each base station and allows the mobile unit to confirm it is part of this base station. Following the digital color code is the actual message. The types of messages sent over the RECC logical channel are acknowledgment of receipt of page, sending the mobile's electronic serial number (ESN)., initiation of a call request along with MIN, and the called party's number.

3. The FVC occupies the same band as the conversation and is thus in-band signaling. Since this is a dedicated channel, it is actually a one-to-one message with no need to identify the receiving mobile unit. Each message is repeated "11 times to increase the probability of proper reception at the mobile unit. The FVC carries such critical messages as an alert of an incoming call, handoff, and power level change. This 28-bit code is further encoded into a 40-bit code using the Bose-Chaudhari Hocquenghem block code before being attached to the sync code and the Barker code and then repeated 11 times and finally transmitted.

4. The RVC is also in the same band as the conversation. It has a similar format to the FVC and carries only messages in response to messages from the base station, such as confirmation of receipt of a power control message. Two other signals of importance are the SAT and ST signals. The SAT, previously mentioned, is the supervisory audio tone, and is transmitted by both the base station and the mobile unit. There are three different SAT signals: 5970, 6000, and 6030 Hz. SAT signals are used to distinguish nearby base stations from each other. The signaling tone (ST) is a 200-ms burst of alternating Is and Os sent by the mobile unit to indicate an end of call.

In the early 1990s, an enhanced analog system, referred to as N-AMPS, was developed and implemented by Motorola. The N stands for narrowband. This system divides the 30-kHz AMPS channel into three 10-kHz channels, essentially increasing the capacity by a factor of three, thus making it useful for densely populated areas.

1.3.7 Time-Division Multiple Access (TDMA)

With the demand for cellular telephony far exceeding expectations, the various companies examined new technologies to extend or replace the analog AMPS system. Not only is system capacity being approached but telephone users have also come to expect more services than can be provided by a typical analog system; a higher-quality signal; a better and more secure user authentication process; and an improvement in roaming. Time-division multiple access (TDMA) is a response to these demands.

TDMA has approximately three times the capacity of AMPS, but because it uses digital technology, it has many other features and improvements as well. Enhanced versions of this system are often advertised as digital personal communications services (PCS) systems. TDMA is actually a hybrid system that implements multiple access by a combination of time and frequency division. Each frequency is divided into time slots, so the total number of physical chamx'ls becomes a product of the number of frequencies in a cell and the number of time slots per frequency times the reuse factor (number of cells before reuse). This combination of frequency- and time-division multiple access is officially referred to as United States digital cellular (USDC) or as digital AMPS (D-AMPS).

The specifications for this first generation of digital cellular are laid out in interim standard 54 (IS-54). IS-54 specifies dual mode, allowing for a smooth transition to digital by allotting the same frequency band and spacing as AMPS but with multiple users on each carrier (time division). So a given base station may have some of its channels dedicated to the digital cellular units and other channels dedicated to analog. As more digital phones are being used more channels become dedicated to USDC.

The voice channels each occupy 30 kHz of bandwidth as in the AMPS systems. However, each frequency is divided into time slots as was shown in Table l-1(b).

Table 1.1 IS-136 digital traffic channel (DTCH) for one slot

Bits	**28**	**12**	**130**	**12**	**130**	**11**	**1**
Field	**SYNC**	**SACCH**	**DATA**	**DVCC**	**DATA**	**DL**	**RES'D**

(a) The Six Data Fields of One Slot for Forward (Base to Mobile)

Bits	**6**	**6**	**16**	**28**	**122**	**12**	**12**	**122**
Field	**G**	**R**	**DATA**	**SYNC**	**DATA**	**SACCH**	**DVCC**	**DATA**

(b) The Five Data Fields of One Slot for Reverse (Mobile to Base)

Duplex transmission is accomplished with different frequencies as in the AMPS system, so both sets of frequencies arc time multiplexed into time slots. Each slot contains a total of 324 bits. There are six slots per frame, therefore each frame transmits a total of 6 x 324 = 1944 bits. Under IS-136 there are two basic logical channels: the digital traffic channel (DTCH), and the digital control channel (DCCH).

Note that both forward and reverse transmissions contain the slow associated control channel (SACCH), which is used to carry control messages between the mobile and the base station spread out over several time slots. Among the information carried by the SACCH are power measurements and requests from the mobile for a handoff. Unlike the AMPS where handoffs are totally out of the control of the mobile unit, IS-54 and IS-136 provide for mobile assisted handoff (MAHO).

Table 1-2 shows the contents of one slot of a DCCH. The total number of bits is, as in the DTCH, 324 bits. In actuality the slots are each subdivided into two blocks. Thirty-two blocks are used to form a superframe and two super frames are combined into a hyperframe. You will also notice that the format and some of the fields for a slot are the same as for the DTCH. Table1-2 (a) shows the forward and 1-2(b) shows the reverse formats for a slot.

Table 1.2 IS-136 digital control channel (DCCH) for one slot

Bits	28	12	130	12	130	10	2
Field	SYNC	SCF	DATA	SFP	DATA	SCF	RES'D

(a) The Six Data Fields of One Slot for Forward (Base to Mobile)

Bits	6	6	16	28	122	24	12	122
Field	G	R	PREAM	SYNC	DATA	SYNC+	DVCC	DATA

(b) The Five **Data** Fields of One Slot for Reverse (Mobile to Base)

In Table 1-2 (a) and (b) we see the data, sync, G, and R fields just as in the DTCH. In Figure 1-6(a), the forward DCCH, the shared channel feedback (SCF) provides feedback information to mobile units in regard to various requests and information sent by the mobile units to the base station. The super frame phase (SFP) indicates what block number the current block is (1 of 32).

In Table 1-2(b), the pream and sync+ fields carry additional synchronization information. The last data field is sometimes replaced with an abbreviated slot containing 78 bit? of data and 44 bits of ramp and guard time.

TDMA requires the storing of information. Thus, although it could be implemented with analog technology, it is more practical with digital technology. On the transmit side, the information is first stored, compressed, and then transmitted during the appropriate time slot. On the receiver side, the information must be stored as received because of the high rate of speed and then played back at the normal rate of speed such that the gaps (between the allocated time slots) are not noticed.

1.3.8 Code-Division Multiple Access (CDMA)

Code-division multiple access (CDMA) is a different approach to multiple access when compared to FDMA and TDMA. CDMA depends on spreading the used spectrum, in fact to the point where a channel requires not 30 kHz but actually 1.25 MHz. The concept of spread spectrum has been around since World War II, when it was used by the military to implement secure and relatively noise-resistant communications.

CDMA had been proposed as a means of multiple access for cellular telephony for several years, but it was not practically developed until the early 1990s by Qualcomm. Qualcomm, a company with headquarters in San Diego, holds most of the patents on CDMA and licenses other companies to use the technology. CDMA systems follow the interim standard 95 (1S-95). 1S-95, similar to IS-54, requires the ability of dual-mode operation; that is, all equipment must implement AMPS as well as CDMA.

Some early optimistic predictions were made regarding the improvement in system capacity relative to AMPS up to 30 or 40 times the capacity but it looks like the actual improvement will be around 10 times the capacity. CDMA spectrum licenses have been bought up very quickly and expensively and as of 1998 CDMA is available virtually across the entire U.S.

The basic idea of spread spectrum as the name implies, is to spread the information to be transmitted across the whole available spectrum, then to extract that information at the receiver. At the expense of heavy processing at the transmitter and receiver, CDMA provides a robust, highly spectrum efficient and highly secure transmission system.

The two most common spread spectrum techniques are; (1) frequency hopping, which is implemented by changing the carrier frequency with every frame and sequencing through a number of frequencies, thereby creating a large bandwidth of frequencies; and (2) using a binary spreading sequence to modulate a digital carrier before this new signal modulates a radio frequency (RF) carrier. The latter technique is used in CDMA systems.

1.3.9 Identifiers in Cellular Telephony

A critical issue in cellular telephony is the different entities identifying themselves. By this we mean such things as base stations having their own IDs, each cell phone having its own ID, and networks also having their own IDs. Unfortunately, most entities need more than one ID. AMPS. NA-TDMA, and CDMA. Notice there is some commonality, but becareful since some of the IDs may be die same but have a different number of bits.

1.3.10 PCS—Cellular Telephony Gets Smarter

Personal communications services (PCS) has achieved the ultimate goal of all the wireless service providers. Regardless of the technology used, a PCS phone would have several additional services beyoiid basic cellular telephony, including voice mail, call forwarding, caller ID, call waiting, conference calls, and short message services. In other words/ voice communications in PCS is the starting point to access all information services.

True PCS, as defined by the FCC, operates in the frequency range of 1850 to 1990 MHz. Other systems may call themselves PCS and offer similar services, but operate at different frequencies. For instance, TDMA and CDMA systems operating at 800 MHz by companies such as AT&T and Bell Atlantic advertise their services as PCS and do provide PCS services. Omnipoint and Sprint PCS operate at 1900 MHZ, with CiSM and CDMA technologies, respectively.

The advantage for companies operating in the 800 MHz range is that they were able to use their existing antenna infrastructure because that frequency range is the same as has been used by the analog cellular system, Also, the phones for these systems have been dual mode, allowing users to tap into AMPS systems in areas where PCS is not yet available. Those companies using the higher-frequency range have had to build new antennas at considerable cost in time and money.

PCS phones, at this time, are considerably more expensive to buy, at 5 to 10 times the cost of a regular cell plume. The call quality is better, with the CDMA systems having the best sound quality. The coverage areas for PCS are gradually increasing, with the areas in and around cities increasing at a faster rate. Security is better, with eavesdropping virtually impossible except that, of course, a dual-mode phone roaming in a non-PCS area is susceptible.

1.3.11 Frequency Allocations

The actual frequencies used by the carriers for the various systems are all in the high MHz range. For the AMPS system there are two sots of frequencies: reverse (mobile to base station or uplink) is 824-849 MHz and forward (base station to mobile or downlink) is 869-894 MHz. The carriers are separated by 30 kHz. Recall that the frequencies are allocated to an A and B provider so that the total number of channels available is halved. Taking 25 MHz (of the forward or reverse direction) and dividing by 30 kHz yields 833 physical channel pairs, but then dividing it by 2 (2 providers) yields 416 pairs of 30 kHz channels. If we now take the 416 pairs and divide by the reuse factor of 7, we end up with 59 channels per cell.

The TDMA system is assigned to the same yet of frequencies as AMPS; however, because of time multiplexing iiito slots, each carrier handles three physical channels. Thus, TDMA has three times the capacity of AMPS. Also, PCS is allotted frequencies of 1850-1910 MHz for reverse and 1930-1990 MHz for forward channels.

CDMA has the same set of frequencies as TDMA, but these frequencies are used differently. The carrier spacing per channel is approximately 1.25 MHz, thus full duplex requires approximately 2.5 MHz. Of course, this is due to the spread spectrum method of CDMA. The number of physical channels cannot be specified exactly but is soft in that it depends on the signal-to-noise ratio.

The one thing that all three systems (AMPS, TDMA, and CDMA) have in common is the method of duplex operation; that is, the forward versus reverse transmissions. This process is accomplished by frequency separation as noted above, a set of frequencies is always assigned for forward or downlink transmissions and a different set, at least 45 MHz away, is assigned for reverse or uplink transmissions. TDMA not only separates by frequency but also by time there is an offset in time of approximately 2 ms so that the mobile does not have to receive and transmit at the same time, thereby simplifying the unit's circuitry.

1.4 Wireless Networking

Wireless local area networks (WLANs) are becoming more common in businesses, convention centers, college campuses, and many other similar facilities. They are generally most useful in confined areas with short distances (a few hundred feet). Also, they are typically operated on a noninterference basis and are thus unlicensed.

There is a huge infrastructure of wiring in businesses and other similar facilities. WLANs are used where it is difficult to add or change wiring or where the employees are very mobile. One of the factors holding back a large growth in the area of WLANs has been the difficulty of increasing the data rates. Also such data communications present many problems in that short E-mail messages would usually be acceptable, but any extended data communications run the risk of errors and lost data due to all the obstacles in radio communications, However, recent hardware breakthroughs and the introduction of a new standard by the IEEE, 802.11, are expected to enable the industry to expand.

One important use of WLANs is with portable computers. Most desktop computers (at companies anyway) are connected to a network but portable PCs are not. Connecting laptops through WLANs would allow for mobility and interconnectedness,

1.5 Satellite Communication

Satellite communications are becoming more prevalent. Aside from. Television broadcasts, which everyone is familiar with, satellites play a role in a variety of other communications applications,

Satellites have revolutionised navigation throughout the globe. Airplanes and ships depend heavily on satellites for accurate measurements of their locations. Individuals in their cars or with hand-held devices can also make use of the global positioning system.

Cellular telephone systems can also make use of satellites. The features and ranges all depend on the number and orbit height of the various satellite systems. Of course, satellites have the advantage of being able to cover wide areas and are particularly useful in hostile or inaccessible areas such as oceans and deserts.

One system, the indium satellite-based cellular system, has satellites in low earth orbit or LEO (700-1000 km), with 12 orbits and 6 satellites per orbit. This system uses a frequency reuse plan similar to AMPS, has 400 beams per satellite, and covers about 50 miles in diameter per beam.

1.6 Wireless Networks

The mid-1980s saw major new developments in the wireless information industry. The transition to digital cellular technology, led by the Pan-European GSM standard has been followed by the EIA-TIA North American Digital Cellular Standard's initiatives and the Japanese Digital Cellular standard. The prime motivation tor these initiatives has been to increase the capacity of cellular telephone systems, which have reached the capacity limits of the analog technology in some highly populated metropolitan areas. The extraordinary success of the cordless-telephone market spurred new standardization efforts for digital cordless and TclePoint in the United Kingdom wireless and in Sweden advanced cordless phone in Japan and the concept of a Universal Digital Portable Communicator in the United States. The success of the paging industry led to development of private wireless packet data networks for commercial applications requiring longer messages. Motivated by the desire to provide portability and to avoid the high costs of installation and relocation of wired office information networks, wireless office information networks were suggested as an alternative. Another major event in this period was the announcement regarding unlicensed ISM bands in May of 1985. This announcement opened the path for development of a wide array of commercial devices ranging from wireless and wireless LANs to wireless fire safety devices using spread spectrum technology.

Figure 1.5 distinguishes the various categories of wireless networks we discuss in this chapter. We first define two broad categories of networks as (1) voice-oriented or isochronous networks and (2) data-oriented or asynchronous networks. Under each main category of networks, we distinguish further between local-area networks and wide-area networks. Each of the resulting four subcategories of networks has a set of characteristics that leads to certain design choices specific to the subcategory. Figure 1.5 which is structured according to the categories defined in Figure 1.6 depicts various dimensions of today's voice and data communications industries, comparing local cordless voice communication with wide-area cellular voice services, and also comparing wireless LANs with wide-area, low-speed data services

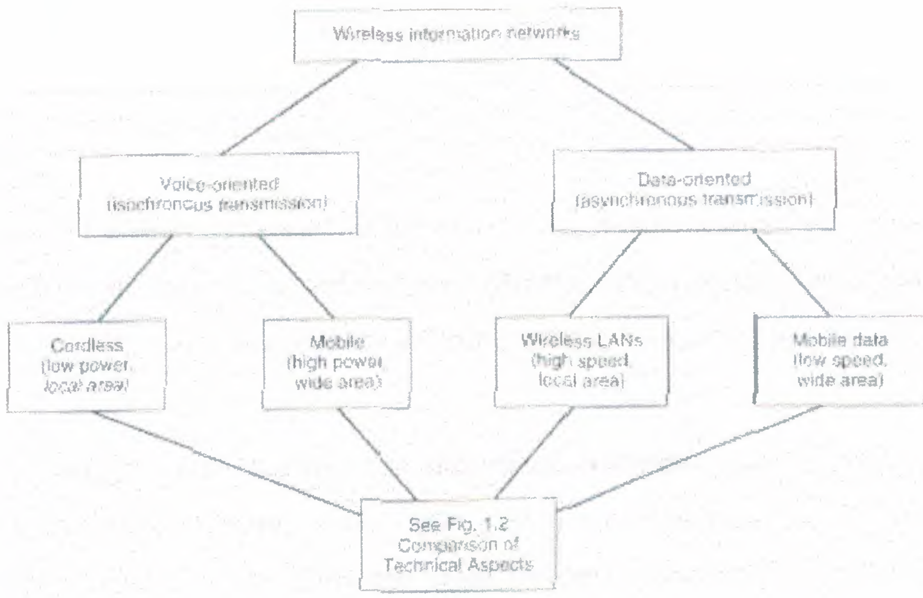
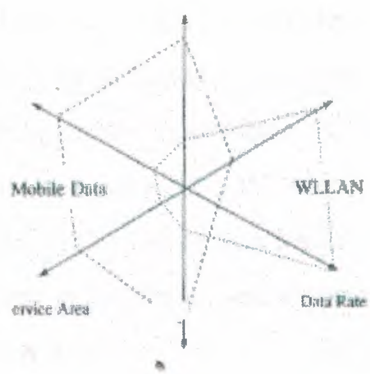


Figure 1.5 Categories of wireless information networks Tariff



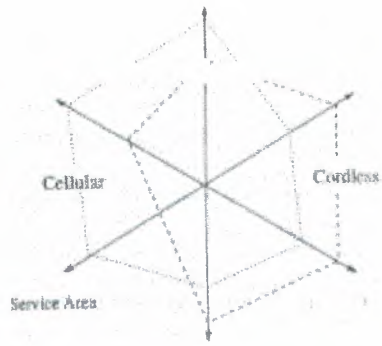


Figure 1.6 Various aspects of the voice (a) and data (b) oriented services offered by the wireless information network industry. Mobile cellular systems are compared with cordless PCS, and WLLAN data systems are compared with mobile data networks.

Figure 1,6(a) compares the importance of various issues or parameters related to the wireless voice industry, while Figure 1.6(b) provides an analogous comparison between wireless data systems. Although from the user's standpoint and the appearance of the handset the digital cellular and PCS systems look very similar, there will in fact be major differences in the operation of the networks supporting the two categories of systems.

A digital cellular system is designed to support mobile users roaming over wide geographic areas, and thus coverage is provided by an arrangement of cells with cell size typically 0.5 to 5 miles in diameter. The radio cell sites for this system are large and expensive, particularly in urban areas where the cost of property is very high. The handset requires an average power of around 1 **W,** which is reflected in limited battery life and the need for frequent recharging of the batteries or reliance on connection to the car battery. The number of users per cell is large; and to provide as many user channels as possible in the allocated bandwidth, complex speech-coding algorithms are used, which minimize the digitized speech transmission rate but consume a significant amount of electronic power, which in turn places a high demand on battery power.

The PCS systems will be designed for small, low-power devices to be carried and used in and around office buildings, industrial complexes, and city streets. The cell size will be less than a quarter-mile, and the relatively small base stations will be installed on utility poles or attached to city and suburban business buildings. The average radiated power will be 10-20 mW, leading to relatively long battery life. The PCS systems are to replace cordless phones in many market areas, and the quality of voice service is intended to be comparable to wirelinc phone service. As a result, simple but high-quality speech-coding algorithms such as 32-kbit/sec are adopted. While a high-quality voice-coding algorithm of this type does not provide the spectral efficiency of the lower-rate (4-8 kbits/sec) vocoders used in the digital cellular standards, it is far less demanding of digital signal processing complexity, and thus permits the use of very low prime power in the portable units. It is expected that PCS service will distinguish itself from digital cellular service by higher voice quality, longer battery lifetime, and lighter terminals. For more details on the digital cellular and PCS initiatives, the reader can refer to and other references cited therein.

Mobile data networks operate at relatively low data rates over well-understood urban radio channels using familiar multiple-access methods. The technical challenge here is the development of a system which makes efficient use of the available bandwidth and the existing infrastructure to support widely separated subscribers. The transmission technology used in mobile data networks is generally rather simple and similar from one network to another. We discuss the major existing and planned mobile data networks and services ..

WLANs and mobile data networks serve somewhat different categories of user applications, and they give rise to different system design and performance considerations. A WLAN typically supports a limited number of users in a well-defined local area, and system aspects such as overall bandwidth efficiency and product standardization are not crucial. The achievable data rate is generally an important consideration in the selection of a WLAN, and therefore the transmission channel characteristics and the application of signal processing techniques are important considerations . Access methods and network topologies used in WLANs are much the same from one system to another, but the transmission technologies are different. Efficient design of these systems requires evaluation of various transmission techniques and an understanding of the complexities of indoor radio propagation. WLAN manufacturers currently offer a number of nonstandardizcd products based on conventional radio modem technology, spread-spectrum technology in the ISM bands, and infrared technology.

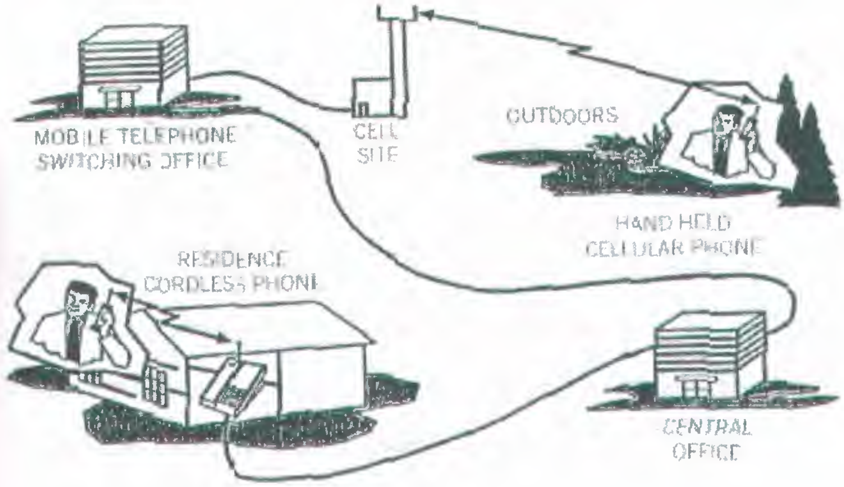


Figure 1.7 Current tetherless communications.

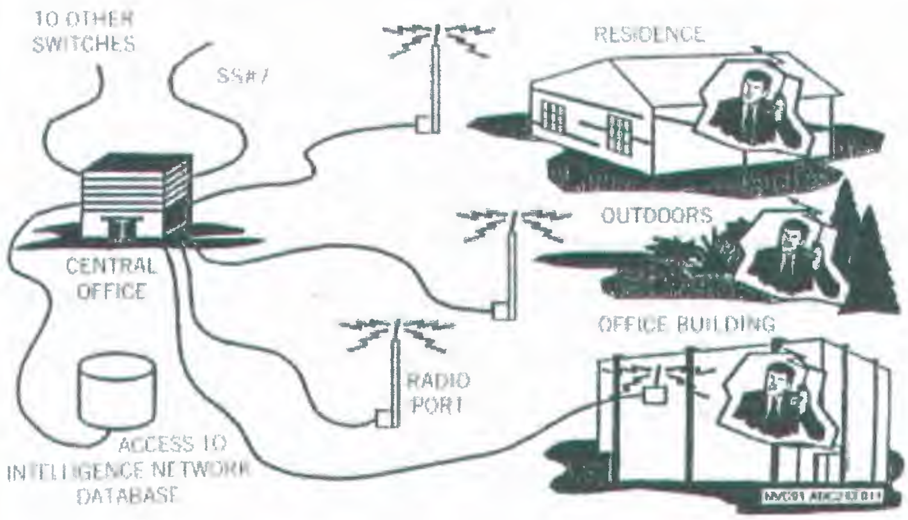


Figure 1.8 Low power, exchange access digital radio integrated with network intelligence

CHAPTER 2

FREQUENCY DIVISION MULTIPLE ACCESS (FDMA)

2.1 Introduction

Transmission in telephony means sending information on electricity or light from one point to another. Voice or data makes up the transmission. We call the device or matter that the information travels on, be it wires, cable, or radio waves, the transmission media

Cellular refers to communications systems, especially the Advance Mobile Phone Service (AMPS), that divide a geographic region into sections, called cells. The purpose of this division is to make the most use out of a limited number of transmission frequencies. Each connection, or conversation, requires its own dedicated frequency, and the total number of available frequencies is about 1,000. To support more than 1,000 simultaneous conversations, cellular systems allocate a set number of frequencies for each cell. Two cells can use the same frequency for different conversations so long as the cells are not adjacent to each other. For digital communications, several competing cellular systems exist, including GSM and CDMA. Analog cellular might use conventional frequency multiplexing division. GSM only works in TDMA.

There are three common technologies used by cell phone networks for transmitting information:

- Frequency Division Multiple Access (FDMA)
- Time Division Multiple Access (TDMA)
- Code Division Multiple Access (CDMA)

Cellular phones enable mobile communication to almost anywhere in the world. On a "complexity per cubic inch" the cellular phones are some of the most intricate devices people use daily. Modern digital cell phones can process millions of calculations per second in order to compress and decompress the voice stream. If you ever take a cell phone apart, you will find that it contains just a few individual parts:

- Circuit board containing the brains of the phone - contains RF, digital signal processor, flash and ROM memory, operating system
- Antenna
- Liquid crystal display (LCD)
- Keypad
- Microphone
- Speaker
- Battery

2.2 Frequency Division Multiple Access (FDMA)

A transmission technology in which each subscriber is assigned a specific frequency channel, that can then be used by no one but them. FDMA reduces interference, but severely limits the number of users

The use of frequency division to provide multiple and simultaneous transmissions to a single transponder. A transponder is an automatic device that receives, amplifies, and retransmits a signal on a different frequency.

- You can think of radio stations signals stacked up next to each other in the frequency domain.
- This method of allowing several signals to be transmitted simultaneously is called Frequency Division Multiple Access.
- **Acronym:** FDMA.
- When you tune your radio to a particular station, you actually tune a bandpass filter to let only the signal from your favorite radio station pass.

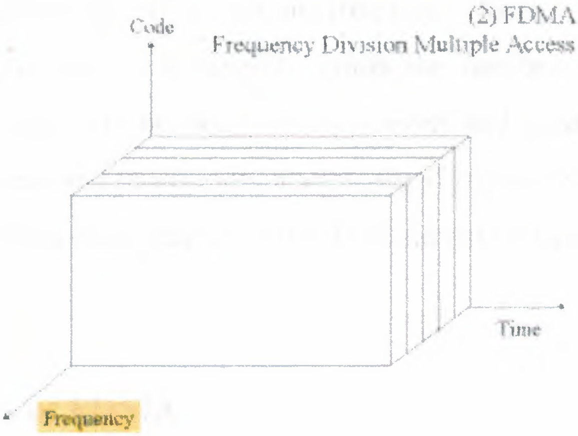


Figure 2.1 Frequency division multiple access.

With FDMA the bandwidth of the channel is divided among the population of stations. For example, with six stations the frequency range of the channel is divided by six and each station gets its own private frequency. In this way there is no interference between users.

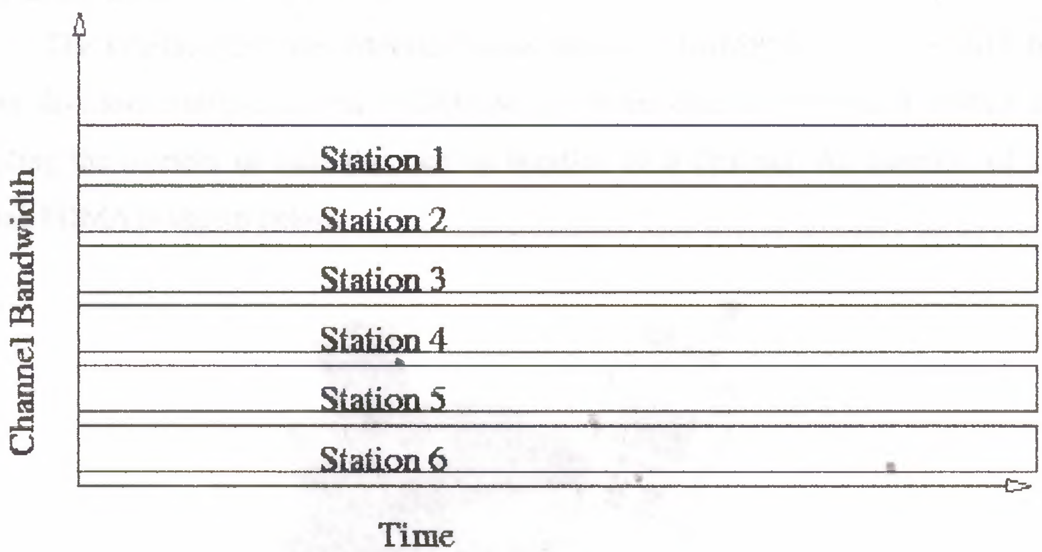


Figure 2.2 Channel bandwidths of Frequency Division Multiple Access

Each FDMA subscriber is assigned a specific frequency channel. No one else in the same cell or a neighboring cell can use the frequency channel while it is assigned to a user. This reduces interference, but severely limits the number of users. This implies that relatively narrow filters are needed in each receiver and transmitter. Most duplex FDMA systems must transmit and receive simultaneously. (Frequency Division Duplex, FDD) This requires expensive and bulky duplex filters to avoid strong transmit signals leaking into the receiver.

2.3 Examples of FDMA

FDMA (frequency division multiple access) is the division of the frequency band allocated for wireless cellular telephone communication into 30 channels, each of which can carry a voice conversation or, with digital service, carry digital data. FDMA is a basic technology in the analog Advanced Mobile Phone Service (AMPS)the most widely-installed cellular phone system installed in North America. With FDMA, each channel can be assigned to only one user at a time. FDMA is also used in the Total Access Communication System (TACS).

The Digital-Advanced Mobile Phone Service (D-AMPS)also uses FDMA but adds time division multiple access (TDMA)to get three channels for each FDMA channel, tripling the number of calls that can be handled on a channel. An example of a system using FDMA is shown below.

Figure 2.3 equipment for private mobile radio system often uses FDMA.

Early cellular telephony mostly used FDMA analogue transmission. Walkie talkies and mobile networks for closed user groups often use FDMA. More advanced systems time-share frequencies among different user groups. This concept is called trunking.

Another example of FDMA is AM or FM radio broadcasting where each station has its own channel.

2.4 FDMA Features

- If channel not in use, sits idle.
- Channel bandwidth relatively narrow (30kHz), i.e, usually narrowband system.
- Simplest.
- Best suited for analog links .
- Continuous transmission implies no framing or synchronization bits needed.
- Requires tight filtering to minimize interference .
- Usually combined with FOD for duple Xing

2.4.1 Frequency Hopping SS

- Pseudo-random frequency changes randomizes channel occupancy
- At any given time, FH signal occupies only a single, narrow channel; makes MA possible
- FHMA is a fast (channel) changing FDMA
- Slow hopping : multiple bits before frequency hop
- Fast hopping : multiple frequency hops per bit

2.4.2 Spread Spectrum

- Techniques known since 1940s and used in military communications systems since 1950s
- Spread the radio signal over a wide frequency range several magnitudes higher than minimum requirement
- Better interference immunity and multiple access ability
- Bandwidth efficient for multi-user systems
- Two main techniques: frequency hopped (FH) and direct sequence (DS) or CDMA

2.5 Transmission and Multiplexing

In telecommunications multiplexing (MUXing) is the combining of two or more information channels onto a common transmission medium using hardware called a multiplexer or (MUX). The reverse of this is known as inverse multiplexing

In electrical communications the two basic forms of multiplexing are time division multiplexing (TDM) and frequency division multiplexing (FDM). In optical communications, the analog of FDM is referred to as wave length division multiplexing (WDM).

2.6 Frequency Division Multiplexing

FDM is a form of signal multiplexing where multiple base band signals are modulated on different frequency carrier waves and added together to create a composite signal.

FDM can also be used to combine multiple signals before final modulation onto a carrier wave. In this case the carrier signals are referred to as sub carriers an example is stereo FM transmission, where a 19 kHz sub carrier is used to separate the left-right difference signal from the central left-right sum channel, prior to the frequency modulation of the composite signal.

Where frequency division multiplexing is used as to allow multiple users to share a physical communications channel it is called frequency division multiple access (FDMA).

FDMA is the traditional way of separating radio signals from different transmitters. The analog of frequency division multiplexing in the optical domain is known as wave length division multiplexing.

Basically Frequency division multiplexing is the simultaneous transmission of multiple separate signals through a shared medium (such as a wire, optical fiber or light beam) by modulating, at the transmitter, the separate signals into separable frequency bands, and adding those results linearly either before transmission or within the medium. While thus combined, all the signals may be amplified, conducted, translated in frequency and routed toward a destination as a single signal, resulting in economies which are the motivation for multiplexing. Apparatus at the receiver separates the multiplexed signals by means of frequency passing or rejecting filters, and demodulates the results individually, each in the manner appropriate for the modulation scheme used for that band or group.

Bands are joined to form groups, and groups may then be joined into larger groups; this process may be considered recursively, but such technique is common only in large and sophisticated systems and is not a necessary part of FDM.

Neither the transmitters nor the receivers need be close to each other; ordinary radio, television, and cable service are examples of FDM. It was once the mainstay of the long distance telephone system. The more recently developed time division multiplexing in its several forms lends itself to the handling of digital data, but the low cost and high quality of available FDM equipment, especially that intended for television signals, make it a reasonable choice for many purposes.

2.6.1 Examples of FDM

Analog cellular use frequency division multiplexing or FDM. It's much simpler than its name suggests. a carrier's assigned radio spectrum is divided into specific frequencies, each separated by space. Like AM radio, which is divided into 10 KHz chunks. Radio station 810, 820, 830, and so on. That's all FDM is. Think of FDM as a single train running on a single track, pulling just one freight car. But what if you've run out of frequencies to handle your customers? What if you need more capacity? You can either separate your existing frequencies by narrower amounts or you can separate your calls over time.

Motorola's Narrowband Advanced Mobile Phone system or NAMPS, used precise frequency control to divide the 30 Khz AMPS channel into three sub channels. Each call takes up just 10Khz. But NAMPS had the same fading problems as normal AMPS, lacked the error correction that digital systems provided and it wasn't sophisticated enough to handle encryption or advanced services. To increase capacity most cellular carriers moved instead to a digital solution, one separating conversations by time or by code.

2.7 Advantages and disadvantages of FDMA

Advantages

1. Simple algorithmically and from a hardware standpoint.
2. Fairly efficient when the number stations is small and the traffic is uniformly constant.
3. Each FDMA subscriber is assigned a specific frequency channel. No one else in the same cell or a neighboring cell can use the frequency channel.

Disadvantages

1. Not conducive to varying station population.
2. If traffic is bur sty, bandwidth is wasted.
3. Inter frequency protection bands waste bandwidth.
4. No broadcast capability

CHAPTER 3

TIME DIVISION MULTIPLE ACCESS (TDMA)

3.1 Definition and Terminology

A digital transmission technology that allows multiple users to access a single radio-frequency (RF) channel without interference by allocating time slots to each user within each channel.

Time division multiple access (TDMA) is digital transmission technology that allows a number of users to access a single radio-frequency (RF) channel without interference by allocating unique time slots to each user within each channel. The TDMA digital transmission scheme multiplexes three signals over a single channel. The current TDMA standard for cellular divides a single channel into six time slots, with each signal using two slots, providing a 3 to 1 gain in capacity over advanced mobile-phone service (AMPS). Each caller is assigned a specific time slot for transmission.

- When signals do not overlap in the time-domain then one signal stops before another one begins.
- Signals are easily separated when the receiver only ``listens" while the signal of interest is sent.
- This multiple-access method is called Time Division Multiple Access.
- **Acronym:** TDMA.
- **Example:** TDMA is employed in the digital portion of the telephone network
- **Example:** TDMA and FDMA together are used in many modern digital cellular telephone systems

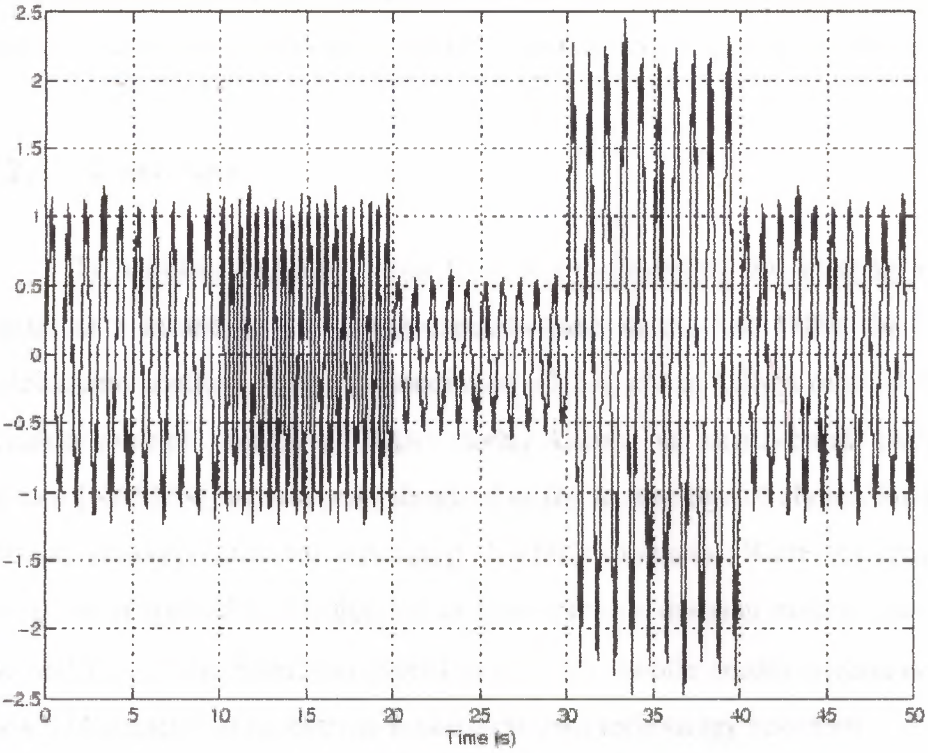


Figure 3.1 Time Division Multiple Access graph

3.2 TDMA is a Growing Technology

TDMA (also known as D-AMPS) is a technology for digital transmission of radio signals between, for example, a mobile telephone and a radio base station. In TDMA, the frequency band is split into a number of channels, which are stacked into short time units, so that several calls can share a single channel without interfering with one another. TDMA is used by the GSM digital mobile standard. TDMA is based on the IS-136 standard. It is one of the world's most widely deployed digital wireless systems. It provides a natural evolutionary path for analog AMPS networks, offers efficient coverage and is well suited to emerging applications, such as wireless virtual private networks (VPNs), and is the ideal platform for PCS (Personal Communication Services).TDMA (Time Division Multiple Access) is a technology for digital transmission of radio signals. The technology is also known as D-AMPS (Digital Advanced Mobile Phone Service). In TDMA the frequency band is split into several channels, which are stacked into short time units. This means several calls can share a single channel without interfering with one another.

3.2.1 Over view

The wireless industry began to explore converting the existing analog network to digital as a means of improving capacity back in the late 1980s. In 1989, the Cellular Telecommunications Industry Association (CTIA) chose TDMA over Motorola's frequency division multiple access (FDMA) (today known as narrowband analog mobile-phone service [NAMPS]) narrowband standard as the technology of choice for existing 800 MHz cellular markets and for emerging 1.9-GHz markets. With the growing technology competition applied by Qualcomm in favor of code division multiple access (CDMA) and the realities of the European global system for mobile communications (GSM) standard, the CTIA decided to let carriers make their own technology selection.

The two major (competing) systems that split the RF are TDMA and CDMA. CDMA is a spread-spectrum technology that allows multiple frequencies to be used simultaneously. CDMA codes every digital packet it sends with a unique key. A CDMA receiver responds only to that key and can pick out and demodulate the associated signal.

Because of its adoption by the European standard GSM, the Japanese Digital Cellular (JDC), and North American Digital Cellular (NADC), TDMA and its variants are currently the technology of choice throughout the world. However, over the last few years, a debate has convulsed the wireless community over the respective merits of TDMA and CDMA.

The TDMA system is designed for use in a range of environments and situations, from hand portable use in a downtown office to a mobile user traveling at high speed on the freeway. The system also supports a variety of services for the end user, such as voice, data, fax, short message services, and broadcast messages. TDMA offers a flexible air interface, providing high performance with respect to capacity, coverage, and unlimited support of mobility and capability to handle different types of user needs.

3.2.2 The Digital Advantage

All multiple access techniques depend on the adoption of digital technology. Digital technology is now the standard for the public telephone system where all analog calls are converted to digital form for transmission over the backbone. Digital has a number of advantages over analog transmission:

- It economizes on bandwidth.
- It allows easy integration with personal communication systems (PCS) devices.
- It maintains superior quality of voice transmission over long distances.
- It is difficult to decode.
- It can use lower average transmitter power.
- It enables smaller and less expensive individual receivers and transmitters.

It offers voice privacy

3.2.3 How TDMA works

TDMA relies upon the fact that the audio signal has been digitized; that is, divided into a number of milliseconds-long packets. It allocates a single frequency channel for a short time and then moves to another channel. The digital samples from a single transmitter occupy different time slots in several bands at the same time.

The access technique used in TDMA has three users sharing a 30-kHz carrier frequency. TDMA is also the access technique used in the European digital standard, GSM, and the Japanese digital standard, personal digital cellular (PDC). The reason for choosing TDMA for all these standards was that it enables some vital features for system operation in an advanced cellular or PCS environment. Today, TDMA is an available, well-proven technique in commercial operation in many systems. To illustrate the process, consider the following situation.

A single channel can carry all four conversations if each conversation is divided into relatively short fragments, is assigned a time slot, and is transmitted in synchronized timed bursts. After the conversation in time-slot four is transmitted, the process is repeated.

Effectively, the IS-54 and IS-136 implementations of TDMA immediately tripled the capacity of cellular frequencies by dividing a 30-kHz channel into three time slots, enabling three different users to occupy it at the same time. Currently, systems are in place that allow six times capacity. In the future, with the utilization of hierarchical cells, intelligent antennas, and adaptive channel allocation, the capacity should approach 40 times analog capacity.

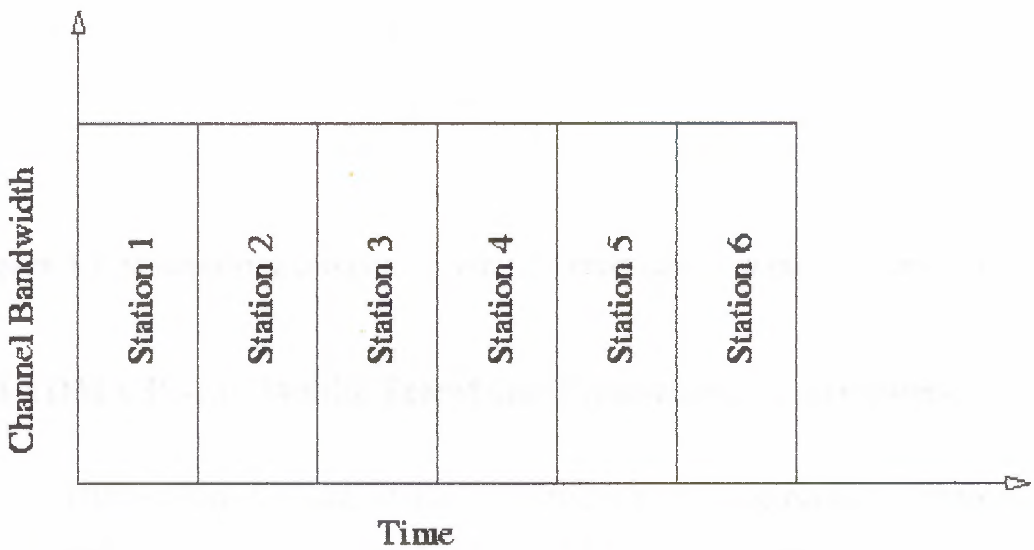


Figure 3.2 Time Division Multiple Access

In TDMA first digitizes calls, then combines those conversations into a unified digital stream on a single radio channel. Time division multiple access or TDMA divides each cellular channel into three time slots. In conventional cellular or AMPS a single call takes up 10Khz. In TDMA based D-AMPS or digital AMPS, three calls get put on that single frequency, tripling a carrier's system's capacity. GSM, D-AMPS, and D-AMPS 1900 (IS-136), and Motorola's iDEN all use or can use TDMA. This scheme assigns a specific time slot, a regular space in a digital stream, for each call's use during a conversation.

Think of a not so drunken cocktail party, with each person speaking in turn. Everyone gets to speak over time. Or think of a train pulling three freight cars. In a TDMA analogy, each caller puts their supplies or payload, their part of the conversation, on every third boxcar in a long train. No need for three separate frequencies like in FDM. With TDMA a single radio channel is not monopolized, rather, it efficiently carries three calls at the same time.

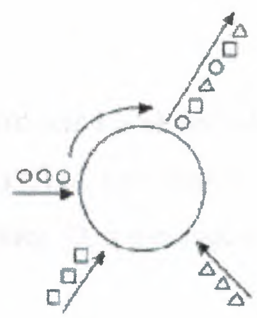


Figure 3.3 Multiplexing combines several different calls into one coherent stream.

3.3 TDMA IS-136 Mobile Telephone Technology, International

TDMA (Time Division Multiple Access) is a second-generation technology used in digital cellular telephone communication, which divides each cellular channel into individual time slots in order to increase the amount of data that can be carried. Several different mutually incompatible implementations of TDMA technologies are in use worldwide, the most prolific being GSM (Global System for Mobile Communications). However, the implementation that is commonly referred to as TDMA is that defined by IS-136 by the Telecommunication Industries Association (TIA).

TDMA forms part of the evolution from first-generation analog systems to second- and then third-generation digital systems. It builds upon the original analog Advanced Mobile Phone Service (AMPS), using the same frequency band of 800MHz, but also operates in the Personal Communication Services (PCS) band of 1,900MHz in the US. Although TDMA could be considered as the least technologically advanced of the second-generation mobile systems, it has proven very popular in the US and developing world as a simple upgrade from analog to digital services. As of December 1999, there were approximately 36 million TDMA subscriptions, accounting for 9% of the digital market.

Although TDMA is currently incompatible with other second-generation systems, there is now a common upgrade path to IMT-2000, which should become the world-wide standard for third-generation mobile communication.

The IS-136 Digital-Control Channel (DCCH): Features And Capabilities are as follows :

The original TDMA standard was IS-54, introduced in 1988-89 by the Telecommunications Industry Association (TIA)/CTIA (see Figure 3.4). It inaugurated a feature set including authentication, calling-number ID, a message-waiting indicator (MWI), and voice privacy

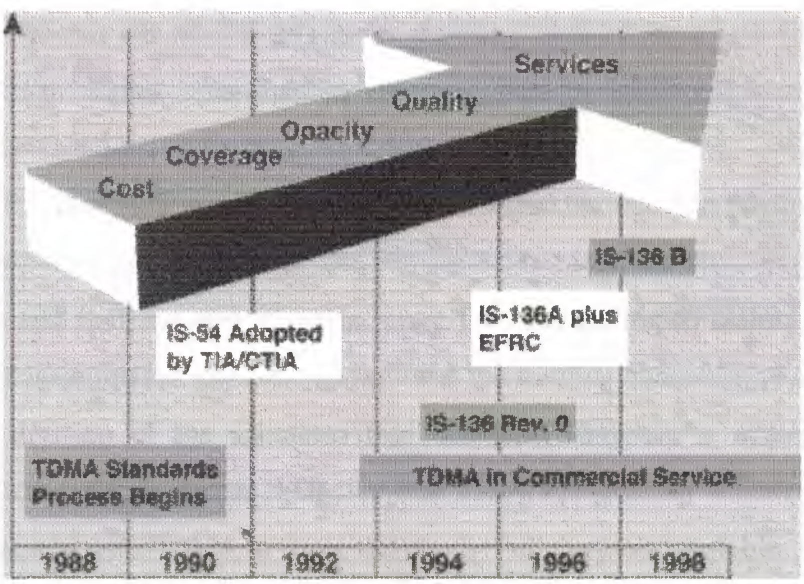


Figure 3.4. TDMA Standards Evolution

- IS-54B was superseded in 1994 with the introduction IS-136 followed closely by revisions A and B.

- IS-136 was backward compatible to IS-54B and included a DCCH and advanced features.

- IS-136A upbanded IS-136 for seamless cellular service between the 800-MHz and 1,900-MHz frequency bands. In addition, it introduced over-the-air activation and programming services.

- IS-136B includes a new range of services including broadcast SMS, packet data, etc.

3.3.1 TDMA Technical Details

TDMA enhances the AMPS service by dividing each of the original 30kHz analog channels into three digital time-division channels, thereby tripling the capacity of the system (called D-AMPS).

Like AMPS, D-AMPS uses frequency ranges within the 800 and 900 MHz spectrum. Each service provider can use half of the 824-849MHz range for receiving signals from cellular phones and half the 869-894MHz range for transmitting to cellular phones. The receiving channels are called reverse channels and the sending channels are called forward channels. The division of the spectrum into sub-band channels is achieved by using frequency division multiple access (FDMA). The TDMA processing is added to each sub-band channel created with FDMA to triple the number of channels available.

TDMA IS-136 was first specified in 1994 and is an evolution of the older IS-54 (also known as Digital AMPS or D-AMPS) standard. IS-54 used the three time-division channels for the voice information only, while IS-136 also used TDMA on the control channel.

A Digital Control Channel (DCCH) increases paging capacity, and sharing TDMA traffic and control on the same digital radio improves efficiency and reduces hardware costs. DCCH also provides the platform for a new generation of advanced wireless capabilities.

TDMA supports text messaging, caller identification and closed-user groups. Using a hierarchical cell structure, it is possible to overlay extra capacity in particular hotspots and offer different services to particular subscribers or areas within the network.

IS-136 supports a variety of digital value-added services, at the same time as being able to coexist with the AMPS network. The inherent compatibility between AMPS and TDMA, coupled with the deployment of dual-mode wireless handsets, ensures ubiquitous network access for the subscriber whether in an analog or digital serving area.

TDMA is designed to allow for seamless interworking and infrastructure sharing with IS-136 TDMA networks at 800MHz and 1,900MHz, as well as the analog AMPS networks. This allows new PCS operators to offer full wide-area coverage from day one through infrastructure sharing or roaming agreements with 800MHz operators in the same geographical area.

The newer IS-136+ and IS-136HS (based upon Enhanced Data Rates for Global Evolution [EDGE] standards) allow a higher bit rate transmission, along with the introduction of General Packet Radio Service (GPRS) data throughput can be increased to over 473Kbs per channel. This packet-switched upgrade can be overlaid on existing networks and allows the system to retain its backward compatibility.

A combined GPRS-136HS technology, known simply as EGPRS, is an ideal bearer for any packet-switched application, including internet connections using TCP/IP. From the end user's point of view, the EGPRS network is an extension of the internet via wireless access.

3.3.2 The Future

The enhancements available to convert an existing TDMA system to a high-throughput packet-switched system can bring some of the proposed advantages from IMT-2000 to existing networks not needing the increased radio spectrum allocation the third-generation (3G) system requires. It can also be seen as part of a gradual progression to 3G systems, eventually allowing interoperability with the other networks using the IMT-2000 standard.

3.4 TDMA Protocols

Time Division Multiple Access (TDMA) protocols have the potential to provide simple but effective broadcast bus communications for embedded systems. However, bus-master based protocols such as TDMA can be undesirable in practice because the bus master node constitutes a single-point failure vulnerability and adds to system expense. We present the Jam-TDMA (J-TDMA) protocol, which eliminates the need for having a bus master through the use of a nondestructive jamming signal for frame synchronization. We give a detailed description of the J-TDMA protocol and show how to minimize the effects of speed differences among nodes on TDMA systems, which can be critical for low-cost implementations. We believe that J-TDMA reaps the benefits of TDMA protocols without suffering the reliability and system complexity drawbacks of other TDMA methods.

3.4.1 Introduction

In this section we discuss Time Division Multiple Access (TDMA) protocols used in broadcast bus communications for embedded systems. TDMA protocols have the potential to be both simple and effective for embedded system applications. In particular, TDMA is at its best when providing highly efficient use of bandwidth for a well characterized, periodic communication traffic workload as found in many embedded systems. Additionally, the simplicity of TDMA lends itself well to embedded systems with limited hardware resources at each node. TDMA also avoids many subtle failure modes associated with more complex protocols, such as duplicate tokens on token bus systems. TDMA can have low protocol overhead if the multiplexed time slices are well balanced with respect to node workloads. And, TDMA does not require collision detection circuitry, which can be difficult or costly to implement in embedded systems.

Unfortunately, it is often difficult to actually use TDMA in practice because of reliability and cost concerns. As described later, classical TDMA uses a single bus-master node to synchronize communications. In many embedded systems, single points of failure are unacceptable; this is true not only in military and avionics systems, but also in many commercial systems, such as elevators and automobiles. Furthermore, a bus master increases size, weight, power consumption and cost. Alternatives to a single bus master seem to simply push complexity comparable to the master node into slave nodes.

The key problem with using TDMA in practice is the need for a physical or logical bus master. This seems to have in practice limited TDMA to those applications that have a natural bus master, primarily satellite communications.

We shall show a way to implement TDMA without using a bus master of any kind. Our technique permits nodes to come on-line and off-line freely, and is accomplished with a minimal increase over the logic complexity of slave nodes over classical TDMA. We feel that this will greatly increase the attractiveness of using TDMA to provide both simple and reliable embedded communications.

Before presenting our new TDMA-based protocol, we first review classical TDMA, then discuss previous solutions to the problems caused by having a bus master.

3.4.2 Classical TDMA

In TDMA, bus access is controlled using a frame-based approach. As shown in Figure 1, transmissions on the bus are grouped into frames. Each frame starts with a frame sync, which is a unique bit pattern transmitted by the bus master. A frame gap following the frame sync is required with some transmission technologies (*e.g.*, transformer coupling) to allow time for the bus master's transmitter to return to quiescent state.

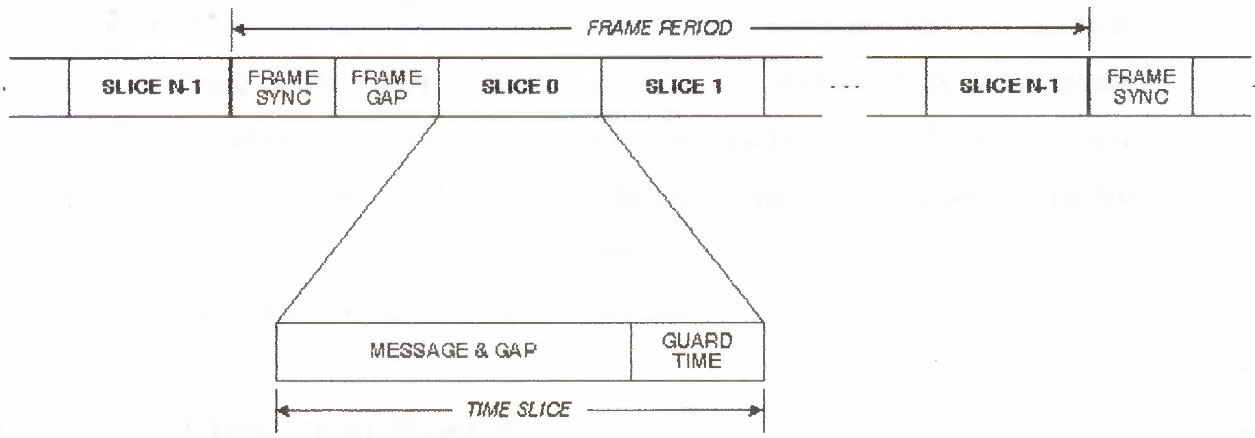


Figure 3.5 TDMA timeline.

The frame gap is followed by N time slices. In the simplest case, one time slice is assigned to each of N slave nodes. When a frame sync is detected by a slave node, that slave node starts a countdown timer that expires at the start of its uniquely assigned time slice. When a slave node's time slice arrives, it transmits a message. In some implementations a gap period after the message is required to allow the transmitter to return to a quiescent state. After the message and gap, a guard time is allocated to accommodate timing skew among the oscillators of the nodes.

When all time slices have elapsed, the bus master transmits another frame sync message to restart the cycle. This frame sync serves as a central time reference point and is used to resynchronize the time bases of each slave node, eliminating cumulative time skew caused by oscillator speed inaccuracies over the duration of a frame. If a slave node has nothing to send during its time slice, or the slave node is off-line, its time slice elapses unused. There are several possible elaborations on this arrangement, such as allocating multiple time slices to nodes with heavy communication workloads and truncating unused slices

3.4.3 The different Guises of Bus Masters

All previous implementations of TDMA seem to use either a permanent or transient bus master of some kind. Unfortunately, use of a bus master tends to introduce complexities and costs that detract from TDMA's advantages. Before proposing a solution, we shall review what we believe to be the common approaches to dealing with bus mastership in TDMA: static allocation of mastership, dynamic allocation of mastership, and initial allocation of mastership with stable time bases.

3.4.4 Static Allocation of Mastership

Static allocation of bus mastership is the classical TDMA approach. In the simplest case, a dedicated bus master is used. An alternative is to replicate the extra logic for mastership within at least one slave node, then designate that node as both a slave and the bus master. For example, node 0 could always be the bus master. The problem with static allocation is obvious: whether the bus master is dedicated or combined with a slave node, failure of the bus master causes network failure.

3.4.5 Dynamic Allocation of Mastership

An alternative scheme is to designate a bus master among the operational slave nodes during network initialization. Thus, rather than statically designating a particular node as the bus master, the first node to be turned on could become the bus master. Once a bus master takes control, it remains in control until it fails. If a bus master fails, another slave node may detect the failure and become bus master itself

The problem with dynamic allocation of mastership is that if two nodes are turned on almost simultaneously (within one bus propagation delay &TAU;pd), a conflict arises. Some arbitration mechanism must be invoked that designates one, and only one, bus master before proper network operation can proceed. This arbitration mechanism increases slave node complexity, and seems a high price to pay for a function that is only used when the system is reset. The arbitration mechanism is often complicated by the fact that collision detection circuitry is not available due to cost and practicality constraints.

Even with dynamic allocation of mastership, the current master still constitutes a single point of failure. If extra logic is included to facilitate automatic network resets and redesignation of a bus master, single-point failures can be minimized. However, the resultant node design is much more complicated than the original TDMA slave node.

3.4.6 Initial Allocation of Mastership with Stable Time Bases

A somewhat different approach is taken by the ARINC-629 protocol. In ARINC-629, there is no single bus master during normal operation. Rather than using a frame sync from a bus master, each node keeps track of time slices as they elapse, whether there are transmissions or not. Frame starts are not explicitly delineated by transmission events. In order to limit the effects of accumulated time-base skew between nodes, two cross-checked time sources are incorporated into each node, and nodes resynchronize at the end of every transmitted message. ARINC-629 implementations must ensure that messages are sent occasionally to avoid excessive timing skew, even with redundant oscillators at each node.

With ARINC-629, there must still be some initial synchronization event to start operation. This is done with an arbitration scheme that must deal with potential message collisions, just as in the case of dynamic allocation of mastership. The difference with the use of stable time bases is that after the initial master gains control (by issuing a non-colliding message that all other nodes synchronize to), mastership is then irrelevant for further operation.

The disadvantages of initial allocation of mastership with stable time bases are that logic for an initial arbitration scheme must be included, and very stable time bases must be used to minimize oscillator skew over the longest possible time between messages on the network. One could reduce the effects of oscillator skew by generating dummy messages periodically, but such messages would have to be sent sparingly to gain the benefits of ARINC-629's variable-width time slice feature that compresses unused time slices to increase efficiency. A dummy message scheme would also resemble a dynamically allocated master arrangement, with the attendant complexity increase and failure modes. A further problem with using stable time bases instead of frame syncs is that if a node is reset or brought on-line after the bus has started operation, there is no predictable reference for determining where the newly activated node's time slice begins; and no guarantee of how long the newly activated node will have to wait before some recognizable signal is transmitted by other nodes on the bus.

3.5 The J-TDMA Protocol

In order to avoid the problems of other TDMA protocols, we propose a new scheme that completely eliminates the need for a bus master. Because the protocol is based on using a "jam" signal as the frame sync, we call it J-TDMA.

Classically, a TDMA bus master's frame sync is used to avoid collisions among slave nodes by limiting accumulated timing skew. This resynchronization at the start of each frame, combined with a guard time at the end of each time slice, prevents slave node transmissions from overlapping. Typically, the frame sync signal consists of a unique waveform pattern such as an intentionally misplaced signal transition edge or a long sequence of ones that is otherwise illegal in a bit-stuffed transmission scheme.

The reason that collisions are undesirable in TDMA is that they are difficult or expensive to detect. If two transmitters were to attempt to send frame syncs concurrently, they might be enough out of phase to cause waveform interference between high and low physical signal levels on some or all of the bus as their signals propagate. This interference could cause some receivers to miss some or all of the frame sync message. Even on systems where such interference might not be a problem at the physical level, TDMA designs traditionally designate a bus master.

The key idea of J-TDMA is to use a nondestructive frame sync signal, so that more than one transmitter can send a frame sync without adverse interaction among signals. While most other TDMA protocols focus on having a single bus master issue frame syncs, J-TDMA is designed to tolerate multiple overlapping frame sync transmissions. Thus, the issue of establishing a unique bus master is rendered moot in J-TDMA.

An excellent candidate for such a frame sync signal corresponds to a "jam signal" used to enforce recognition of collisions in collision-based protocols . For example, in a baseband multimode fiber optic system, one or more transmitters can jam by emitting light (baseband "on") for a period of several bit times. As another example, current-mode transformer coupled systems can jam by having one or more transmitters assert a physical "high" value for longer than a bit time. In general, any signal which nondestructively propagates throughout the communication medium can serve as a jam signal. No data need be communicated by the jam signal only the presence of a jam signal need be detected in order to establish a synchronization event.

Detecting a jam signal is inherently easier than detecting a collision, because during jamming all transmitters are asserting mutually non-destructive waveforms, whereas during a collision between arbitrary data transmissions the waveforms may destructively interfere. Care must be taken to ensure that communications bus noise is unlikely to falsely trigger jam detection; in most cases this simply means that the jam signal must be longer than a bit time rather than shorter.

A jam signal is not the same as a "bit dominance" signal such as that used by the Controller Area Network (CAN) . In bit dominance a transmitter broadcasting a logical "1" must dominate over some other transmitter broadcasting a logical "0". For jamming, it is sufficient that transmitters not interfere with each other while each is broadcasting only a "high" level. So, bit dominance may be used to implement jamming, but jamming does not require a full bit dominance capability.

Once we decide to use a jamming signal as the frame sync, all questions of establishing a unique bus master disappear. It is perfectly acceptable for multiple nodes to be designated as bus masters and issue overlapping frame sync signals, because they won't interfere with each other; only a single elongated frame sync will be detected by any receiver on the bus. Furthermore, issues of implementing an arbitration mechanism to momentarily pick a bus master (as in the case of ARINC-629) also disappear. There truly never needs to be a unique bus master, because all nodes can assert frame syncs without concern for collision.

3.5.1 J-TDMA Protocol Description

J-TDMA follows the same time sequence shown in Figure 1.5 for TDMA. The major difference is that more than one node may issue overlapping frame sync signals. Figure 3.6 gives a Finite State Machine (FSM) diagram for the logic contained in each node when implementing J-TDMA. In the description we shall use "frame sync" to mean the logical operation of establishing a time reference, and "jam" to mean the physical act of transmitting a jam signal to implement a frame sync operation.

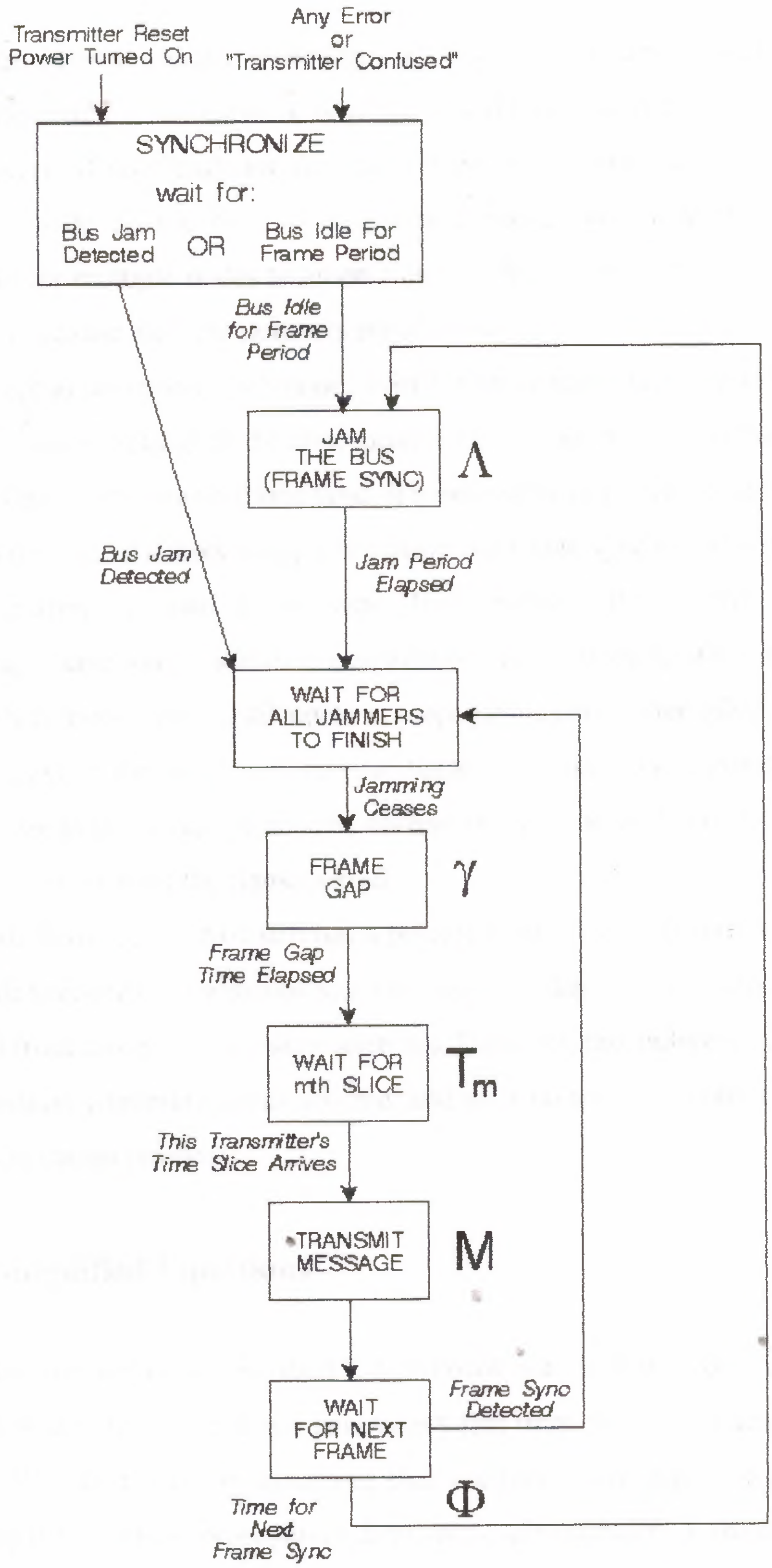


Figure 3.6. Finite State Machine for J-TDMA protocol.

Initialization is handled by having each newly activated node wait for an entire frame period to determine if the network is active. If a frame sync is detected, the node joins the active network. If no signals are detected for an entire frame period, then the node is the only active node on the bus, so it asserts a frame sync to start bus operation. It is permissible for multiple nodes to assert this first frame sync without arbitrating for initial mastership, because multiple jammers are allowed.

In normal operation, each node waits for its assigned slot interval beyond the frame sync, and sends a message at the appropriate time. It then waits until the anticipated end of the frame time, then emits a frame sync. If a node detects a frame sync before its computed end of frame time, it simply accepts the incoming frame sync signal as the start of a frame without emitting its own frame sync. It is possible that multiple nodes will start transmitting frame syncs within a propagation delay of each other, because they won't receive other frame syncs until up to a propagation delay after other nodes assert them. With this method, the nodes with fast oscillators will assert frame syncs, while other nodes resynchronize to them. As nodes come on-line and off-line, and components age, the fastest operating nodes will set the frame period.

In all fairness, JTDMA still has a potential single-point failure mode: jabbering. If a single node transmits a continuous jamming signal or data transmission, other nodes will be prevented from using the communication bus. However, this failure mode is inherent in any shared-medium communications scheme, and so is no worse a problem than that found in other media access protocols.

3.5.2 Simplified Equations

Now that we have described the protocol, we shall turn our attention to deriving equations to describe the various parameters that must be used to actually implement the protocol. We shall start by assuming that oscillators are highly accurate as a way of illustrating the effects of propagation delay on synchronization. In the next section we shall augment the equations with factors that account for oscillator skew found in real systems.

In systems using highly accurate oscillators, calculating the parameters of interest is relatively straightforward. In all cases we assume that computational delays at each node are negligible. Table 1 is a summary of our notation. All times are considered to be measured at each individual node unless otherwise noted.

$f_{min}, f_{nom}, f_{max}$ = oscillator frequency (minimum, nominal, maximum)
τ_{pd} = maximum signal propagation delay
Λ = minimum frame sync length
γ = frame gap
M = maximum message length (including any preamble and message gap)
T_m = start time of *m*th time slice after frame sync
Φ = frame length
σ = oscillator skew ratio
G = guard time within a time slice
V = guard time overhead ratio
Λ', G', T', Φ', and V' apply to optimal-length time slice scheme
Λ', G'', T'', Φ'', and V'' apply to fixed-length time slice scheme

Table 1. Summary of notation.

The minimum frame sync length &LAMBDA;, measured at node *m*, must account for a 2τpd worst case round-trip time in order to guarantee that potential signals from all jamming nodes (which can be skewed in starting time by up to one propagation delay) have time to reach all receivers (requiring a second propagation delay). This minimum jamming period ensures that the jamming will be received as an uninterrupted waveform at all receivers without gaps caused by skewed jam starting times.

$$\Lambda = 2\,\tau_{pd} \qquad (3.1)$$

We define the minimum time to the start of the *m*th time slice &TAU;m, to be measured, at node *m*, from the end of the collective frame sync waveform. Therefore, &TAU;m must take into account the frame gap length γ, and maximum message length M allowed within the system. There will be up to one τpd skew between nodes resulting from the travel time of the frame sync signal. Another τpd is required to prevent collision between the end of one message and the start of the next message caused by propagation delay between successively transmitting nodes. Both τpd terms must be added into each time slice so that every pair of transmitters has a 2τpd gap between transmissions; this accommodates the case where the transmitter that was first to jam (from a global perspective) is followed in transmission by the transmitter that jammed just short of a τpd later (also from a global perspective) where the two transmitters are at opposite ends of the communications bus.

$$T_m = y + m(2\,\tau_{pd} + M) \qquad (3.2)$$

The frame period &PHI;, measured at node *m*, is simply &LAMBDA; plus the time of the start of the *N*th time slice (where the highest transmitter number is N-1):

$$\Phi = \Lambda + T_N = 2\,\tau_{pd} + y + N\,(2\,\tau_{pd} + M) \qquad (3.3)$$

3.5.3 Optimal Time Slice Starting Times in the Presence of Oscillator Skew

In all real communication systems, the oscillator at each node operates at a slightly different frequency than the oscillators at other nodes. Most current systems, including TDMA systems, are relatively insensitive to slight inaccuracies in oscillator speed in normal operation. This insensitivity stems from the use of commonly available, high accuracy crystal oscillators (*e.g.*, +/-0.01% or better), and operation at moderately high data rates (*e.g.*, 1 Mbit/sec), resulting in a small cumulative skew between resynchronizations.

As distributed microcontrollers become more common in embedded systems, there will be cost pressure to accommodate significant amounts of time skew caused by use of less expensive crystal oscillators (*e.g.*, +/-0.1% or worse), relatively inaccurate RC-based on-CPU oscillators (*e.g.*, +/-10% or worse), and slower data rates over inexpensive communication channels. In order to address these issues we explicitly include skew introduced by oscillator speed inaccuracies in the following equations. Again, all times are measured independently at each individual node m.

We use a skew parameter, σ, to compensate for the differences among oscillator speeds within the communications network. The strategy used is to make all time delays long enough to accommodate the worst-case skew between any pair of nodes, thus avoiding overlapped transmissions (except during frame sync, when it is used to ensure overlapped transmissions without gaps). We define the oscillator skew ratio σ to be the inaccuracy of the oscillator:

$\sigma = f_{nom} / f_{min}$ where in general practice $f_{nom} / f_{min} > f_{max} / f_{nom}$ (3.4)

where the nominal system frequency (fnom) may vary to be as fast as (fmax) or as slow as (fmin) as specified by the oscillator manufacturer for expected system operating conditions. Specifications in real oscillators are typically given as nominal operating frequency combined with a skew percentage, leading to a slight asymmetry in skew ratios between the worst-case fast and slow oscillators. We assign σ the slightly higher value of the two (while we could have defined σ as the ratio of the maximum to minimum frequency, this definition would have been at odds with data sheet specifications).

The minimum frame sync length &LAMBDA;' must account for the possibility of some nodes having fast oscillators, and so uses the skew parameter σ to extend the length of the frame sync, ensuring that a small gap in the jamming waveform is not caused by oscillator speed differences. The minimum frame sync length &LAMBDA;' is given by:

$$\Lambda' = 2\,\tau_{pd}\,\sigma \qquad ; 1 \leq \sigma \qquad (3.5)$$

The start of the *m*th time slice &TAU;m', must use the skew factor σ to account for the different oscillator speeds at different nodes, because each node measures slightly different times for the frame gap and time slices. For convenience, we define α as:

$$\alpha = 2\,(\sigma - 1) \qquad ; 1 \leq \sigma << 3/2 \qquad (3.6)$$

α accounts for σ with a scaling factor of 2 to ensure that a node with a fast oscillator (σ times faster than nominal) does not start a message conflicting the end of a message from some preceding node with a slower oscillator (σ times slower than nominal). The range restriction on σ ensures physical realizability of the following equations. If σ is close to 3/2, skew time will account for most of the network bandwidth, and the system will be impractical.

Now we define the guard time, Gm', required for each time slice as:

$$G_{m'} = \alpha\,(T_{m'} + 2\,\tau_{pd} + M) + \alpha\,G_{m'} \qquad (3.7)$$

$$G_m' = \alpha / 1 - \alpha\,(T_{m'} + 2\,\tau_{pd} + M) \qquad (3.8)$$

This definition of guard time accounts for both the oscillator skew up to the start of the *m*th time slice, skew during the message (with 2τpd included as in Equation 2), and skew that occurs during the Gmth guard interval itself. Based on this guard time definition, we can iteratively define the start of each time slice &TAU;m' as:

$$(9) \qquad T_0' = \gamma + \alpha T_0' = \frac{1}{1-\alpha}\gamma$$

$$(10) \qquad T_m' = T_{m-1}' + 2\,\tau_{pd} + M + G_{m-1}' \qquad ; m > 0$$

The first time slice starts after the frame gap (and associated skew). The start of each time slice is delayed by increasingly long guard times as skew accumulates throughout the frame time. This has a closed-form solution of:

$$(11) \qquad T_m' = \left(\frac{1}{1-\alpha}\right)^{m+1}\gamma + (2\,\tau_{pd} + M)\sum_{i=1}^{m}\left(\frac{1}{1-\alpha}\right)^{i} \qquad ; m >= 0$$

The frame period &PHI;' is simply the frame sync &LAMBDA;' plus the time of the start of the *N*th time slice (where the highest transmitter number is *N*-1):

$$(12) \qquad \Phi' = \Lambda' + T_N'$$

This value of &PHI;' is optimal in the sense that it is the shortest possible frame period that safely accommodates worst case oscillator skew. We define the guard time overhead V' of this method to be:

$$(13) \qquad V' = \frac{\sum_{i=0}^{N-1} G_i'}{\Phi'}$$

V' provides a measure of the effect of skew on overall system efficiency.

3.5.3 Fixed-Length Time Slices in the Presence of Oscillator Skew

While the value of &PHI;' from the preceding section is optimal, this value may be difficult to achieve in practice, because it results in each time slice in the system being a different length (larger values of *m* are later in the frame, have more accumulated skew, and thus a longer guard time). It is easier to implement and perform configuration management on systems where the size of each time slice is constant (as in the traditional TDMA scheme). Therefore, we derive the equations for constant-size time slices below. Once again, all times are measured independently at each individual node *m*. A comparison of optimal-size slice *vs.* constant-size slice efficiency is given in the next section.

The starting time of the *m*th time slice is simply the sum of m constant-length time slices and the frame gap:

(14) $$T_m'' = \gamma + m\,(2\,\tau_{pd} + M + G'')$$

where the guard time G" for each constant-width time slice must account for accumulated skew over all N time slices plus the skew that accumulates during the guard time of each slice:

(15) $$G'' = \alpha T_N' = \alpha\left(\gamma + N\,(2\,\tau_{pd} + M + G'')\right)$$

(16) $$G'' = \frac{\alpha}{1 - \alpha N}\left(\gamma + N\,(2\,\tau_{pd} + M)\right) \qquad \text{where } \alpha < \frac{1}{N} << 1$$

and the frame length &PHI;" is:

(17) $$\Phi'' = A' + T_N''$$

Finally, we define the guard time overhead V" of this method to be:

(18) $$V'' = \frac{\sum_{i=0}^{N-1} G_i''}{\Phi''} = \frac{NG''}{\Phi''}$$

3.5.4 Simulation Results

We have developed an SES/workbench model to simulate a J-TDMA network. In particular, the model accounts for signal propagation delays along the bus and models time skew among nodes having different speed oscillators. The simulation also detects any unintentional collisions among node transmissions.

We do not report simulation throughput and latency results here, because the behavior of TDMA networks is well known. Rather, the purpose of the model was to help us validate the finite state machine and equations. The equations presented have been tested via simulation, including checking optimal-length time slices to verify optimality. Understanding and properly accounting for of clock skew and propagation delay effects was harder than we anticipated; the simulations proved a valuable tool in furthering our understanding of skew effects.

3.5.5 Variations to Improve Performance

Because J-TDMA is an improved frame synchronization mechanism for TDMA, most techniques for improving TDMA performance also apply to J-TDMA.
More efficient use of the bus may be made by assigning multiple time slices to nodes with heavy workloads. For example, if one node transmits twice as many messages as any other node, it may be assigned two slices instead of one. Of course this technique results in a minor increase in node logic complexity.

Another efficiency improvement technique is to use time slice compression, as done in ARINC-629. With this scheme, if a time slice goes unused for a predetermined period of time shorter than the entire slice time, all nodes automatically progress to the next time slice without any signaling taking place. This scheme compresses unused time slices to improve efficiency. In this case, the frame period varies with the number of messages actually transmitted.

3.6 Advantages and Disadvantages of TDMA

Advantages :

In addition to increasing the efficiency of transmission, TDMA offers a number of other advantages over standard cellular technologies. First and foremost, it can be easily adapted to the transmission of data as well as voice communication. TDMA offers the ability to carry data rates of 64 kbps to 120 Mbps (expandable in multiples of 64 kbps). This enables operators to offer personal communication-like services including fax, voiceband data, and short message services (SMSs) as well as bandwidth-intensive applications such as multimedia and videoconferencing.

1. Unlike spread-spectrum techniques which can suffer from interference among the users all of whom are on the same frequency band and transmitting at the same time, TDMA's technology, which separates users in time, ensures that they will not experience interference from other simultaneous transmissions.

2. TDMA also provides the user with extended battery life and talk time since the mobile is only transmitting a portion of the time (from 1/3 to 1/10) of the time during conversations.

3. TDMA installations offer substantial savings in base-station equipment, space, and maintenance, an important factor as cell sizes grow ever smaller.

4. TDMA is the most cost-effective technology for upgrading a current analog system to digital.

TDMA is the only technology that offers an efficient utilization of hierarchical cell structures (HCSs) offering pico, micro, and macrocells. HCSs allow coverage for the system to be tailored to support specific traffic and service needs. By using this approach, system capacities of more than 40-times AMPS can be achieved in a cost-efficient way.

5. Because of its inherent compatibility with FDMA analog systems, TDMA allows service compatibility with the use of dual-mode handsets.

Dual band 800/1900 MHz offers the following competitive advantages:

- Identical applications and services are provided to subscribers operating in both bands.
- Carriers can use the same switch for 800- and 1900-MHz services.
- Seamless interworking between 800- and 1900-MHz networks through dual-band/dual-mode phones.
- Using dual-mode, dual-band phones, subscribers on a TDMA 1,900 channel can hand off both to/from a TDMA channel on 800 MHz as well as to/from an analog AMPS channel.

Disadvantages :

1. One of the disadvantages of TDMA is that each user has a predefined time slot. However, users roaming from one cell to another are not allotted a time slot. Thus, if all the time slots in the next cell are already occupied, a call might well be disconnected. Likewise, if all the time slots in the cell in which a user happens to be in are already occupied, a user will not receive a dial tone.

2. Another problem with TDMA is that it is subjected to multipath distortion. A signal coming from a tower to a handset might come from any one of several directions. It might have bounced off several different buildings before arriving (see Figure 3.7) which can cause interference.

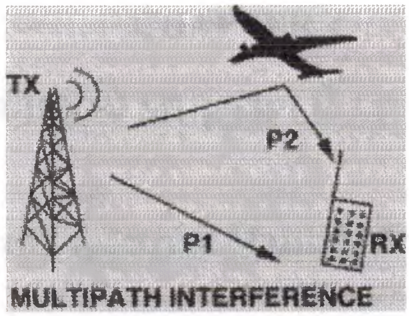


Figure 3.7 Multipath Interference

One way of getting around this interference is to put a time limit on the system. The system will be designed to receive, treat, and process a signal within a certain time limit. After the time limit has expired, the system ignores signals. The sensitivity of the system depends on how far it processes the multipath frequencies. Even at thousandths of seconds, these multipath signals cause problems.

All cellular architectures, whether microcell- or macrocell-based, have a unique set of propagation problems. Macrocells are particularly affected by multipath signal loss-a phenomenon usually occurring at the cell fringes where reflection and refraction may weaken or cancel a signal.

CHAPTER 4

CODE DIVISION MULTIPLE ACCESS (CDMA)

4.1 Evolution of spread spectrum Multiple Access communications

The history of spread-spectrum and multiple-access systems is intertwined with the history of electrical science in many intriguing and curious ways. At some times, developments have been shrouded in veils of secrecy, and at other times the latest ideas have been open to public scrutiny. Here is one engineer's sketch of this history, including the enabling technologies for communication, the emergence of multiple-access concepts and spread spectrum techniques, and events that have led to current developments in the commercialization of spread-spectrum signalling.

With out a doubt, even in the eighteenth century, electrical devices were envisioned as the enabling technology for communication systems that could operate at night and in bad weather, replacing the semaphore (visual telegraph). Communication may very well be the first major use of electrical technology. As concepts were refined and electrical telegraphy became a commercial success, inevitably the pressure to use resources efficiently led to many improvements, including the concepts of frequency-division and time-division multiplexing.

The most pressing need at the beginning of the twentieth century for the infant wireless technology was for mobile communication with ships at sea. The development of tuned circuits allowed different radios to asynchronously frequency-access the air waves with little or no interference. As more applications of radio technology were considered, the advantages of using more than the minimum bandwidth necessary to communicate an information-bearing signal were uncovered.

4.1.1 Mathematical Electricians

Oscillatory electrical phenomena were observed and predicted in the early- 19th century. In 1827 F. Savary conjectured about the discharge of a Leyden jar that "the electric motion during the discharge consists of a series of oscillations." Henry stated a similar conjecture in 1842. Hermann Von Helmholtz theoretically deduced the existence of oscillatory behavior of the discharge of a Leyden jar in 1847; the mathematical demonstration of this was given by William Thomson (later to be Lord Kelvin) in 1853 and experimental verification was provided by W. Feddersoh in 1859 This might be viewed as an early demonstration of the resistance, inductance, capacitance equations of electrical circuit theory.

In the mid-1840s, the insulating qualities of gutta percha in low temperature environments were discovered, and it finally was possible to effectively insulate submarine cable. By 1850 the first gutta-percha-coated cable was laid from Dover to Calais. In 1854, Michael Faraday demonstrated that such a cable "may be assimilated exactly to an immense Leyden battery; the glass of the jars represents the gutta-percha; the internal coating is the surface of the copper wire". This demonstrated that the electrostatic capacity must be considered in the theory of electrical signals on the wire. In the same year, correspondence between G. G. Stokes and Thomson analyzed the effect of capacity on submarine cable signals. In 1857 Gustav Kirchoff also took into account the self-inductance of the telegraph wire, and showed that an electrical disturbance is propagated along the wire with a specified velocity.

4.1.2 The development of wireless communication

The physical demonstration of propagation of electromagnetic waves in the weather, without the benefit of a conductor, is generally credited to Heinrich Hertz in 1888. While demonstrating an experiment with two short flat coils of insulated wires, Hertz noticed that the discharge of a small Leyden jar through one was able to induce currents in the other, provided that a small spark-gap was made in the first coil. In one experiment, Hertz had employed both an exciter of electromagnetic waves and a reasonably matched detector of such waves.

Remarkably, seven years earlier in 1879, David Hughes had successfully demonstrated virtually the same spark transmitter with a coherer (a receiving device employing metal filings, usually credited to Edouard Branly, ca. 1885-1891) and telephone as a receiving system separated by as much as a few hundred yards, but could not convince learned observers that it was wave propagation, not mere electromagnetic induction effects. Hughes never published his work.

4.1.3 Code Division Multiple Access (CDMA) Techniques

In a 1949 technical memorandum. John Pierce describes a multiplexing system in which a common medium carries coded signals that "need not be synchronized in any fashion." Its technological underpinnings probably evolved from pulse code modulation (PCM), although we might now classify this system as time-hopping spread-spectrum multiple-access. Pierce noted that as "... an increasing number of channels are transmitted over the medium, there is a gradual degradation of quality." Furthermore, Pierce credited Shannon as having earlier suggested that this sort of performance "could be obtained by using as 'code functions' voltages which are approximately orthogonal functions of time." It was declared that in a given frequency band, any number of noise functions could be found that were approximately orthogonal over a long enough time period. Many of these ideas later appeared in open publication in 1952 Undoubtedly, the quasi-orthogonality of two long noise signals was realized by engineers much earlier, e.g., possibly by Guanella when working on noise radar design

The two-user multiple-access communication model was carefully investigated in several information-theoretic papers during the early 1970' The result of this investigation was a description of the possible pairs of information rates that two transmitters might use to reliably communicate simultaneously with one receiver. Generally the resulting rate region exceeded what one might achieve with TDMA in this case. The effects of code asynchronism on the useful rate regions has been investigated in.

The first direct sequence spread-spectrum systems were built during the 1950's. As designers developed means for storing pseudo-random sequences and solving the synchronization problems associated with detecting the pseudorandom carriers, the concept of asynchronous code-division multiplexing (what is now called code-division multiple-access) on the air waves by independently operating radios was explored with these systems. Using Shannon's formula, the classic paper in 1959 by Costas evaluates the capacity of a spread-spectrum multiple-access (SSMA) system under the section title "The Question of Channels." Here he concluded that SSMA is better than FDMA over comparable bandwidths when the offered signal traffic is intermittent.

4.1.4 Commercialization of Spread-Spectrum Communications

The first extensive private mobile radio systems were used for public safety purposes, e.g.. New York City communications to harbor patrol boats in 1916, and the Detroit Police Department's patrol car communications, tested in 1921. The first public correspondence system for land mobile radio was created in St. Louis just after the Second World War. Almost immediately, the cellular radio concept was "verbalized in 1947 by D. H. Ring of Bell Laboratories in unpublished work." After thirty years of research and development contributions, the first cellular system (AMPS) in the USA went into developmental system trials. Certainly, the advantage of frequency reuse and the "roamer problem" of mobility tracking and handoff were uncovered in the design process. Even as AMPS was going into system trials in 1978, George Cooper and Ray Nettleton were suggesting spread-spectrum signalling as potentially providing a more efficient (i.e., traffic per unit bandwidth per unit area) cellular communication system than other signalling means

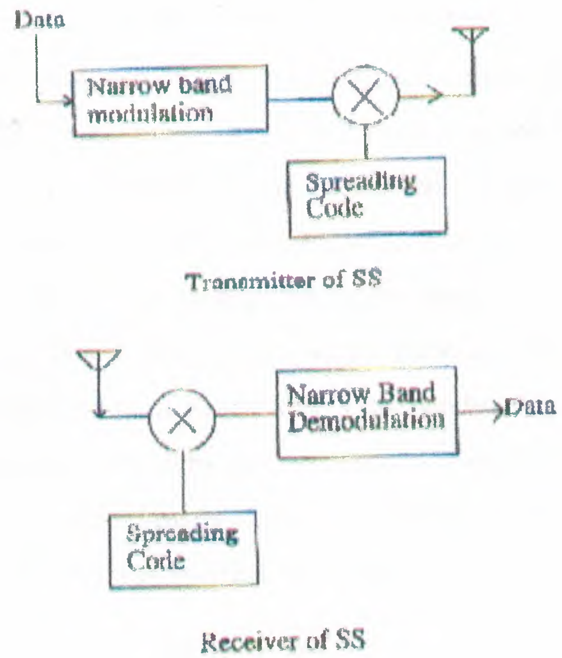


Figure 4.1 Spread Spectrum Communication System

4.2 What is a Spread Spectrum signal ?

In his brilliant treatise that established the field, Shannon called information theory the "mathematical theory of communication". We have often maintained that, in a very real sense, mathematics is definitions. Once the definitions are in place, all the lemmas, theorems and corollaries are determined; one has only to find them and prove them. If we wish to say something about the information theory of spread-spectrum systems, it follows that our unavoidable first task must be to define such systems. Of course, it is "signals" rather than "systems" that have spectra so that our task, more precisely formulated, is to define spread-spectrum signals. This task may well strike the reader as either superfluous or quixotic. Like the U.S. supreme court justice who admitted the difficulty of defining pornography but claimed that he knew it when he saw it, many communication engineers might maintain that a definition is not needed; they know a spread-spectrum signal when they see it. One such friend described a spread-spectrum communication signal to us as "one that uses much more bandwidth than it needs". There seems to be a certain coarse truth in this description, but it will hardly do for mathematical purposes. After some futile attempts to make this description more precise, our friend concluded that a satisfactory general definition of a spread-spectrum signal is not possible, which whetted our appetite to take a stab at formulating one. Every communication engineer is familiar with the ordinary notion of bandwidth, which we will call Fourier bandwidth both to honor the French pioneer in this field and to distinguish it from a less familiar but no less important type of bandwidth. The "sine pulse" m(t) = sinc(2Wt), where sinc(x) = sin(7t-x)/(7C-x), has a Fourier Bandwidth of W Hz, as one sees immediately from its Fourier transform M(f) shown in Fig. 1. For less dichotomous spectra, there are many options for calculating the precise Fourier bandwidth (rms bandwidth, 3 dB bandwidth, 99% energy bandwidth, etc.), but they are all roughly equivalent and any is good enough for our purposes. The notion of Fourier bandwidth extends 31easily from deterministic signals to stochastic processes (such as modulated signals) in a way familiar to all communication engineers.

$$m(t) = \mathrm{sinc}(2Wt)$$

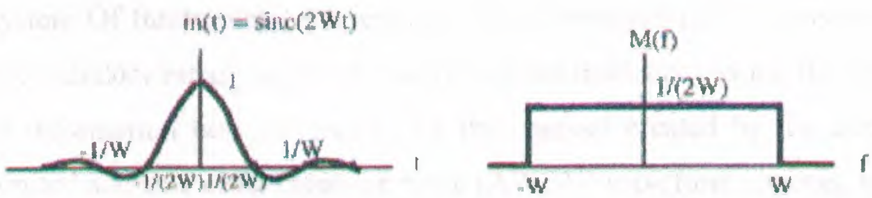


Figure 4.2 The Sine pulse and its Fourier Transform

The second type of bandwidth, which we will call Shannon bandwidth because Shannon was the first to appreciate its importance, makes no real sense for a deterministic signal since it always zero for a single time function. Non-zero Shannon bandwidth implies a "variable" signal (or a stochastic process) such as a modulated signal s(t) that can take on any of a multiplicity of time functions as its value. To determine the Shannon bandwidth of such a signal, one must in principle consider a signal-space representation of s(t) over some very long time interval, say the interval 0 S t < T. By this we mean that one must find orthonormal functions, so that one can represent (or very well approximate) every possible realization of s(t) by some choice of the coefficients si, s2,..., and SN in the linear combination

For $0 < t < T$. One says then that one has a signal-space representation of s(t) as a vector s = (si, S2,..., SN) in N-dimensional Euclidean space. When one does this in such a way as to minimize the dimensionality N of the signal space, i. e., to minimize the number of orthonormal functions used, then one has arrived at the Shannon bandwidth B.

Equivalently, the Shannon bandwidth is one-half the minimum number of dimensions per second required to represent the modulated signal in a signal space. [In earlier papers where we used the notion of Shannon bandwidth, we omitted the division by 2. Emboldened by Emerson's dictum that "a foolish consistency is the hobgoblin of small minds" we have now opted for the the factor 2 in the denominator, in order to avoid many such factors elsewhere.

4.3 Coding, Spreading and noise

It is time now to take a more strictly information-theoretic look at the advantages, if any, provided by spectrum spreading. To keep matters simple, we consider the single user system Of fundamental interest are the information rate, R, measured in information bits (i.e., random binary digits) per second at the modulator input; the capacity, also measured in information bits per second, of the channel created by the modulator and the band-limited additive white Gaussian noise (AWGN) waveform channel, which is the upper limit of rates R for which reliable (in the sense of arbitrarily small positive probability of error) recovery of the information bits is possible at the receiver when an appropriate coding system is used; the average power, S, of the modulated signal; the one-sided noise power spectral density. No, of the AWGN; the Fourier bandwidth, W, of the bandlimited AWGN waveform channel (which we take without loss of essential generality as equal to the Fourier bandwidth of the modulated signal, as there is no point in transmitting anything outside this band and, if one transmits in a smaller bandwidth, then one might as well reduce the channel bandwidth accordingly); and finally the Shannon bandwidth, B, of the modulated signal. Because W is the Fourier bandwidth of s(t), it follows from the fundamental theorem of bandwidth that $B < W$.

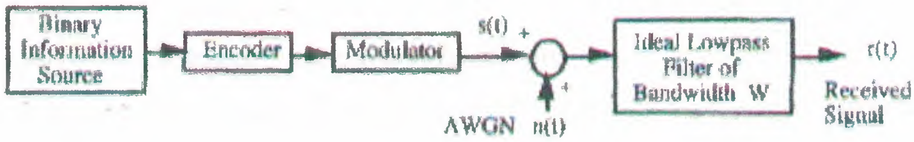


Figure 4.3 The single user communication system

The reader may be surprised to see the Shannon bandwidth B rather than the Fourier bandwidth W in this expression for C. But he or she will find that is precisely the equation that Shannon derives. It is easy to check that the right side of increases monotonically with increasing B; because B < W , with equality if and only if B = W, i.e., if and only if there is no spectrum spreading! The reason that Shannon wrote with an equality sign, rather than , in his final expression for the capacity of the AWGN channel is that he assumed that the choice of the modulated signal was up to the sender and that thus the sender would choose a signal with B = W to obtain (maximum) capacity.

Here we must in honesty point out that we have been somewhat cavalier in writing without stating the precise condition for which this expression gives the true capacity. This condition is that all the coefficients si, s2, ... SN in the expansion must be independently controllable by the choice of the modulator input symbols. This is indeed the case for all of the signals in the above six examples with the exception of the PPM modulated signal in example 3. We see from that in fact for this signal only one of the N = M coefficients can be non zero so they are certainly not independently controllable. For such modulation systems, the expression in gives only an upper bound on capacity-which is why PPM modulation is not "energy efficient" except for high "signal-to-noise ratios."

It is important not to draw the wrong conclusion from and . The real question is not whether spectrum spreading can increase capacity (it never can!), but whether spectrum spreading, which may be desirable for other reasons such as those considered in the previous section, necessarily entails a substantial loss of capacity for the used Fourier bandwidth W. This time the answer is more subtle and more interesting. As B increases without limit, the right side of tends to 1.44 S/No. Thus, with near equality when the Shannon bandwidth B (and hence also the Fourier bandwidth W) is very large. For instance, when the capacity given is at least 90% of C^g. As long as the Shannon bandwidth is large enough to satisfy, then no matter how large a spreading factor is used, the capacity will be at least 90% of the maximum capacity achievable with the given Fourier bandwidth W. Spectrum spreading cannot increase capacity, but it need not reduce it significantly.

4.4 CDMA System Model

In this section, a model of CDMA is described in order to clarify a problem in CDMA. For the sake of simplicity, a DS/CDMA system

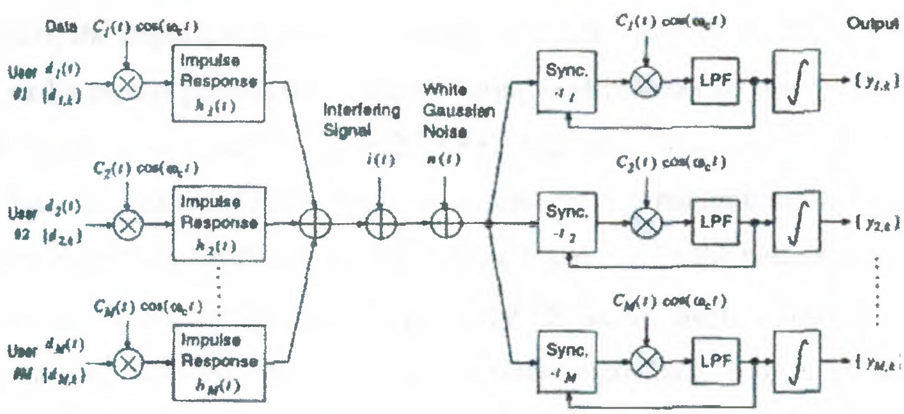


Figure 4.4 A Model of a DS/CDMA System

model is formalized.

Figure 4.4 shows a model of DS/CDMA system, where M users are trans- mitting individually spread spectrum signals Sm(t),'rn = 1,2, • • •,M at the same time and in the same frequency band. This system can also be considered as the reverse link or uplink of a single cell in a cellular mobile communication system

In order to spread a signal spectrum in a transmitter of the mth user, the message sequence signal dm(t) is directly multiplied by signature, spreading or pseudo-noise (PN) sequence signal Cm(t) which has a much higher symbol rate than dnt(t). Sm(t) can be written

U'p{t) is the unit rectangular pulse defined as Ur(t) == 1 for (0 < t < T) and (7r(<) = 0 otherwise. dm,k is a symbol of the message, sequence {dm,k} of the mth user at instant kTd, Cm,j is a symbol of the spreading sequence {Cn»j} assigned to the mth user at instant jT. In the case of BPSK, dm,k,C'mj € {+1, -1}. Here, T is the duration of a 'chip' in the spreading sequences, while TJ is that of a bit of message sequences. We assume that T^ is a multiple of T and define N = T^/T.

In a receiver, multiplexed SS signals from all users are received and multiplied again by the same spreading sequence signal used in transmitter, in order to transform the spread spectrum of the desired SS signal into its original narrow band.

where hm(t) is an impulse response of the transmission channel from the mth user to the receiver, i(t) is an interfering or jamming signal, n(t) is channel noise. In multipath fading channels, the impulse response of the mth user's link takes on the form

Each path's excess path delay, <m,A» i^8 organized in order of increasing magnitude with A such that tm,i == 0 and Tc < <m,2 < • • • < tm,Lm < ^'Moreover, each is uniformly distributed over the interval [7c,A] where A represents the maximum excess path delay possible. gm,x is the path amplitude fro the A-th path and 0m,\ is the path phase.

All the multiplexed SS signals except the desired SS signal interfere with the desired SS signal, due to the crosscorrelation of the different spreading sequences assigned to individual users, and is described as co-channel interference (CCI).

The output signal of the correlator for the nth user can be derived by integrating r(t)Cn(t) over Trf = NT. When spreading sequence acquisition is performed, the sample value y^ k of the output at instant kT^ can be represented in the form

In the first term of the right hand side indicates the desired base band data, the second term, the intersymbol interference (ISI) in the desired user's signal, the third, the co-channel interference (CCI) due to the undesired user, the fourth, the interfering signal component, and the fifth, the noise component. The magnitude of the co-channel interference is determined by the periodic or even crosscorrelation function

©n,m0') = Rn,mU)+Rn,m(J) and Qn,mU) = Rn,mU) - Rn,mU) depend-

ing on dm,k = dm,k-l and d^,k = -dm,k-l, respectively.

In the presence of CCI, the crosscorrelation between the spreading sequences of desired and undesired users prevents acquisition and tracking of the spreading sequence and restricts the number of users simultaneously accessing the channel. This is because side lobes of the output signal of the correlator hamper the detection of the main lobe of auto correlation and the side lobes increase with the number of available users.

4.5 Multi user detection for CDMA

Note from the above-discussion that CCI due to crosscorrelation of spreading sequences hampers establishment of acquisition and tracking, and limits the capacity of CDMA. CCI results in the "near-far " problem that relates to the problem of very strong signals from undesired users at a receiver swamping out the effects of weaker signals from desired user. Therefore, CDMA receivers should be designed so as to improve the synchronization and the capacity against the near-far problem.

4.5.1 Single user receivers and Multi user receivers

CDMA receiver structures can be classified as either multi-user or single-user structures in general. The conventional single-user detector assigned to each user is described in the previous section. Another example is the multipath combining receiver by Lehnert. Optimum single-user detector was proposed by Poor. The con- ventional single-user receiver decides the correlator output for each user only in the interval corresponding to the data symbol individually among multiple-access users. In the decision, CCI is considered as an undesired component like noise.

From an information theoretical viewpoint, however, we can utilize CCI as redundant information which multiple-access users are sharing in a common channel or a multi-user channel, in order to afford performance gains over the conventional single-user receiver. The receiver by which all multiple-access users' signals can be correctly detected using CCI is termed the multi-user receiver. The multi-user receiver can be used at a cell site, that is a base station, to detect all multiple-access users' signals.

In contrast with the single-user receiver, for multi-user receivers, the decision of all the users' data is an inter-dependent process. Althoug their structures are much more complicated, their BER performance can greatly exceed that of the conventional correlator. These receivers vary in complexity from the optimum multi-user detector presented independently by Kohnn and Verdu to the suboptimum detectors developed independently by Kohno Masamura

4.5.2 Optimal Multi user receiver

Without loss of generality and for the sake of notational simplicity, can be rewritten if there is no intersymbol interference or multi-path distortion in a channel, no narrowband interference and only two users accessing the channel.

It is noted from these equations that the correlator outputs y_1,k and y_2,k can be considered as multi-dimensional finite-state machine outputs corrupted with noise or multi-dimensional analogue convolutional encoder output with noise. Figure 4.5 illustrates its equivalent multi-dimensional tapped delay lines. Therefore, we can achieve optimum multi-user detection for DS/CDMA Channel (two users)

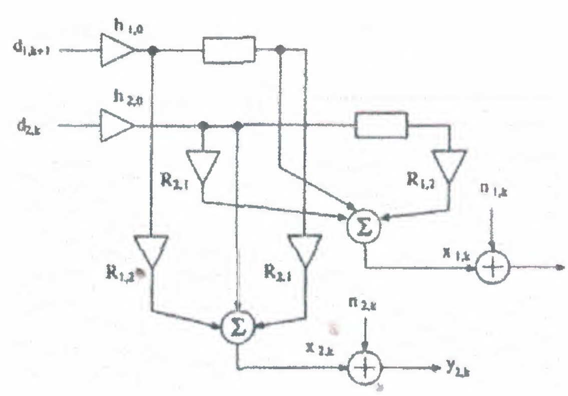


Figure 4.5 An Equivalent Tapped Delay Line Model of Asynchronous

CDMA by using a Viterbi algorithm that is well known as an efficient maximum likelihood sequence estimation (MLSE) for a convolutional code, a trellis coded modulation and so on. In general, the optimum.

4.6 CDMA Applications

4.6.1 Key elements in designing a CDMA system

The design of a CDMA system is much more sophisticated than the design of a TDMA system. In analog and TDMA systems, the most important key element is C/I. There are two different kinds of C/I. One is the measured (C/I) which is used to indicate the voice quality in the system. The higher the measured value is the better. The other is called the specified (C/I)s which is a specified value for a specified cellular system. For example, the (C/I)s in the AMPS system is 18 dB. Since in analog and TDMA systems, due to the spectral and geographical separations, the interference (I) is much lower than the received signal (C), sometimes we can utilize field strength meters to measure C to determine the coverage of each cell. The field strength meter therefore becomes a useful tool in designing the TDMA system. In CDMA all the traffic channels are served solely by a single radio channel in every cell([1,2,3,4]). Therefore, in an w-voice channel cell, one of the m traffic channels is the desired channel and the remaining m-l traffic channels are the interference channels. In this case, the interference is much stronger than the desired channel. Then C/I is hard to obtain by using the signal strength meter which will receive more interference than the desired signal. Thus, the key elements in designing a CDMA system are different from the key element in designing a TDMA system.

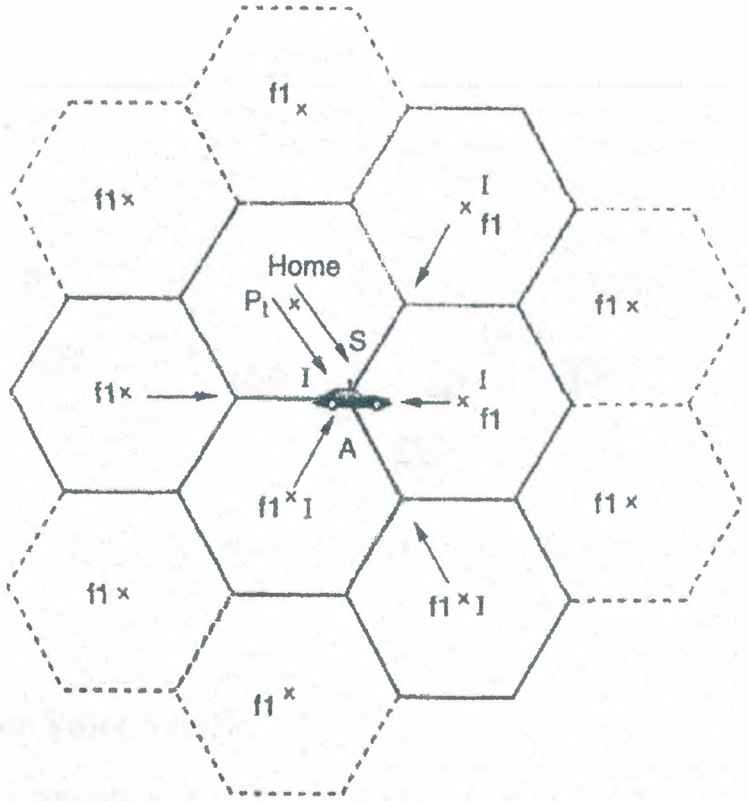


Figure 4.6 CDMA System and its Interference (From a Forward Link)

4.7 System model

The CDMA network may have any generic architecture. The basic ingredients of our work are applicable to terrestrial (cellular, PCN) or satellite networks. Figure 1 shows an example of a CDMA network with voice and data traffic.

The voice call or data message of each user admitted in the network is packetized with the same fixed length packet. Time is divided into slots of duration equal to the transmission of one packet. In our models, packet transmissions start at common clock instances and packets have constant length. The typical packet length is 1000 to a few thousand bits.

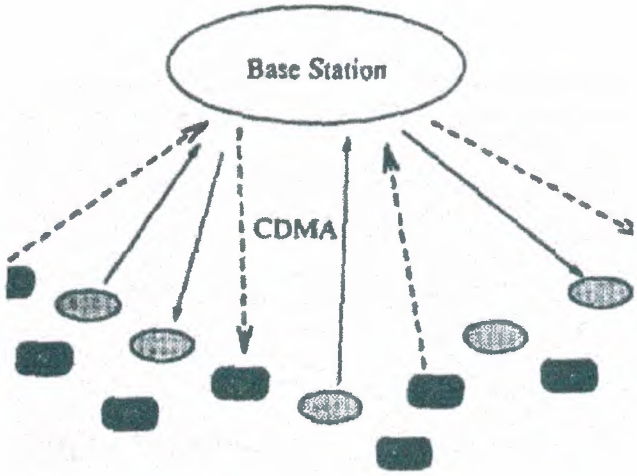


Figure 4.7. Model for Mobile CDMA System

4.7.1 Model for Voice Traffic

Let *Ny* be the total population of voice traffic in the CDMA system. The traffic generated by each voice user is modeled as a three-state (idle, silent, and talkspurt) discrete-time Markov processes each with transition probabilities *p^*, *p^y*, *p^* and *p^*, as shown in Figure 4.8 Figure 4.9 is a two state Markov chain, which is the voice model for the system with different priorities of voice and data users.

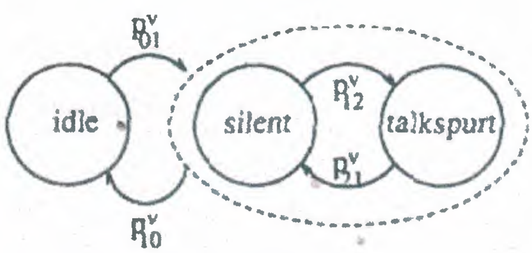


Figure 4.8. Model for Voice Users

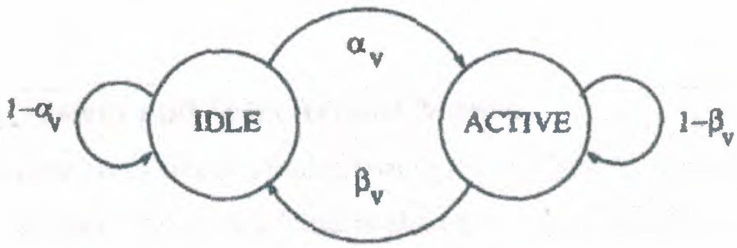


Figure 4.9 Model for Voice Users

4.7.2 Model for Data Traffic

Let N4 and N^ be the total population of two types of data traffic. Each data user of high priority is modeled by a two-state (OFF/ON or idle/active) Bernoulli process, shown in Figure 3 with transition probabilities p^ andpfo- Similarly, the mean duration of the idle and active periods of data users are 1/p^ and 1/pfg F$^{01\text{,tne}}$ $^{\wedge\text{ov/}}$ priority data traffic we assume the number of arriving data packets follows a Poisson distribution with mean rate ^. There is some justification for using different models for the high and low priority data users. The high priority data users are treated in the same manner as voice users and the finite population assumption is essential in deriving the optimal allocation strategy. On the other hand the low priority data users are assigned only the codes left unused by all other traffic (voice and data), no optimization takes place, and a Poisson population model simplifies our analysis and admission policy; we could also use a Bernoulli model for the low priority users, but that will complicate matters. Further, when the population of data users is large or when they have packets to transmit with small probability, the binomial model approaches the Poisson model. Finally, we assume that the data streams of all voice and data users are statistically independent.

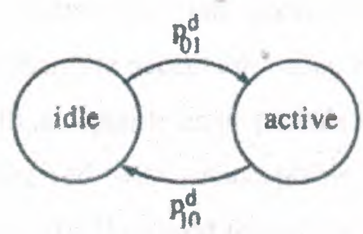


Figure 4.10 Model for Data Users

4.7.3 CDMA System and Interference Models

Direct-sequence code-division multiple-access (DS/CDMA) is employed by all users in (he network. The same frequency band is shared by voice and data traffic. Voice and data traffic have the same data rate; thus the same pool of CDMA codes is used. Each user (data or voice node) employs a distinct code for the transmission of its packets.

We define the multiple-access capability (MAC) index Ky as the number of voice users that can be accommodated simultaneously, such that the expected packet error probability of voice traffic remains below a specified threshold. Similarly, the MAC index K^ and K^ for data users is the number of data users that can transmit simultaneously with a tolerable packet error. Thus we have

$$P_E(k) < P_E^{V} \ \forall \ k < K_v \qquad (4.1)$$

$$P_E(k) < P_E^{d1} \ \forall \ k < K_{d1} \qquad (4.2)$$

$$P_E(k) < P_E^{d2} \ \forall \ k < K_{d2} \qquad (4.3)$$

are the maximum tolerable voice and data packer error probabilities, respectively, and PEW is the packet error probability in the presence of k simultaneous packet transmissions, where k includes both voice users and active data users. In practice, P^ > max \P^, P^\, and, therefore, Ky > max \K^.,K^ \. Here we also assume P^ ^P^, so that K^ <i K^ ; this assumption can be easily relaxed and any relationship between P^ and P^2 can be accommodated. If the total number of simultaneous users is k ^ KJ then all k data or voice packets are received with acceptable error probabilities. If K^ <k^K^ , then, among the k packets, the voice packets and the data packets with lower priority are received with acceptable error probabilities, whereas the data packets with higher priority are received with error probabilities higher than the acceptable value P^ If K^ < k < Ky then only the voice packets are received with acceptable error probabilities. Finally, if Ky < k, then all voice or data packets are received with unacceptable error probabilities. The model described above is referred to as the threshold model; its usefulness and simplicity lies in characterizing acceptable operating conditions in terms of maximum allowable numbers of simultaneous users. Shown in Figure 4.10 are the boundaries (moving) for allocation of CDMA codes to voice and low priority data users.

For the multiple-reception of data packets both the threshold model and the graceful degradation model can be used. According to the latter model, there is a non-zero probability of correct reception for any arbitrary number of packets (even for numbers exceeding the MAC index), which depends on the total number of simultaneously transmitted packets (data and voice) in the CDMA network. This takes the form{l succesful data packets |m voice and n data packets are transmitted).

4.7 Advantages of CDMA

A number of advantages are:

- Low power spectral density. As the signal is spread over a large frequency-band, the Power Spectral Density is getting very small, so other communications systems do not suffer from this kind of communications. However the Gaussian Noise level is increasing.
- Interference limited operation. In all situations the whole frequency-spectrum is used.
- Privacy due to unknown random codes. The applied codes are - in principle - unknown to a hostile user. This means that it is hardly possible to detect the message of an other user.
- Applying spread spectrum implies the reduction of multi-path effects.
- Random access possibilities. Users can start their transmission at any arbitrary time.
- Good anti-jam performance.

CONCLUSIONS

Wireless communication has passed through several generations. It is a new technology which will have a large impact on data communication, satellite communication and in networking also.

The biggest issue with wireless protocols today seems to be the efficiency of business functions the protocols can deliver and the adoption of these functions by users. Wireless media has been undergoing a rapid innovation process in search for a reliable, simple and business-viable solution to consumer demands for fast, easy, and inexpensive information access. Over the last five years, a number of wireless protocols have been developed and a variety of application vendors have begun to ship wireless products to the market. Although the specific content of the new service is still under consideration, the service is most certainly become popular among younger users of wireless systems. It is also foreseeable to have an MP3 player and a stereo integrated with a wireless phone so that consumers can download audio and hear it with only one device you already carry with you. As Java-enabled I-mode phones and next-generation mobile communications system, known as W-CDMA

Advances in technology and competitive access are also driving the revolution towards wireless access infrastructure for the provision of basic telephone service. The communication based on the infrared has many advantages over that on the radio as the infra red standard provides a communication interface with a 1- and 4- Mb/s transmission speed and it has a low error bit rate about 10^{-9} bit error rate with in a 1 m range. The Radio based communications media has received more attention than infrared in future developments of the wireless world. Some major benefits of the radio based media, is that it does not require a free point of view between receiver and transmitter, and upcoming standards allow much higher transfer rates within the radio based media than the infrared.

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