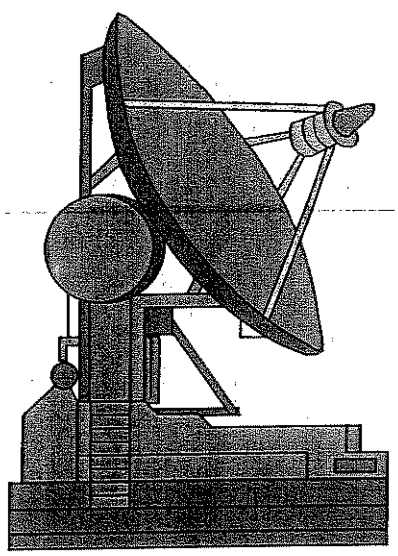
Telecommunication & Navigation Institute

Transmission Section



رمز القسم
TR-151

Telecommunication II

TR 151

رقم المقرر

First Edition on ; 2007

1.500

معـــهــد الاتـــصـــالات و المـــلاحـــة

Telecommunication & Navigation Institute

الخطة الدراسية للمقرر النظري

Theoretical course Lesson Plan

Transmission	Dept. القسم	Telecommunication - 11	Course Name اسم المقرر
4	Weekly Hrs. عدد الساعات الأسبوعية	TR 151	Course Code رمز المقرر

Course Objectives أهداف المقرر

On completion this course the trainee will know :

- The different types of modulation and principles
- Analogue and digital access technology
- The Types of noise affect the signal
- Multiplexing Technology
- CDMA-1 and CDMA-2000 Systems

Weeks	Content
1	Introduction noise effect
2	Analogue modulation (AM & FM & PM)
3	Digital modulation (ASK & FSK & QPSK)
4	Pulse modulation
5	PCM system & SDH
6	Multiple access in radio
7	MID –TERM EXAMINATION
8	TDMA &SDMA & FDMA technology
9	CDMA-1 (IS-95)
10	CDMA -2000 technology
11	CDMA-2000 advantages
12	CDMA-2000 1x and EV-1x Applications
13	Radio structure using CDMA technology

Weeks	Content
14	Final Review
15	موعد بداية الاختبارات النهـائيـة Final Exams
16	

موعد اختبار منتصف الفصل

Mid-term Exam Date

20 / /

يبلغ المدرب/المدرس جميع المتدربين بالموعد

توزيع الدرجات الكلية والتي مجموعها **100** درجة

Grading Policy (Total of 100)

الاختبار الفصلي Final Exam	% 60
اختبار منتصف الفصل Mid-term Exam	% 20
الاختبارات القصيرة و التطبيقات Quizzes & Applications	% 10
النشاط Classroom/Lab Activities	% 5
الانتظام Attendance	% 5

ختم القسم

Dept. Seal

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Chapter 1

Modulation Principles

1. Introduction

Telecommunication is the extension of communication over a distance. In practice it also recognizes that something may be lost in the process; hence the term 'telecommunication' covers all forms of distance and/or conversion of the original communications, including radio, telegraphy, television, telephony, data communication and computer networking.

The elements of a telecommunication system are a transmitter, a medium (line) and possibly a channel imposed upon the medium (see base band and broadband as well as multiplexing), and a receiver. The transmitter is a device that transforms or encodes the *message* into a physical phenomenon; the *signal*. The transmission medium, by its physical nature, is likely to modify or degrade the signal on its path from the transmitter to the receiver. The receiver has a decoding mechanism capable of recovering the message within certain limits of signal degradation. In some cases, the final "receiver" is the human eye and/or ear (or in some extreme cases other sense organs) and the recovery of the message is done by the brain (see psychoacoustics.)

Telecommunication can be point-to-point, point-to-multipoint or broadcasting, which is a particular form of point-to-multipoint that goes only from the transmitter to the receivers. One of the roles of the telecommunications engineer is to analyze the physical properties of the line or transmission medium, *and* the statistical properties of the message in order to design the most effective encoding and decoding mechanisms. When systems are designed to communicate through human sense organs, (mainly vision and hearing); physiological and psychological characteristics of human perception will be taken into account. This has important economic implications and engineers will research what defects may be tolerated in the signal yet not affect the viewing or hearing experience too badly.

Examples of Human (tele) communications

In a simplistic example, consider a normal conversation between two people. The message is the sentence that the speaker decides to communicate to the listener. The transmitter is the language areas in the brain, the motor cortex, the vocal cords, the larynx, and the mouth that produce those sounds called speech. The signal is the sound waves (pressure fluctuations in air particles) that can be identified as speech. The channel is the air carrying those sound waves, and all the acoustic properties of the surrounding space: echoes, ambient noise, and reverberation. Between the speaker and the listener (the receiver), might be other devices that do or do not introduce their own distortions of the original vocal signal (e.g. telephone, HAM radio, IP phone, etc.) The penultimate receiver is the listener's ear and auditory system, the auditory nerve, and the language areas in the listener's brain that will "decode" the signal into meaningful information and filter out background noise.

All channels have *noise*. Another important aspect of the channel is called the bandwidth. A low bandwidth channel, such as a telephone, cannot carry all of the audio information that is transmitted in normal conversation, causing distortion and irregularities in the speaker's voice, as compared to normal, in-person speech.

Other Background

Bell Labs scientist Claude E. Shannon published *A Mathematical Theory of Communication* in 1948. This landmark publication was to set the mathematical models used to describe communication systems called information theory. Information theory enables us to evaluate the capacity of a communication channel according to its bandwidth and signal-to-noise ratio. Original theory on communication principles was provided by Harry Nyquist and Émile Baudot after whom the term Baud was conceived to represent a single piece of transmitted information.

Early telecommunication systems were predominantly based on analog electronic circuit design and used a single encoding technique. The introduction of mass-produced digital integrated circuits has enabled telecom engineers to take full advantage of information theory and simultaneously use multiple encoding techniques. From the demands of telecom circuitry, a whole specialist area of integrated circuit design has emerged called digital signal processing.

Early phone systems used analog transmission lines between central offices, but in the 1960s digital multiplexed circuits were used to send voice calls over a Time Division Multiplexed (TDM) circuit. This was done at speeds of either 1.544 Mbps (a T1), or at 2.04 Mbps (an E1). A T1 circuit was capable of carrying 24 voice channels while an E1 was capable of carrying 30 voice channels. Each voice channel uses 64 Kbps worth of digital bandwidth to convey the analog waveform. The development of the computer modem from 1980 is a clear testimony of increases in information transfer capability through the use of multiple mechanisms. A modem today uses frequency, phase and data compression techniques to squeeze data through what originally seemed an impossibly small bandwidth.

Possible imperfections in a communication channel are: shot noise, thermal noise, latency, non-linear channel transfer function, sudden signal drops, bandwidth limitations, signal reflections (echo). More recent telecommunications systems take advantage of some of these *imperfections* to actually improve the quality of the channel.

Modern telecommunication systems often make extensive use of a clock signal which is used to decode a transmitted data stream, synchronization. In order to accumulate and manage such streams a "*Telco*" always provided the clock signal. With the advent of global communications it became necessary to have a single worldwide standard derived from a master atomic clock, or to secondary clocks synchronized to that clock. Synchronous circuits are often used between routers. Asynchronous Transfer Mode, ATM is a relatively new standard, operating at very high bit rates where synchronization outside of the data stream can result in errors.

Examples

Examples of digital channel coding systems: Hamming coding, Gray coding, Binary coding, Turbo coding.

Examples of telecommunications systems:

- Semaphore.
- Telegraphy.
- Radio teletype.
- The global telephone network (also known as the Public Switched Telephone Network or PSTN).
- Radio.
- Television.
- Communications satellites.
- Ethernet.
- Predictive Dialers.
- The Internet.

1.1 Noise

Consisting of undesired, usually random, variations that interfere with the desired signals and inhibit Communication, noise originates both in the channel and in the communication equipment. Although it cannot be eliminated completely, it's effect's can be reduced by various means. It is helpful to divide noise into two types: **internal noise**, which originates within the communication equipment, and **external noise**, which is a property of the channel.

External noise consists of man-made noise, atmospheric, and space noise. Man-made is generated by equipments that produce sparks, such as automobile engines and electric motors with brushes. Also, any equipment with fast rise-time voltage or current Gran generates interference, like light dimmers and computers. Atypical solution for a computer, for instance, involves shielding and grounding the case and all connecting cables and installing aloe-pass filter on the power line where it enters the enclosure.

Atmospheric noise is often called *static* because lightning, which is a static-electricity discharge, is its principal soured. Since it occurs in short, intense bursts with relatively long periods of time between bursts, it is often possible to improve communication by simply disabling the receiver for the duration of the burst. This technique is called noise blanking. **Space noise** is mostly solar noise, which can be a serious problem with satellite reception when the satellite is in line between the antenna and the sun. It is more important at higher frequencies because most of the space noise at lower frequencies is absorbed by the upper atmosphere. On the other hand, atmospheric noise dominates at lower frequencies.

Internal noise generated in all electronic equipments, both passive components like resistors and cables, and active devices like diodes and transistors. Thermal noise is produced by the random motion of electrons in a conductor due to heat. It is equal mixture of noise of all frequencies, and sometimes called white noise, by analogy with white light, which is an equal mixture of all colors. The term noise is often used alone to refer to this type of noise, which is found everywhere in electronic circuitry. The noise power in a conductor in function of its temperature, as shown by equation:

$$P_N = kTBW$$

Where P_N = internal noise power in watts

K = Boltzmann's constant, 1.38×10^{-23} joules/Kelvin (j/k)

T = absolute temperature in Kelvin (k)

BW = operating bandwidth in Hertz

The temperature in degrees Kelvin can be found by adding 273 to the Celsius temperature. The previous equation shows that noise poser is directly proportional to bandwidth, which means that high bandwidth communications are associated with higher noise. The only way to reduce noise is to decrease the temperature or the bandwidth of a circuit, or both. Amplifiers used with very low signal levels are often cooled artificially to reduce noise. The technique is called *cryogenics* and may involve, for example, cooling the first stage of a receiver for audio astronomy by immersing it in liquid nitrogen. The other method of noise reduction, bandwidth reduction, will be referred to many times throughout this book. Using a bandwidth greater than required for a given application is simply an invitation to problems with noise.

Shot noise has a power spectrum that resembles that for thermal noise by having equal energy in every Hertz of bandwidth, at frequencies from dc into the GHz region. It is created by random variations in current flow in active devices such as transistors and semiconductors diodes. *Excess noise*, also called *flicker noise* or *pink noise*, varies inversely with frequency. It is rarely a problem in communication circuits, because it declines with increasing frequency and is usually insignificant above approximately one kHz.

The main reason for studying and calculating noise power or voltage is the effect that noise has on the desired signal. In analog systems, noise makes the signal unpleasant to watch or listen to, and in extreme cases, difficult to understand. Once noise and distortion are present, there is usually no way to remove them. In addition, the effects of these impairments are cumulative: noise will be added in the transmitter, the channel, and the receiver; and if the communications system involves several trips through amplifiers and channels, as in a long-distance telephone system, the noise will gradually increase with increasing distance from the source.

In digital transmission of analog signals, the conversion of infinitely variable analog signal to digital form introduces error. This will inevitably result in the loss of some information, and the creation of a certain amount of noise and distortion.

In communications, it is not really the amount of noise that concerns us, but rather the amount of noise compared to the level of the desired signal. That is, it is the *ratio* of signal to noise power that is important, rather than the noise power alone. This **Signal-to-Noise Ratio (SNR)**, usually expressed in decibels (dB), is one of the most important unit for specifications of any communication system. The decibel is a logarithmic unit used for companies of power levels or voltage levels. In order to understand the implication of dB, it is important to know that a second level of zero dB corresponds to the threshold of hearing, which is the smallest sound that can be heard. (See Appendix I)

A normal speech conversation would measure about 60 dB. The SNR is given by the following equation:

$$\textbf{SNR (dB)} = 10\log_{10}\left(\frac{P_S}{P_N}\right)$$

Where P_S is the signal power

P_N is the noise power

Example:

A receiver has an input power of 42.2 mW while the noise power is 33.3 µW. Find the SNR for the receiver.

Solution:

$$\textbf{SNR (dB)} = 10\log_{10}\left(\frac{P_S}{P_N}\right)$$

$$= 10\log_{10}\left(\frac{42.2}{.0333}\right)$$

$$= 31.03 \text{ dB}$$

Typical values of SNR range from about 10 dB for barely intelligible speech to 90 dB or move for compact-disc audio systems. A SNR of zero dB would mean that the noise has the same power as the signal, which would be absolutely unacceptable for any transmission system. Another quantity that is used to determine the signal quality is the ***noise figure*** **(NF)** also called the ***noise factor,*** which is related to the noise ratio (NR). These can be computed by using the following equation:

$$\mathbf{NR} = \frac{(SNR)_{input}}{(SNR)_{output}}$$

Where (SNR) $_{input}$ is the signal –to-noise at the input.

(SNR) $_{output}$ is the signal -to-noise at the output

$$\mathbf{NF = 10 \log NR}$$

Therefore,

$$\mathbf{NF\ (dB) = SNR_{input}\ (dB) - SNR_{output}\ (dB)}$$

Example:

Suppose the SNR at the input amplifier is 25 dB. Find the SNR at the amplifier output. Assume that NF = 10 dB.

Solution:

$$NF(dB) = SNR_{input}\ (dB) - SNR_{output}\ (dB)$$

$$SNR_{output}\ (dB) = SNR_{input}\ (dB) - NF\ (dB)$$

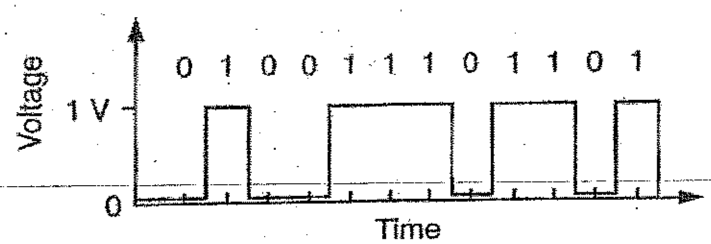
$$= (25 - 10)\ dB$$

$$= 15\ dB$$

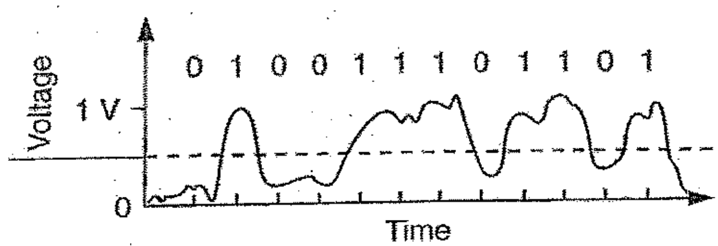
An amplifier or receiver will always have more noise at the output than at the input because the amplifier or receiver generates internal noise, which will be added to the signal. And even through the signal may be amplified, that noise will be amplified along with it. Since the SNR at the output will be less than the SNR at the input, the noise figure will always be greater than 1. A receiver that contributes zero noise to the signal would have a noise figure of 1, or 0 dB; but such a noise figure is not attainable in practice. The lower the noise figure, the better the amplifier.

Data and voice signals exhibit entirely different tolerance to noise. Data signals may be satisfactory in the presence of white noise, but the same can be bothersome to humans. On the other hand, impulse noise (clicks, pops, or sometimes frying noise) will destroy a data signal on a circuit but might be acceptable for speech communication.

Digital systems are not immune from noise and distortion, but it is possible to reduce their effect. Consider the simple digital signal shown on figure1-1. Suppose that a transmitter generates (1) V for binary one and (0) V for a binary zero. The receiver examines the signal in the middle of the pulse, and has a detection threshold at (0.5) V; that is, it considers any signal with amplitude greater than (0.5) V to be a one, and any amplitude less than that to represent a zero. This is achieved mainly by a *quantizer* circuit at the receiver end, whose function is to determine whether the incoming digital signal has a voltage level corresponding to binary (0) or binary (1). The basic design concern is to minimize the impact on channel noise at the receiver.



(a) Digital signal as transmitted



(b) Received signal with some noise and distortion

Figure 1-1: Removal noise and distortion from digital signal

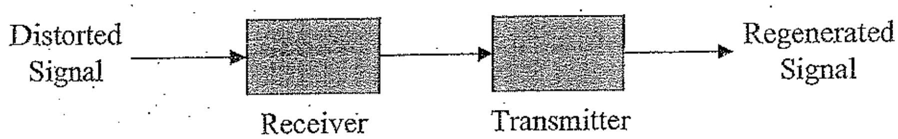
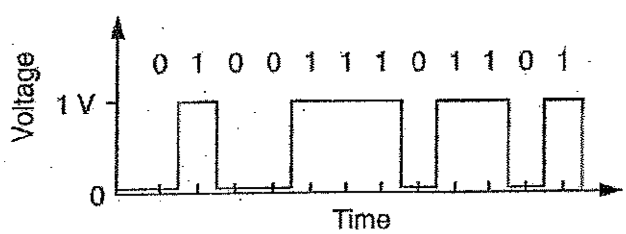
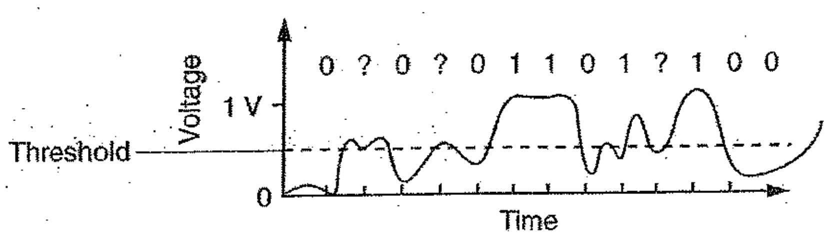


Figure 1-2: Digital repeater

Figure 1-3 (a) shows the signal as it emerges from the transmitter and Figure 1-3 (b) shows it after its passage through a channel that adds noise and distorts the pulse. In the spite of the noise and distortion, the receiver has no difficulty deciding correctly whether the signal is a zero or a one. Since the binary value of the pulse is the only information is the signal, the distortion has had no effect on the transmission of information.



(a) Digital signal as transmitted



(b) Received signal with excess noise and distortion

Figure 1-3: Excessive noise on a digital signal

The received signal of Figure 1-2 could now be used to generate a new pulse train to send further down the channel. The receiver-transmitter combination, which is called a *repeater* and illustrated in Figure 1-2 has not only avoided the addition of any distortion of its own, but has also removed the effects of noise and distortion that were added by the channel preceding the repeater. Unfortunately, since noise is random, it is possible for a noise pulse to have any amplitude, including one that will cause a transition to the wrong level. Extreme distortion of pulses can cause errors as demonstrated in Figure 1-3. Errors can never be eliminated completely, but, by judicious choice of such parameters as signal levels and bit rates, it is possible to reduce the probability of error to a very small value. There are even techniques to detect and correct some of the errors.

While signal-to-noise ratio is used as a performance measure for analog systems, the **bit error rate** (BER) is a prime factor is a digital system. It is the number of bits in error expressed as a portion of transmitted bits. For example, a BER of 10^{-9} (which equals $1/10^{9}$) means one bit is in error for each one billion bits received.

1.2 Modulation

Modulation is a means of controlling the characteristics of a signal in a desired way. The modulation is done at the transmitter, while an inverse process, called demodulation or detection, takes place at the receiver to restore the original baseband signal. There are many ways to modulate a signal, such as Amplitude modulation (AM), Frequency modulation (FM), Phase modulation (PM), and Pulse modulation. Both AM and FM are used is radio broadcast. Pulse modulation is mainly used for analog-to-digital conversation. In modulation, amplitude, frequency, or phase of a carrier wave is changed in accordance with the modulation signal in order to transmit information. The resultant is called a modulated wave. This concept is illustrated in figure 1-4.

A carrier, which is usually a sine wave, is generated at a frequency much higher than the highest modulating signal frequency.

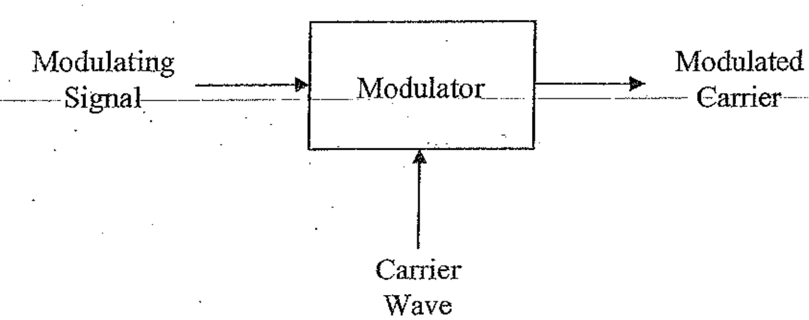


Figure 1-4: Concept of modulation

1.2.1 Amplitude Modulation

AM is one of the oldest and simplest forms of modulation used for analog signals. In AM, an audio signal's varying voltage is applied to a carrier. Its amplitude changes in accordance with the modulating voice signal, while its frequency remains unchanged. This is principle is shown in Figure 1-5.

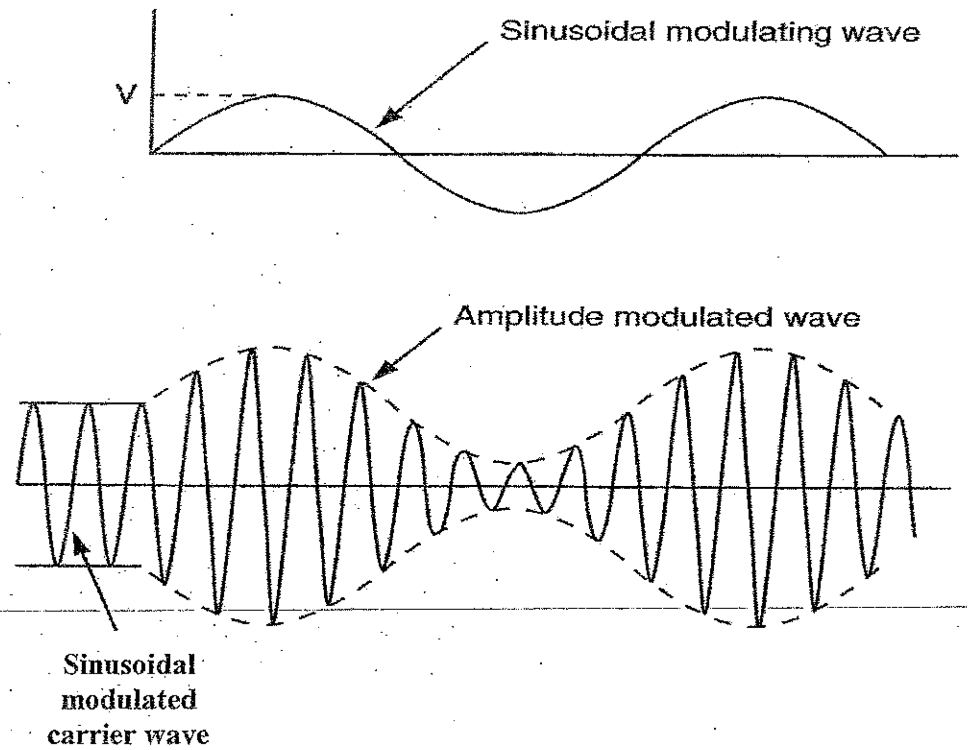


Figure 1-5: Amplitude modulation

1.2.2 Frequency Modulation

In FM, frequency of the carrier changes in accordance with the amplitude of the input signal, but its amplitude remains unchanged as shown in Figure 1-6. This makes FM modulation more immune to noise than is AM and improves to-the overall signal–noise ratio of communications system. Since the amplitude (voltage) stays the same, the output power of a FM signal is constant, unlike the varying AM power output. However, the amount of bandwidth necessary to transmit a FM signal is greater than that necessary for AM - a limiting Constraint for some systems. Also, the circuits used for FM are much move complex than those used for AM.

As an example, let us consider a carrier frequency, also called Center frequency; of 1 MHz. assume that because of FM modulation the center frequency is made to deviate 75 kHz by the audio baseband signal. This change from center is the frequency deviation, which in this example, + 75 kHz or 150 KHz. The 75 KHz deviation is for the loudest audio signal with the greatest amplitude in the baseband modulating signal. The FM radio broadcast band is 88 to 108 MHz, with stations spaced every 200 kHz or 0.2 MHz. examples of a carrier frequencies are 92.1, 96.3, and 104.5 MHz. the 200 KHz spacing between carrier frequencies is needed to allow for atonal swing of 150 KHz, with a guard band of 25 KHz on each side to prevent interference between adjacent stations.

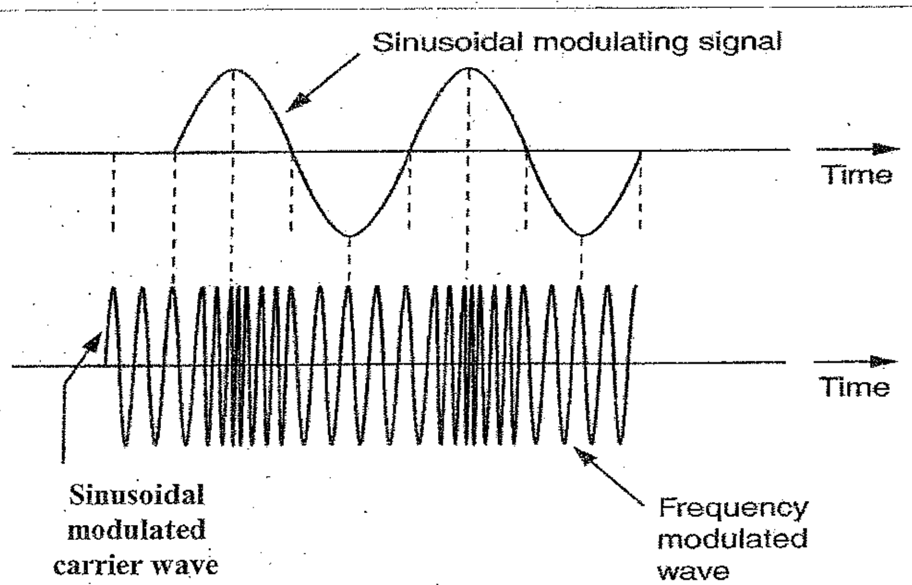


Figure 1-6: Frequency modulation

1.2.3 Frequency Shift Keying

Frequency shift keying (FSk) is a popular implementation of FM for data applications and was used in low-speed modems. A carrier is switched between two frequencies –one for mark (logic 1) and the other for space (logic 0) –as indicated in Figure 1-7. There are always guard bands that reduce the effects of bleed over between adjacent channels, there are two pairs of mark and space frequencies. All these frequencies are well inside crosstalk between the sidebands that are generated by modulation. This technique is not applicable for high-speed modems and is rarely used. Beside modems, FSK has applications for digital communication via high-frequency radio waves. Here, the system specifies the frequency shift between mark and space for alerter frequency. So when a mark (logic 1) is transmitted, the center frequency may be lowered, for example, by 42.5 Hz, and when a space (logic 0) is transmitted, the center frequency may be raised by 42.5 Hz. Thus, if the center frequency is 425 Hz, while a space represents 467.5 Hz. This process is called FSK.

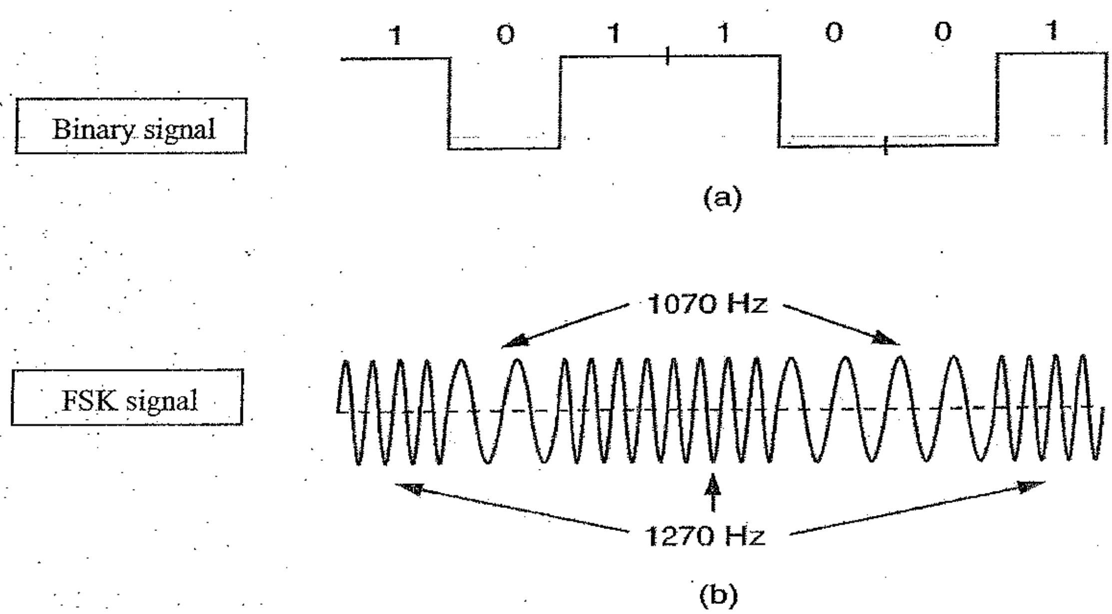


Figure 1-7: Frequency-shift keying

1.2.4 Phase Modulation (PM)

In PM, the amount of phase-shift of the carrier changes in accordance with the modulated signal; in effect, as the amount of phase-shift changes, the carrier frequency changes. Since PM results in FM, it is often referred to as indirect FM. Phase shift is a time difference between two sine waves of the same frequency. Figure 1-8 illustrates several examples of phase shift. Note that a phase shift of 180° represents the maximum difference and is also known as phase reversal. The advantage of using PM over FM is that the carrier can be optimized for frequency accuracy and stability. This type of modulation is easily adaptable to data or digital application.

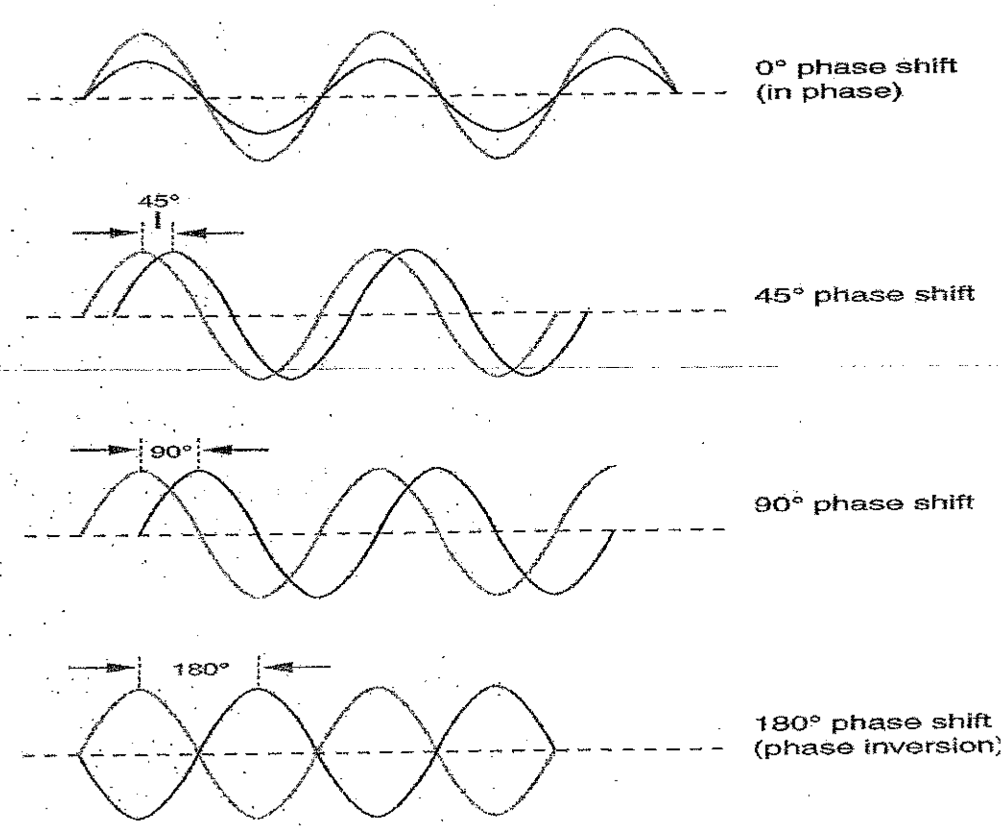


Figure 1-8: Examples of phase shift

1.2.5 Phase Shift Keying (PSK)

Phase stiff-keying (PSK) is the most popular implementation of PM for data applications. In PSK, the binary signal, 0 or 1 to be transmitted changes the phase shift of a Sine wave accordingly. Figure 1-9 illustrates the simplest form in of PSK known as binary PSK (BPSK). During the time that a binary 0 occurs 'the carrier signal is transmitted with one phase, but when binary 1 occurs, the curvier signal is transmitted with 180° phase shift. The main problem with BPSK is that the speed of data transmission is limited in a given bandwidth. One way to increase the binary data rate; while not increasing the bandwidth requirement for the signal transmission is to encode more than one bit per phase change. Most PSK modems use Quadrature PSK (or 4 - PSK), where each symbol represents two bits, as illustrated in Figure 1-9.

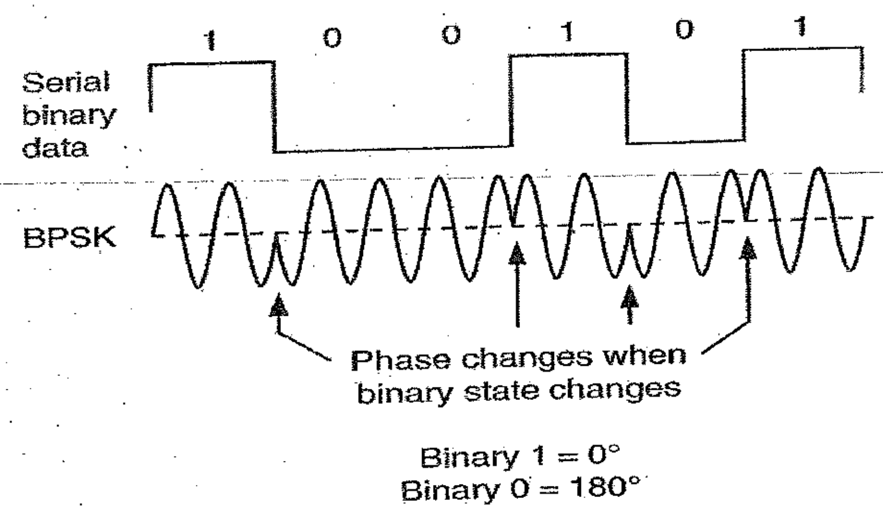


Figure 1-9: Binary phase shift keying (BPFK)

Baud rate is defined as the number of symbols (or signal transitions) transmitted in one second. The following equation gives the relationship between the baud rate and the bit rate.

Bit rate = Baud rate x Bit per symbol

Example:
Find the transmission bit rate if the baud rate is 1200 and there are two bits per symbol or signal transmission.

Solution:
Bit rate = Baud rate × Bits per symbol
Therefore, Bit rate = 1200 × 2
Bit rate = 2400 bps

1.2.6 Quadrature Amplitude Modulation (QAM)

A Quadrature Amplitude Modulation (QAM) modem uses two amplitude-modulated carriers with a 90° phase angle between them. These are added to produce a signal with amplitude and phase that can vary continuously. The number of amplitude-phase combinations could be infinite, but a practical limit is reached when the difference between adjacent combinations becomes too small to be detected reliably in the presence of noise and distortion. For example, the V.32bits modem has a modulation rate of 2400 baud and 14,000 bps (14.4 kbps), where each signal transition represents six data bits, as shown in Figure 1-10. The term *bit* comes from Latin, meaning *second*; in other words, the second and enhanced release of the standard. Third releases are designated, translated from Latin as *third*. The V.90 modem has a potential top speed of 56.6 kbps, but the FCC prohibits the 56 kbps modems from operating above 53.3 kbps to prevent excessive crosstalk in local loop cable bundles. High-speed modems make use of data compression techniques in reduce the number of bits that must pass over the communications medium in order to reduce transmission time.

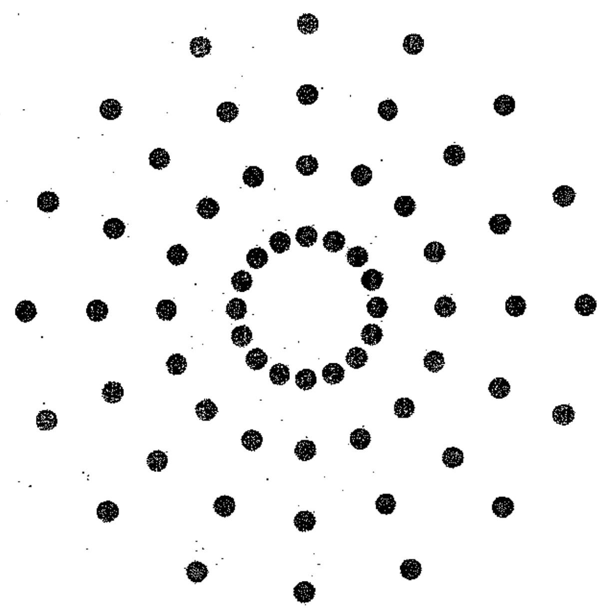


Figure 1-10: V.32 bit 64-point signal constellation

1.3 Pulse Modulation

Pulse modulation, which includes a variety of schemes, is used for both analog and digital signals. For analog signals, the process involves sampling where a snapshot (sample) of the waveform is taken for a brief instant of time, but at regular intervals. These instantaneous amplitudes are the sample values, or samples, of the signal waveform. The rate at which a signal is sampled is called the sampling rate, and it is expressed as the number of the samples per second. The sampling interval is the time interval between each sample. The sampling rate is the reciprocal of the sampling intervals.

In 1928, Henry Nyquist determined the optimum sampling rate. The *Nyquist sampling theorem* states that if a waveform is sampled at a rate at least twice the maximum frequency component in the waveform, then it is possible to reconstruct that waveform from the periodic samples without any distortion. Therefore, if the maximum frequency component in the signal is F_{max}, then the optimum sampling rate equals $2F_{max}$. The sampling rate is sometimes called the Nyquist frequency. If a signal has a maximum frequency component of 5 KHz, then the sampling rate is 10,000 KHz, which is the same as 10,000 samples per second. The sampling process converts as analog signal into a train of pulses of varying amplitude but at a constant frequency.

Analog-to-digital Conversation consists of three stages:

- ☐ The first stage is a low-pass filtering of the analog signal, called an anti-aliasing filter, to prevent any alias frequencies from appearing due to under-sampling of an unexpected high frequency. Aliasing, a penalty for a sampling rate that is too low, is a form of distortion in which the reconstructed original signal results in a lower-frequency signal.
- ☐ The second stage is the sampling of the analog signal at the Nyquist rate, the result of which a series of pulses at the Nyquist sampling rate with amplitudes equal to the sample values. These pulses represent a Pulse Amplitude Modulation (PAM) signal.
- ☐ The third stage transforms these pulses into a digital signal. The amplitude of the pulses is quantized, and the quantized values are coded as binary numbers. The binary numbers become a stream of on-off pulses. A number of pulses together then represent a binary

number. The process of encoding analog samples as a series of on-off pulses is referred to as Pulse Code Modulation (PCM).

1.3.1 Pulse Amplitude Modulation (PAM)

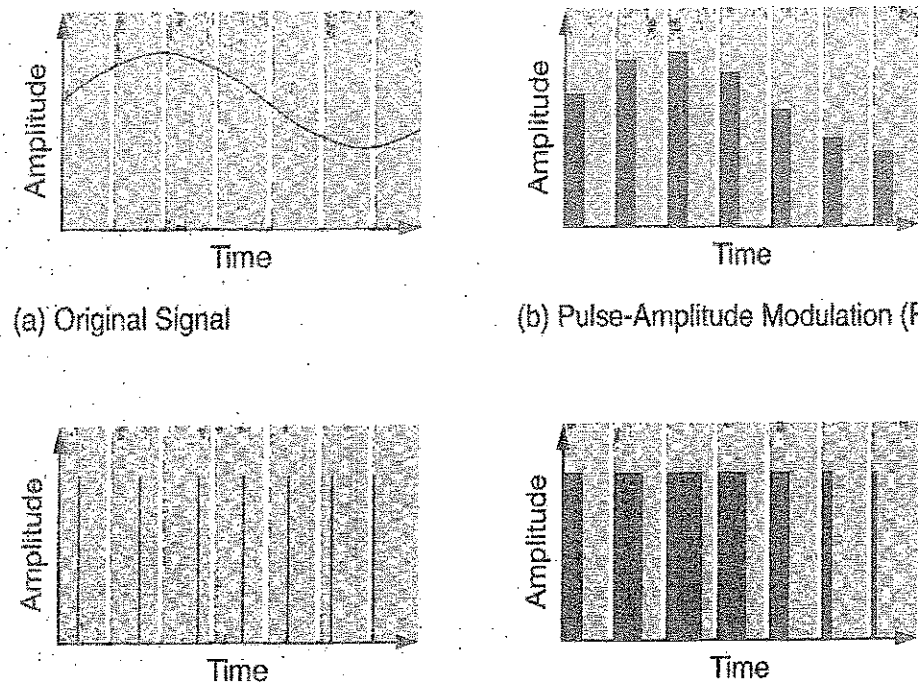
Pulse Amplitude Modulation (PAM) generates pulses whose amplitude variation corresponds to that of the modulating waveforms, as shown in Figure 1-11. Like AM, it is very sensitive to noise. While PAM was deployed in early AT&T Private Branch Exchange, there are no practical implementations in use today. However, PAM is an important first step in PCM.

1.3.2 Pulse Position Modulation (PPM)

Pulse Position Modulation (PPM) is closely related to PWM. All pulses have the same amplitude and duration but their timing varies with the amplitude of the modulating signals, represented in Figure 1-11c. The random arrival rate of pulses makes this unsuitable for transmission.

1.3.3 Pulse Width Modulation (PWM)

The Pulse Width Modulation (PWM) technique generates pulses at a regular rate, whose length or width is controlled by the modulation signals as depicted in figure 1-11 d. PWM is unsuitable for transmission because of the varying pulse-width.



(a) Original Signal (b) Pulse-Amplitude Modulation (PAM)

(c) Pulse-Position Modulation (PPM) (d) Pulse-Width Modulation (PWM)

Figure 1-11: Analog pulse modulation

1.4 Pulse Code Modulation (PCM)

Pulse Code Modulation (PCM) is the only technique that renders itself well to transmission. it is the most commonly used of coding digital signals and also used for transmitting telephone (analog) signals digitally. For analog signals, the amplitude of each sample of a signal is converted to a binary number. A common pattern for coding the transmitted information is by using a *character code* such as ASCII. A character code specifies a unique string of 0s and 1s t identify a character. The receiver detects the pattern of 0s and 1s in a given period of time; it interprets the transmitted code by finding the corresponding character represented by it. The frequency range that can be represented through PCM modulation depends upon the sampling rate.

T-1 carrier uses PCM as depicted in figure 1-12. The allotted bandwidth per voice channel is 4 KHz. According to the Nyquist theorem, an analog signal must be sampled at twice its highest frequency to obtain an accurate digital representation of the information content of the signal. Therefore, the voice channel must be sampled at 8 KHz. A pulse code modulator samples the voice 8,000 times every second, converts each sample to an eight-bit word, and transmits it over a line over a line interspersed with similar digital signals from 23 other channels. Each PCM voice channel operates at 64 kbps (8 bit/sample and 8000 sample/sec). Repeaters spaced at appropriate intervals regenerate the 24-channel signal with an aggregate of 1.536 Mbps (equals 24 X 64 kbps). With additional 8 kbps for synchronization, this technique results in a 24-channel 1.544 Mbps digital signal known as T-1. Each of the 24 channels can be used for either data or digital voice communication.

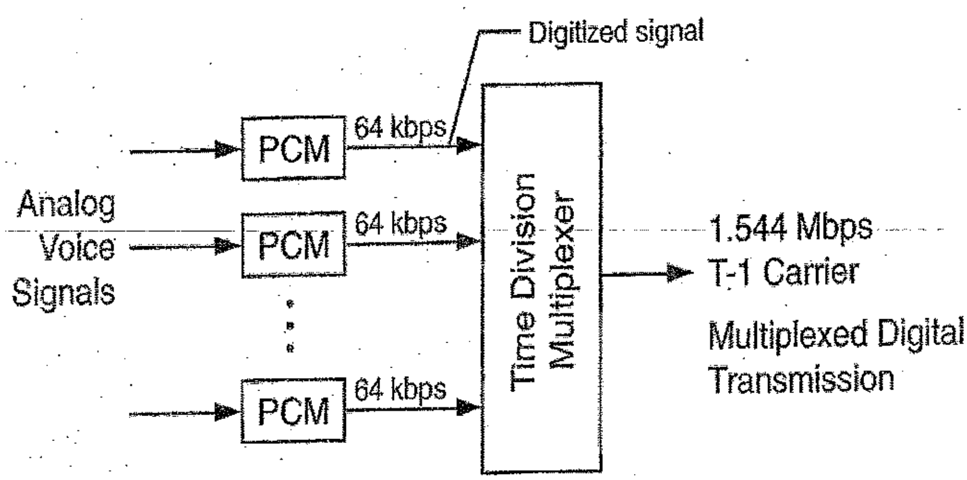


Figure 1-12: PCM and TDM applications for a T-1 carrier

1.5 Pulse Code Modulation & SDH

SDH (Synchronous Digital Hierarchy) is an international standard for high speed telecommunication over optical/electrical networks which can transport digital signals in variable capacities. It is a synchronous system which intends to provide a more flexible, yet simple network infrastructure.

SDH (and its American variant- SONET) emerged from standard bodies somewhere around 1990. These two standards create a revolution in the communication networks based on optical fibers, in their *cost* and *performance*.

With some 750 million telephone connections is use today and the number on internet users continuing to grow rapidly, network providers have been faced with the task of trying to deal effectively with increased telephone traffic. In response to the growing market needs, a number of methods and technologies have been developed within the last 50 years to address these market needs in as economical a way as possible.

In the field of communications engineering, this resulted in the introduction of frequency division multiplex (FDM) systems whereby each individual telephone channel was modulated with a different carrier frequency. The signal could then be shifted into different frequency ranges enabling several telephone connections to be transmitted over a signal cable.

With the advent of semiconductor circuits and the continuing demand for telephone capacity, a new type of transmission method, pulse code modulation (PCM) was developed in the 1960s.

With PCM (multiple use of a single line by means of digital time-domain multiplexing), the analog telephone signal is first sampled at a bandwidth of 1.3 kHz, quantized and encoded then transmitted at a bit rate of 64 kbps. When 30 such coded channels are collected together into a frame along with the necessary signaling information, a transmission rate of 2048 kbps is used.

This is known as the primary rate and is used throughout the world with the exception of the USA, Canada, and Japan, where a primary rate of 1544 kbps (formed by combining 24 channels) is used.

The demand for greater bandwidth however, meant that more stages of multiplexing were needed throughout the world. A practically synchronous digital hierarchy was developed in response. As there are slight differences in timing signals, justification or stuffing is necessary when forming the multiplexed signals.

Inserting or dropping an individual 64-kbps channel to or form a higher digital hierarchy however requires a considerable amount of complex and expensive multiplexer equipment.

Towards the end of the 1980s, the synchronous digital hierarchy (SDH) was introduced, paving the way for a worldwide, unified network structure. SDH is ideal particularly for network management system that can be easily adapted to accommodate the demand for "bandwidth-hungry" applications and services.

1.6 Why SDH?

With the introduction of PCM technology in the 1960s, communications networks were gradually converted to digital technology during the years that followed. To cope with the demand for ever-higher bit rates, a multiplex hierarchy or plesiosynchronous digital hierarchy (PDH) evolved. The bit rates start with the basic multiplex rate of 2 Mbps with further stages of 8.34 and 140 Mbps. This fundamental difference in developments made the set up of gateways between the networks both difficult and expensive. In response to the demand for increased bandwidth, reliability, and high-quality service, SDH developed steadily during the 1980s eliminating many of the disadvantages inherent in PDH. In turn, networks providers began to benefit from the many technological and economic advantages this technology introduced including the following:

- High transmission rates:

  Transmission rates of up to 10 Gbps can be achieved in modern SDH making it the most suitable technology for backbones – the superhighways in today's telecommunications networks.

- Simplified add & drop function:

  Compared to the older PDH system, low bit rate channels can be easily extracted from and inserted into the high-speed bit streams in SDH. It is now no longer necessary to apply the complex and costly procedure of de-multiplexing then re-multiplexing the plesiosynchronous structure.

- High availability & capacity matching:

  With SDH, network providers can react quickly and easily to the requirements of their customers. For example, leased lines can be switched in a matter of minutes. The network provider can use the standardized network elements (NE) than can be controlled and monitored from a central location via a telecommunications management network (TMN) system.

- Reliability:

Modern SDH networks include various automatic back-up circuit and repair mechanisms which are designed to cope with system faults and are monitored by management. As a result, failure of a link or an NE does not lead to failure of the entire network.

- Future-proof platform for new services:

SDH is the ideal platform for a wide range of services including POTS, ISDN, mobile radio, and data communications (LAN, WAN, etc.). it also able to handle more recent services such as video on demand and digital broadcasting via ATM.

Review Questions:

Q1 : Complete the following **:**

A) The main parts of Telecommunication system are :

-
-
-
-

B) The main types of Telecommunication systems are :

-
-

C) The imperfections in communication CH's are :

-
-
-
-
-

D) Mention Examples of communication systems :

-
-
-
-

Q2 : Tick (√) or (x) for the following and *correct the mistakes*

1- The thermal noise in communication equipment called external noise ( )

2- The property of the CH related to the internal noise ( )

3- The atmospheric noise happened due to solar effect ( )

4- The white noise affects in satellite Frequency band ( )

5-The space noise is a solar noise affect in SHF Band ( )

Q 3 : Solve the following problems

A) A received signal with power 60.5 mw at the Rx. Input while the noise power is 25 μw calculate the SNR of the receiver in [dB]

B) Suppose the SNR at the amplifier input is 15 dB calculate the SNR at The receiver output assume the NF is 6 dB

Chapter 2

Channel Access Technologies

2. ANALOG VERSUS DIGITAL ACCESS

Until recently, most wireless data transmitted through radio communications have been analog. In analog systems, the actual sound of the seller's voice pattern is transmitted over the airwaves by means of a continuous wavelike signal. Next generation wireless systems have turned toward the use of digital signals. Digital systems have several advantages, including allowing better coverage, more calls per channel, broadband communications, less noise interference, and the ability to add new features and functions. Analog and digital technologies and their salient features are identified in the following table:

System Generation	Technology	Operation Frequency	Advantages	Disadvantages
First Generation	AMPS based on FDMA	800 MHz or 1800 MHz	Widest Coverage including rural areas	▪ Poor security ▪ Not optimized for data ▪ Limited capacity
Second Generation	TDMA	800 MHz or 1900 MHZ	▪ Better security ▪ Higher capacity	May experience an interruption during handoff
Third Generation	CDMA	800 MHz or 1900 MHZ	▪ Very high security ▪ High capacity ▪ Greater immunity ▪ Soft handoff with no interruption	▪ Limited coverage at this time.

Table 2-1: Analog & Digital Technologies

2.1 Analog Access

The Analog cellular systems are referred to as first generation cellular technologies. The North and South American analog cellular systems confirm to the AMPS standard, which operates on the 800 MHz or 1800 MHz frequency band. But there are several other types of analog systems standards in the rest of the world. In Europe and Asia, these include Total Access Communications System (TACS), North Mobile Telephone (NMT), and others. The AMPS and TACS are both based on FDMA.

2.2 Digital Access

Until now, digital wireless technologies have developed mainly because of the need for increased system capacity for voice transmission. However, this trend is now changing. During the last three years, there has been a rapid growth in multimedia communications via the internet, which has resulted in an increased demand for high-bandwidth transmissions rather than dedicated voice transmissions, even in the wireless world. Another important factor is the increasing need for global coverage.

North America currently has multitude of digital cellular technologies for wireless ratio communication. Collectively referred to as Personal Communication System (PCS), they operate sat 1900 MHz, each with different coverage and capacities try definitions, analog cellular technology is not included as a PCS technology because PCS only refers to digital technologies that were specifically to provide improvements over analog. In the United States, the IEEE and TIA have developed standards for digital cellular systems such as TDMA and CDMA.

2.3 CHANNEL ACCESS

In particular several wireless systems, before several channels can operate independently of each other, the shared should be split up. There are four basic processes are used to do this:

- Time **D**ivision **M**ultiple **A**ccess (TDMA)
- **F**requency **D**ivision **M**ultiple **A**ccess (FDMA)
- **S**pace **D**ivision **M**ultiple **A**ccess (SDMA)
- **C**ode **D**ivision **M**ultiple **A**ccess (CDMA)

Multiple access is a signal transmission situation in which two or more users wish to simultaneously communicate with each other using the same propagation channel. This is precisely the uplink transmission situation in a wireless communication system. In the uplink or reverse channel, multiple users will want to transmit information simultaneously. Without proper coordination among the transmitting users, collisions will occur when two or more users transmit simultaneously. Access methods that incur collision are referred to as random access and variants of random access. We will discuss the throughput characteristics of two popular random access methods: Aloha and carrier-sense multiple access (CSMA). Multiple access strategies based on orthogonally among the competing transmissions are collision-free. Orthogonally can be in the form of frequency division or code division. Techniques with built-in conflict resolution capability presented later in this chapter are frequency-division multiple access (FDMA), time-division multiple access (TDMA) and code-division multiple access (CDMA). Performance analysis and evaluation of these conflict-free multiple access methods in terms of spectral efficiency and system capacity are described and discussed.

2.3.1 Multiple Access in Radio Cell

In each radio cell, the transmission form the base station in the downlink can be heard by each and every mobile user in the cell. For this reason, this mode of transmission is referred to as *broadcasting*. On the other hand, transmission from the mobile users in the uplink to the base station is many-to-one, and is referred to as *multiple access*.

Transmissions in the uplink have the following attributes:

a) Multiple mobile users want to access the common resource (base station) simultaneously.
b) If the transmission from two or more users arrives at the base station receiver at the same time, there will be destructive interference, unless the multiple arriving signals are manually orthogonal.
c) Orthogonally between two signals means that their inner product over the signaling interval vanishes.

The key element in multiple access is to make the transmitted signals from the different users orthogonal to each other. This raises the fundamental question of how this orthogonally condition should be mechanized.
Orthogonally can be mechanized using SDMA, FDMA, TDMA, or CDMA. In theory, the four methods are conflict-free multiple access techniques. The conflict-free property is achieved through coordination among all the participating users. In the case of SDMA, FDMA and TDMA, the coordination among all participating users is performed through fixed assignment. For example, in the FDMA the system bandwidth, is partitioned into frequency bands and each user is assigned a unique frequency band for information transmission for the entire duration of the connection. CDMA is a spread spectrum technique. Each user is assigned a unique spreading function from a

set of wideband orthogonal function. Based on the orthogonolity property, an individual user can transmit using the entire system bandwidth, B_s, during one use of the channel. Thus, FDMA is a narrowband multiple access plans while CDMA is wideband.

2.3.2 Random Access and Variants

In certain situations, depending on the traffic and load and mixture, it may be advantageous to employ a non-conflict-free multiple access scheme. In non-conflict-free multiple access, transmissions by the different users are either uncoordinated or are only partially coordinated. A completely uncoordinated scheme is referred to as *random access*. In a random access, a user contends for usage of the same resource, independent of any other users. For this reason, random access is also referred to as *contention access*. In a random access scheme, a user transmits whenever it has information to be transmitted, independent of the status of any other users.

In conflict-free multiple access systems, random access are often used to gain the initial access to the system. For example, in GSM systems, there is a random access channel (RACH) among the control channels which provide the necessary control functions. The RACH is used by a mobile user to originate a call or to respond to a paging signal in the reverse link. The RACH uses a slotted Aloha access scheme. In responding to a call from a mobile user via the RACH, the base station allocates a conflict-free channel to the user during the call connection.

2.3.3 Random Access

While we will not be dwelling much on non-conflict-free multiple access techniques in this section, it seems appropriate to briefly study some of the popular methods for non-conflict-free multiple access.

2.3.4 Aloha Systems

Random access was used by a research group from the University of Hawaii in the late 1960s and early 1970s for its satellite communications with the U.S. mainland at a transmission speed of 50 kbps. This system was called Aloha and the term Aloha has been used as a general name for random access. Aloha is a packet-switching system. The time interval required to transmit one packet is called a *slot*.

2.4 Time Division Multiple Access (TDMA):

Time Division Multiple Access (TDMA) is based on the fact that the communications elements belonging to different communications participants can be transferred one after the other over the same channel. If this time sequence involves the output and feedback channel between two communications partners, it is also known as *Time Division Duplex* (TDD). In this case the time division multiple access procedure uses a high-bit-rate transfer channel, which is distributed among several channels with a lower data rate during the time sequence. The central feature of time division multiple access is that it is relatively easy implement a central station. It is especially easy to implement a frequency mixer. Supporting handover by changing the base station also appears to be equally uncomplicated because it does not interrupt the transfer of the of the data stream. The data rates of the individual communications participants can also be increased by simply extending the duration of the time slots. In real life this type of time slot bundling is carried out in packet-oriented transfer as part of the *General Packet Radio Services* technique (GPRS) in Global System for Mobile Communication Systems (GSM) and for data transfer in the DPRS technology used in DECT systems. However, the time slots for the various participants are usually managed centrally. One disadvantage is that a TDMA channel needs relatively high bit rates for time division to be practical and effective. Synchronization between the transmitter and the receiver is also somewhat more time-consuming because the transmission between the two stations is constantly interrupted.

2.5 Frequency multiplex procedure

Frequency Division Multiple Access (FDMA) supplies range for a communications stream that is independent of the frequency ranges used by the other communications participants. It is also comparatively easy to allow any station that is ready for communications to select its own frequency alongside centralized frequency management procedure. To do this a station can, for example, check the field strength of all possible transfer frequencies and assume that the transfer frequency with the lowest field strength is free.

Combined procedure

Combinations of both procedures can also be used, increasing the number of possible users. For example, DECT systems use a combination of TDD, TDMA and FDMA procedures with twenty time slots in which every ten times slots are assigned a direction of transfer, and twelve frequencies.

However, the QAM modulation illustrates such a combination of amplitude and pulse modulation.

2.6 Space Division Multiple Access

In *Space Division Multiple Access* (SDMA) the channels used for transmission are used time and time again at specific geometric intervals as part of what is known as "cluster formation". In this way an unlimited amount of traffic can be transferred in an unlimited amount of space, despite a limited number of channels. This procedure is based on the fact that the field strength of a radio signal becomes lower the further it gets from the transmitter because of the attenuation effects. If two transmitters are far enough apart their signals are so weak that they cause no noticeable interference to each other.

2.7 Code Division Multiple Access

Code division multiple access (CDMA) is a form of multiplexing and a method of multiple access that divides up a radio channel not by time (as in time division multiple access), nor by frequency (as in frequency-division multiple access), but instead by using different pseudo-random code sequences for each user. CDMA is a form of "spread-spectrum" signaling, since the modulated coded signal has a much higher bandwidth than the data being communicated.

CDMA also refers to digital cellular telephony systems that make use of this multiple access scheme, such as those pioneered by Qualcomm, and W-CDMA by the International Telecommunication Union or ITU.

CDMA has been used in many communications and navigation systems, including the Global Positioning System and in the Omni-TRACS satellite system for transportation logistics. CDMA consistently provides better capacity for voice and data communications than other commercial mobile technologies, allowing more subscribers to connect at any given time, and it is the common platform on which 3G technologies are built.

This is where CDMA technology fits in. CDMA consistently provides better capacity for voice and data communications than other commercial mobile technologies, allowing more subscribers to connect at any given time, and it is the common platform on which 3G technologies are built.

CDMA is a "spread spectrum" technology, allowing many users to occupy the same time and frequency allocations in a given band/space. As its name implies, CDMA (Code Division Multiple Access) assigns unique codes to each communication to differentiate it from others in the same spectrum. In a world of finite spectrum resources, CDMA enables many more people to share the airwaves at the same time than do alternative technologies.

The CDMA air interface is used in both 2G and 3G networks. 2G CDMA standards are branded CDMA-One and include IS-95A and IS-95B. CDMA is the foundation for 3G services: the two dominant IMT-2000 standards, CDMA2000 and WCDMA, are based on CDMA.

2.7.1 How the Technology Works

The following illustration, which was created with the assistance of Klein Gilhousen, co-inventor of CDMA, shows how bits are encoded at the base station and decoded in the cell phone. A single bit example is used to take you through the Boolean math.

Transmitting from the Base Station

Each voice conversation is compressed with a vocoder. The output is doubled by a convolution encoder that adds redundancy for error checking. Each bit from the encoder is replicated 64 times and exclusive OR'd with a Walsh code that is used to identify that call from the rest.

The output of the Walsh code is exclusive OR'd with the next string of bits (PN sequence) from a pseudo-random number generator, which is used to identify all the calls in a particular cell's sector. At this point, there is 128 times as many bits as there were from the vocoder's output. All the calls are combined and modulated onto a carrier frequency in the 800 MHz range.

Receiving at the Cell phone

The received frequencies are quantized into bits ("chips") by the analog-to-digital converter (ADC). The output is run through the Walsh code and PN sequence correlation receiver to recover the transmitted bits of the original signal. When 20ms of voice data is received, a Viterbi decoder corrects the errors using the convolution code. The Viterbi output goes to the vocoder and digital-to-analog converter (DAC), which decompresses the bits and turns them back into waveforms (sound).

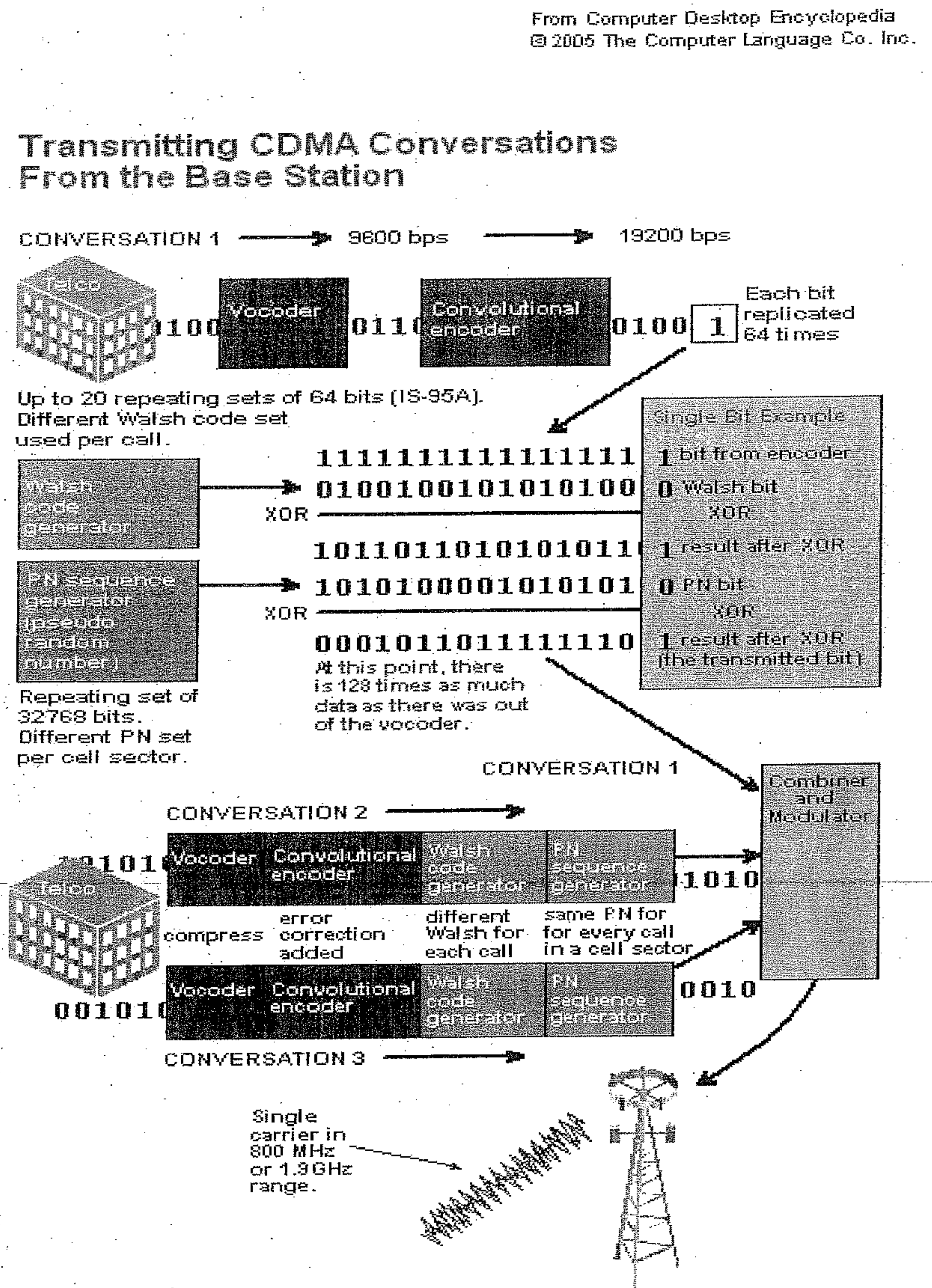


Figure 2-1: Transmitting CDMA from Base Station

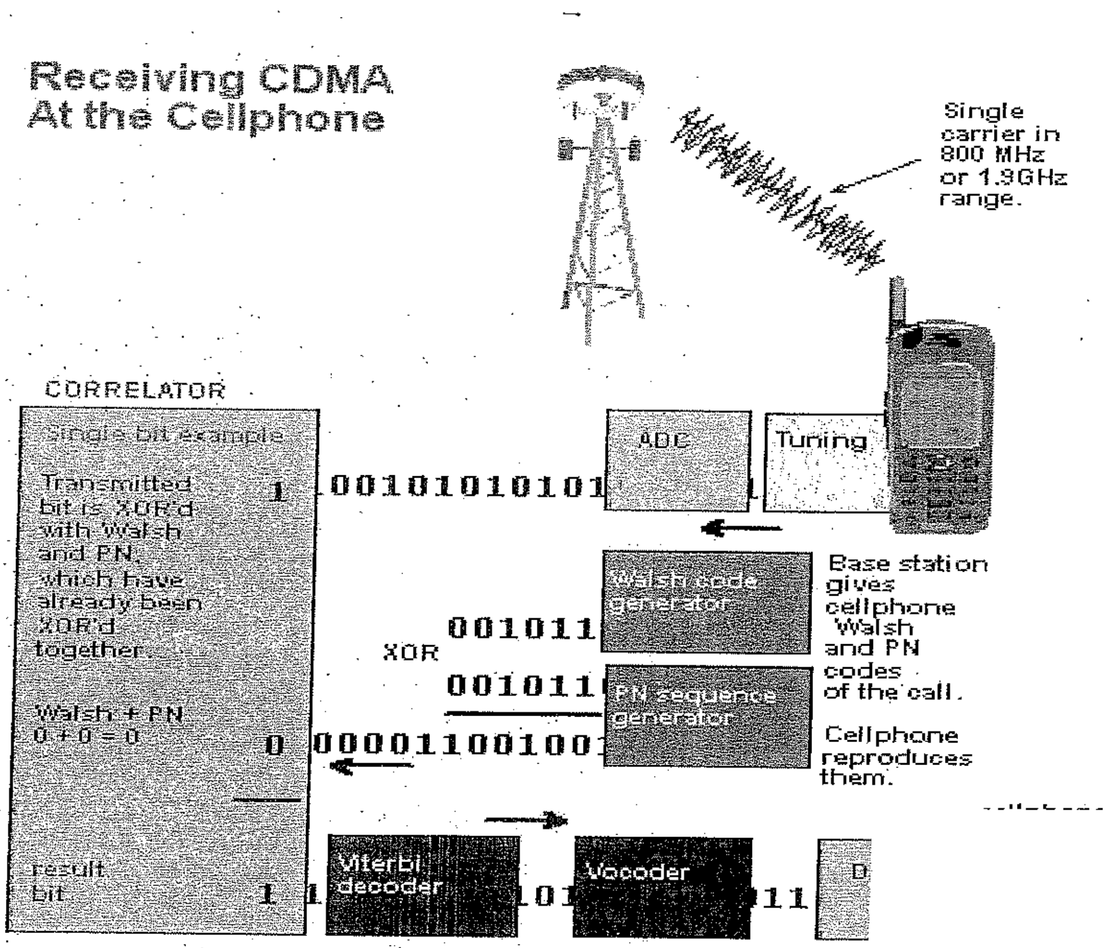


Figure 2-2: Receiving CDMA at the Cell Phone

EXCLUSIVE OR		
IN	IN	OUT
0	0	0
0	1	1
1	0	1
1	1	0

Table 2-2: X-OR truth table

Follow the Single Bit Example:

This exclusive OR truth table shows you the Boolean algebraic rules to prove the single bit example in the illustrations above. The example bit is a 1, and the Walsh and PN codes are 0.

Review Questions:

Q1 : Write the complete name for the following abbreviations :

- AMPS :
- TACS :
- CDMA :
- TDMA :
- FDMA :
- GPRS :
- GSM :
- BS :
- ECDMA :

Q2 : Transmission in the UP-LINK have the following attributes :

-
-
-

Review Questions :

Q1: Draw the basic B.D of digital repeater ?

Q2: What is the function of the repeaters ?

Q3 : Mention different technologies of digital modulation ?

-
-
-
-

Q4 : Write the long name for the following :

- FM :
- AM :
- PM :
- PCM :
- BPSK :

Q5 : compare between TDMA &CDMA technology

Chapter 3

Multiplexing & CDMA-2000

3. Multiplexing

Multiplexing is a process in which two or more signals are combined for transmission over a signal communications path. This concept is conveyed in figure 3-1.

Multiplexing has made communications very economical by transmitting thousands of independent signals over a signal transmission line.

There are three predominant ways to multiplex:

- Frequency Division Multiplexing. (FDM)
- Time Division Multiplexing. (TDM)
- Wavelength Division Multiplexing. (WDM)

WDM is used exclusively in optical communications.

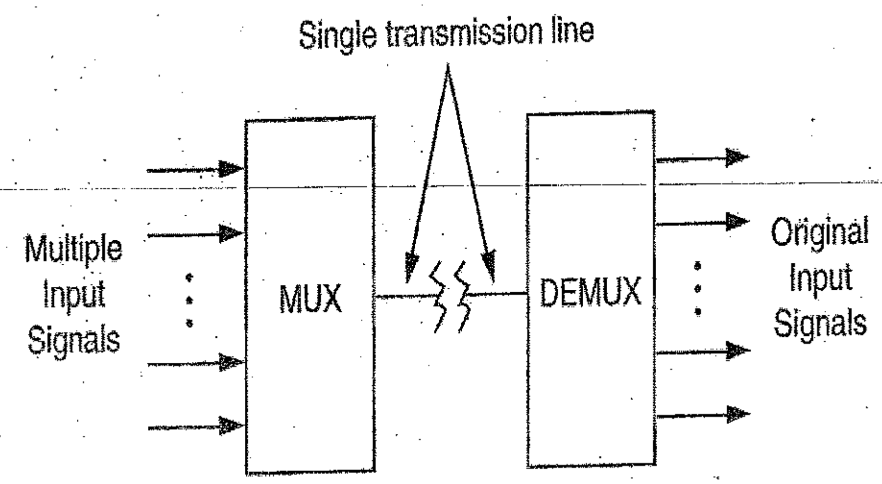


Figure 3-1: Concept of multiplexing

3.1 Frequency Division Multiplexing (FDM)

Frequency Division Multiplexing (FDM) is predominantly used in analog communications. Figure 3-2 shows a general block diagram of an FDM system where each signal is assigned a different carrier frequency. The modulated carrier frequencies are combined for transmission over a single line by a multiplexer (MUX). There is always some unused frequency range between channels, known as guard band. At the receiving end of the communication link, a de-multiplexer (DEMUX) separates the channels by their frequency and routes them to the proper end users. A two-way communication circuit requires a multiplexer/de-multiplexer at each of the long-distance, high-bandwidth cable. FDM was the first multiplexing scheme to enjoy wide-scale network deployment.

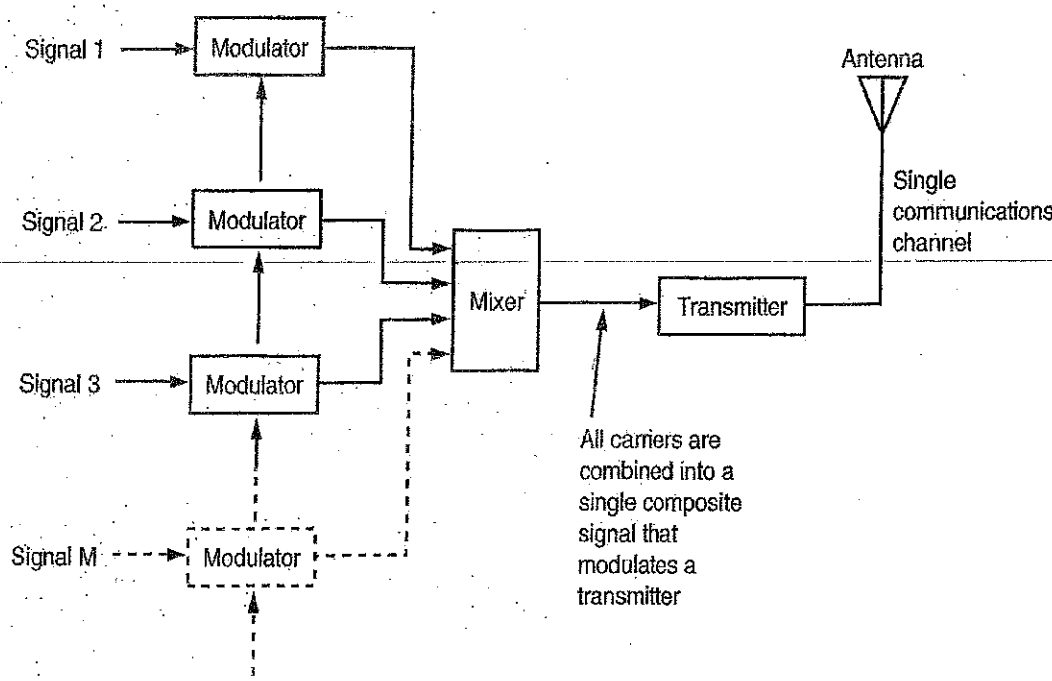


Figure 3-2: Transmitting end of FDM system

FDM is widely used in FM stereo broadcast. It prevents compatibility with mono-receivers and requires only a slight increase in BW. in stereo, two microphones are used to generate two separate audio signals, generally called the left *L* and right *R*. the two microphones provides sufficient difference in the audio signals to provide more realistic reproduction of the original sound. The L and R are fed to a circuit where they are combined to form sum L+R and difference L-R signals. FDM techniques are used to transmit these two independent signals on a signal channel.

To explore this concept further, consider how different voice channels can be placed on a signal wire or cable using FDM. Each requires a maximum 4 kHz apart. The 12 carrier frequencies are 60 kHz, 64 kHz, and so on, through 108 kHz, causing the 12 voice channels to occupy non-overlapping frequencies. The resulting separate bandwidths are summed so the channels can be stacked on top of each other in the frequency spectrum. As shown in Figure 3-3 twelve voice channels are combined into a *group*. Five groups form a *super-group*, and ten supergroups from a *mastergruop*. This mastergruop can handle total of 12 × 5 × 10 = 600 channels. Figure 3-4 provides the bell system's hierarchy of FDM group.

FDM's disadvantages stem from analog circuitry, crosstalk and the difficulty on interfacing an FM transmitter with digital source such as a computer; also, an FM channel remains idle when not in use.

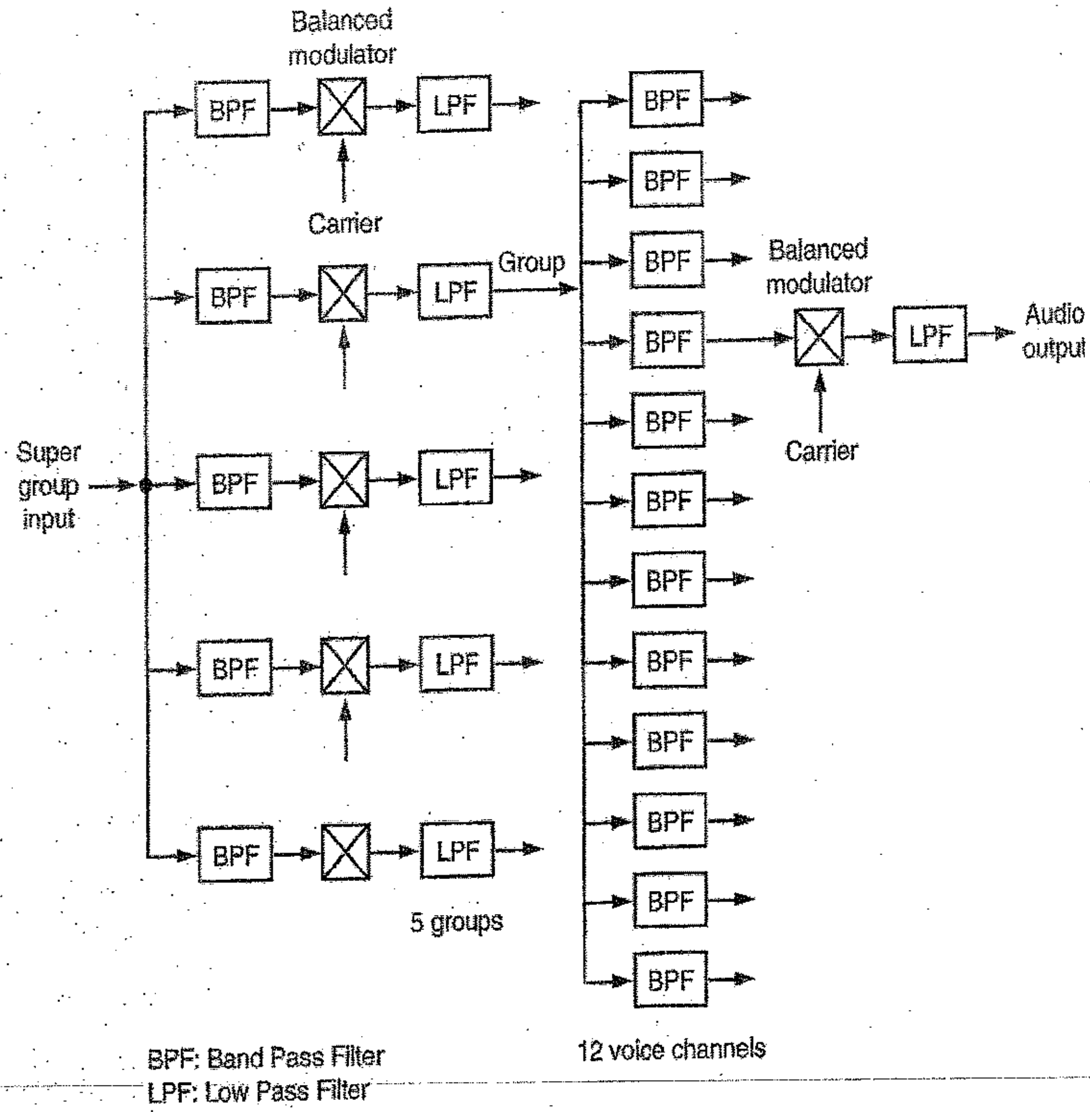


Figure 3-3: Demultiplexing of the telephone signals in an FDM system

3.2 Time Division Multiplexing (TDM)

While FDM has been used to great advantage in increasing system capacity, the use of TDM offers even greater system improvements. TDM is protocol insensitive and is capable of combing various protocols and different types of signals, such as voice and data, onto a single high-speed transmission link. It is more efficient than FDM, as there is no need for guard bands. In order to use TDM, the transmission must be digital in nature so an essential component of TDM is the process of sampling the analog signal in time.

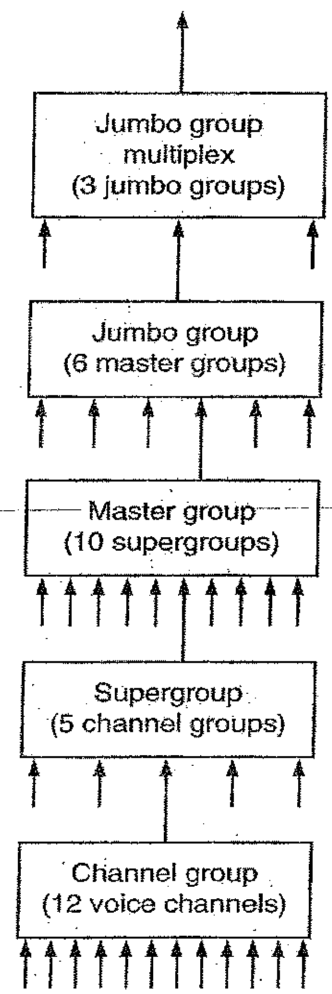


Figure 3-4: Hierarchy of the Bell System's FDM group

In order to transmit telephone conversations, speech, this is an analog signal, is converted to a digital signal, transmitted, and the reconverted into analog at the receiving telephone. The main disadvantages of TDM are the greater complexity of digital systems and greater transmission bandwidth, required. Large – scale, low-cost ICs are reducing the difficulty and expense of constructing complex circuitry, and data. Compression techniques are beginning to decrease the bandwidth penalty. In general, the advantages outweigh the disadvantages.

A T -1 carrier uses TDM where each of the 24 channels is assigned an 8-bit time slot, as depicted in Figure 3-5. A framing bit is used to synchronize the system. For 24 channels, there area total of 193 bits (24x2+1 framing bit) occurring 8,000 times a second, as shown in figure 3-6. This gives a bit rate of 1.544 Mbps (193x 18000). Digital channels offer much move versatility and much higher speed than analog channels. Furthermore, the digital signal is much move immune to channel noise than is the analog signal.

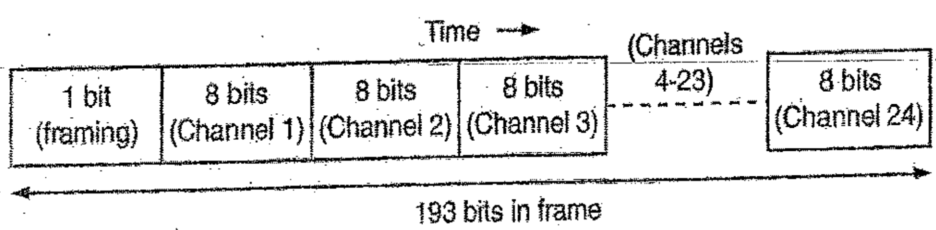


Figure 3-5: T-1 frame

However, at 1.544 Mbps, T-1 lines simply do not have sufficient bandwidth to deal with the new demands being made on networks. Yet fiber – based T-3s at 45 Mbps bandwidth and 10 times the cost are overkill for many small and mid-sized businesses. Moreover, T3 circuits are not easily available to many businesses, while T-1 lines are ubiquitous. The price, bandwidth, and availability gap between T-1 and T-3 has businesses and service providers searching for cost-effective ways to fulfill needs. *Inverse multiplexing* of T -1s benefits carriers and end users alike in bridging this bandwidth gap between 1.5 Mbps and 45 Mbps, while is a critical range for many wide area network applications.

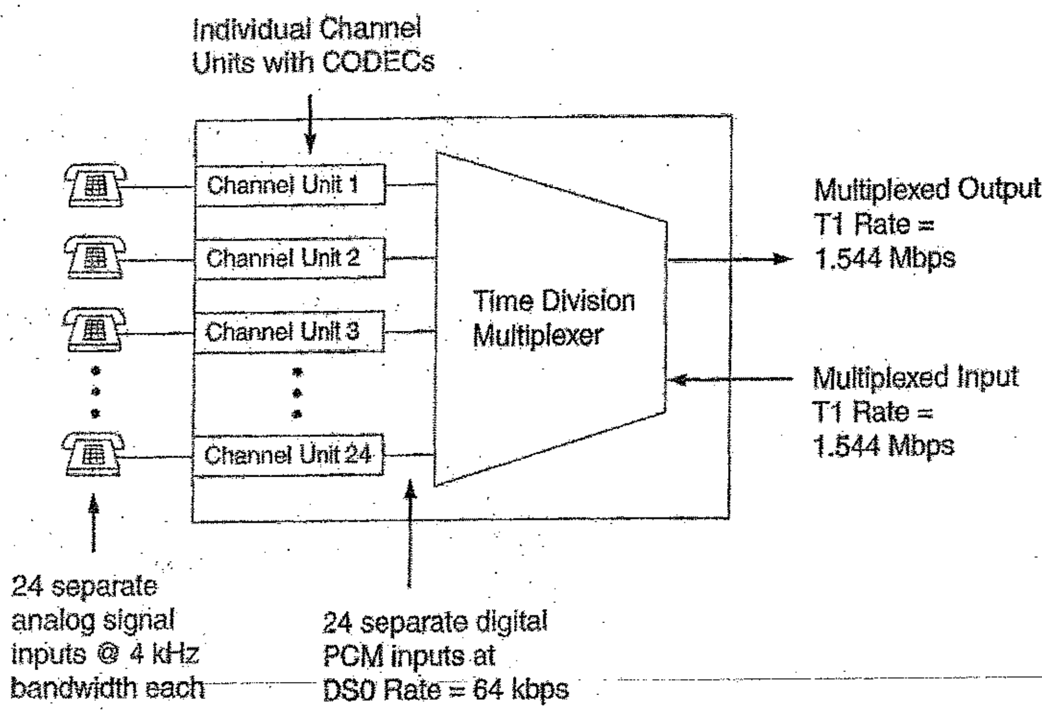


Figure 3-6: Time Division Multiplexing (TDM) in a T-1 frame

Inverse multiplexing of T-1s is the process of distributing a serial data Stream, bit by bit onto multiple T –1s, then reassembling the original data stream at the receiving end. The chief benefit of T-1 inverse multiplexing is that it uses the ubiquitous T-1 infrastructure to create clear data channels from 3 to 12 Mbps. The primary work of the inverse multiplexer is to assure that the bits are reassembled in the correct order. Avery small portion of the T-1 payload is taken over for *metaframing*, which keeps the T-1s aligned in spite of minor timing differences and unequal circuit delays. Since there is no industry standard as of yet for bit-based T-1 inverse multiplexing, inverse multiplexers use a proprietary metaframing technique, which means that the devices at both ends of a data channel must be from the same vendor. Specially, channels 1 through 8 of the T-3 are assigned to voice, channels 25 through 28 are assigned for Internet access, and channels 9 through 24 are available as spare capacity for voice and/or data.

There are basically three different TDM schemes: conventional TDM, Statistical TDM (STDM), and cell-relay or ATM. STDM includes Conventional STDM, Frame Relay, and X. 25 networking.

3.2.1 Conventional TDM

Conventional TDM systems usually employ either bit-interleaved or byte-interleaved multiplexing schemes. Clocking (bit timing) is critical in conventional TDM. All sources of I/O and clock frequencies must be derived from a central, traceable source for the greatest efficiency. In bit-interleaved TDM, a single data bit from an I/O port is output to the aggregate or the single communication channel, followed by a data bit from another I/O port, and so on, with the process repeating itself. A time-slice is reserved on the aggregate channel for each individual I/O port. Since the time-slices for each I/O port are known to both the transmitter and the receiver, the only requirement is for the transmitter and reliever to be instep. This is accomplished through the use of a synchronization Channel between the two multiplexers. The synchronization channel transports a fixed pattern that the receiver uses to acquire synchronization. Total I/O bandwidth cannot exceed that the aggregate minus the bandwidth requirements for the synchronization channel.

Bit-interleaved TDM is simple and efficient and requires little or no buffering of I/O data, but it does not fit in well with microprocessor-driven, byte-based environment. In byte-interleaved multiplexing, complete words (bytes) from I/O channels are placed sequentially, one after another, onto the high-speed aggregate channel. Otherwise, the process is identical to bit-interleaved multiplexing. Byte-interleaved systems were heavily deployed from the late 1970s to around 1985. In 1984, with the divestiture of AT&T and the launch of T-1 facilities and services, many Companies jumped into the private networking market, pioneering a generation of intelligent TDM called STDM networks.

With conventional TDM, the timeslots are allocated on a constant basis. Thus, if a channel does not need to transit data, the channel bandwidth goes unused during that time slot. This inefficiency is overcome by STDM technique. The term *statistical* refers to the fact that the time slots are allocated on a need-basis.

3.2.2 Statistical Time Division Multiplexing (STDM)

Statistical Time Division Multiplexing (STDM) allocates slices on demand, but it needs to know the address of the station, which is an additional overhead. A block diagram of a STDM application is shown on Figure 3-7. Its advantage is that there is no idle time, but a buffer is needed to handle simultaneous requests. In this scheme, the underlying assumption is that not all channels are transmitting all the time. A statistical multiplexer (stat mux) has an aggregate transmission BW that is less than the sum of channel BWs because the aggregate bandwidth is used only when there is actual data to be transported from I/O ports. The receiver knows the destination port for the data it receives because the transmitter sends not only the data but also an address. The address identifies the port the data is destined for. The *stat mux* assigns variable time slots every second depending upon the number of users and the amount of data transmitted by each. Frame Relay, X.25, and Switched MultiMegabit Data Service (SMDS) are all categorized as STDM systems.

STDM's biggest disadvantage is that it is I/O protocol sensitive. Therefore, a *stat mux* has difficulty supporting transparent I/O data and unusual protocols. To support these I/O data types, many STDM systems have provisions to support conventional TDM I/O traffic through the use of adjunct/integrated modules. This conventional STDM was very popular in the late 1970s to mid 1080s and still used, although the market for these units is dwindling. In conventional STDM, as I/O traffic arrives at the *stat mux* it is buffered and then inserted into frames. The relieving units remove the I/O traffic from the aggregate frames. Statistical multiplexers are ideally suited for the transport of asynchronous I/O data as they can take advantage of the inherent latency in asynchronous communications. However, they can also multiplex synchronous protocols by *spoofing*, again taking advantage of the latency between blocks or frames. Spoofing refers to simulating a communications protocol by a program that is interjected into a normal sequence of processes for the purpose of adding some useful function.

Time Assignment speech Interpolation (TASI) represents an analog STDM scheme. These systems were in limited use in the 1980s and were particularly adept at sharing voice circuits, specifically Private Branch Exchange (PBX) trunks. In normal telephone conversations is spent in a latent (idle) state. TASI trunks allocate snippets of more Common voice from another channel during this idle time. As digital speech processing became move common, TASI systems called Digital speech Interpolation (DSI) were created. These had analog inputs and digital outputs.

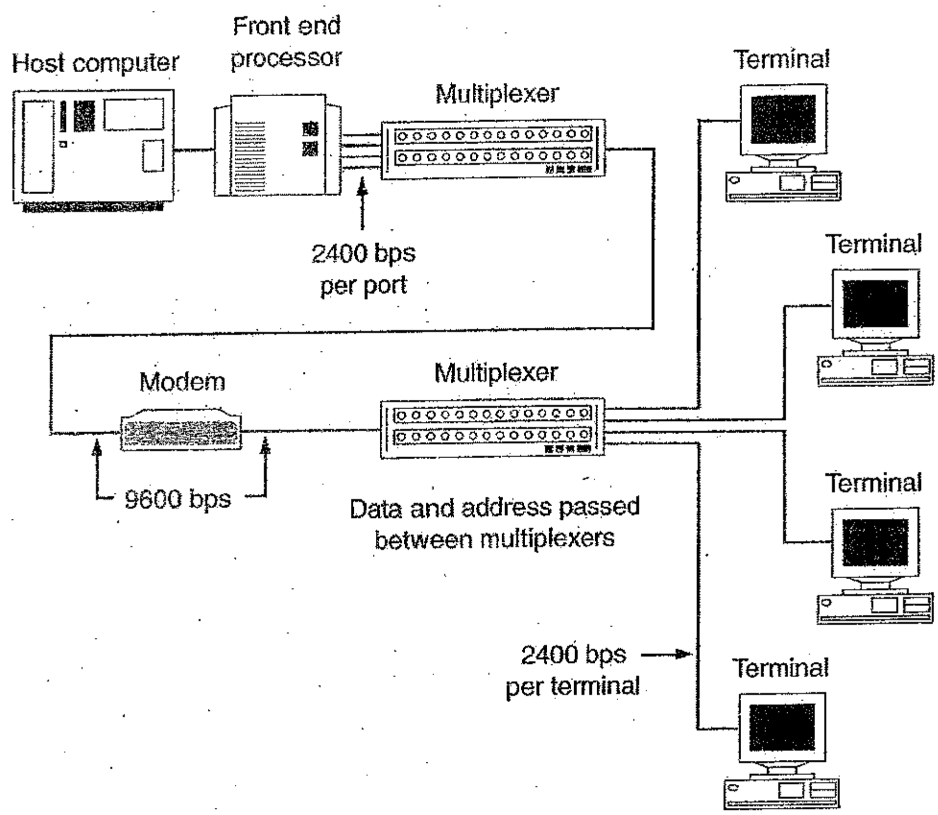


Figure 3-7: Block diagram of a Statistical Time Division Multiplexing application

Both TASI and DSI systems suffer from some major drawbacks. First, users can notice a lot of voice clipping when a little bit of speech is lost while waiting hr the TASI mux to detect valid speech and allocate bandwidth. Clipping also occurs when there is insufficient bandwidth. In addition, TASI and DSI units are very susceptible to audio input levels and may have problems with the transport of voice-band data, for example, modem signals.

3.3 Wavelength Division Multiplexing (WDM)

Wavelength Division Multiplexing (WDM) is a cost-effective way to increase the capacity of fiber optic communications. The key elements of a WDM optical system are tunable semiconductor lasers, electro-optical modulators, multiplexing components, single-mode optical fibers, and optical amplifiers. This system, depicted in Figure 3-8 makes use of the optical fiber's available intrinsic bandwidth by multiplexing many wavelengths (or colors) of coherent light along a single-mode optical fiber channel. Each wavelength of light can transmit encoded information at the optimum data rate. Therefore, multiplexing the distinct wavelengths of light lead to a significant increase in the total throughput.

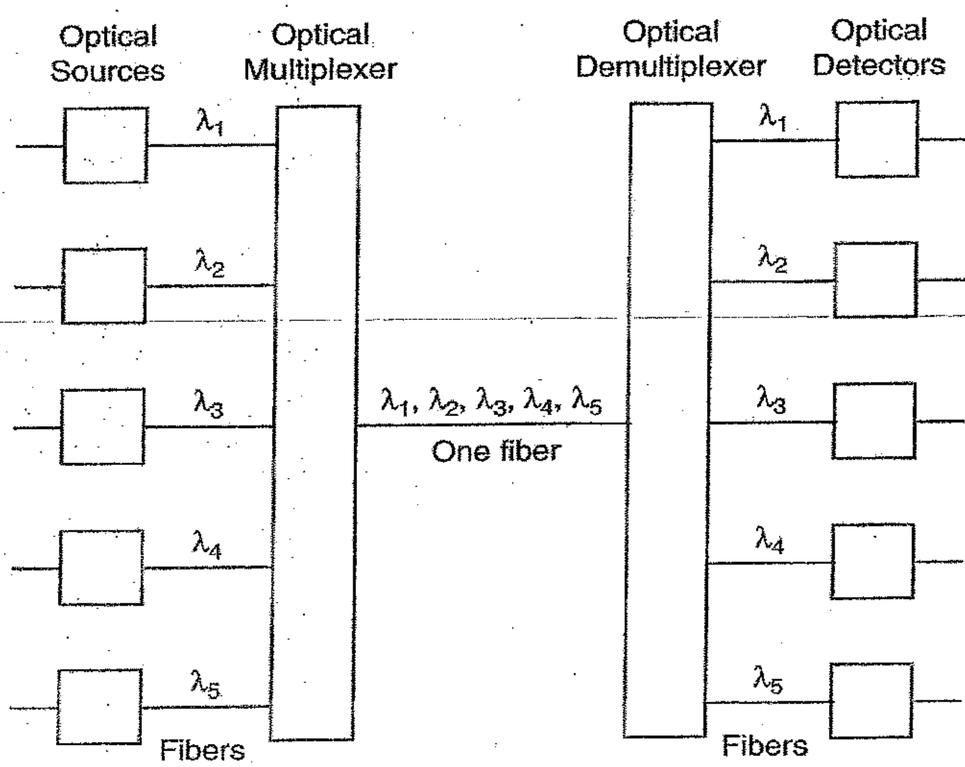


Figure 3-8: Wavelength Division Multiplexing (WDM)

For example, a single-mode optical fiber with an attenuation of 0.2 dB per km at 1,550 nm is capable of accommodating a set of wavelengths each spaced apart by a few tenths of nm (50 GHz to 100 GHz). Thus, it has an estimated transmission capacity in the THz regime. This indicates that instead of using a single wavelength laser to transmit far more information a long the same channel, thereby increasing the total capacity of optical transmission. The use of 48 distinct wavelength lasers, each modulated at 2.5 Gbps, represents an effective transmission rate of 48 times 2.5 Gbps, which is equal to 125 Gbps. the use of 100 distinct wavelength lasers could increase the effective data capable of achieving these speeds; this tend will move wide-area networking speeds from Mbps to Gbps and eventually to Tbps. the idea appears to be moving toward reality as many companies are providing advanced WDM technologies that allow the service or trunk providers to upgrade their system capacity in accordance with the ever-increasing demand for information.

3.3.1 High-Density or Dense WDM (DWDM)

High-density or *Dense* WDM (DWDM) technology is typically found at the core of carrier networks. Optical fiber technology has undergone many improvements since the first lines were laid in ground nearly 20 years ago. Rather than digging up and replacing these lines whenever new technology outdates them, telecommunications companies have searched for ways to maximize bandwidth and minimize dispersion in order fibers that lack the advantages of more recent design. The challenge of delivering greater bandwidth surged research efforts in managing wavelengths.

DWDM on the WAN was created significant new high-speed opportunities by assigning individual optical signals to specific wavelength and multiplexing the signal as separate channels across a single optical fiber. Until

1998, the predominant driver in DWDM deployment was long-distance transport applications with the network architecture being point-to-point DWDM. The goal was to send as many channels across a single fiber as far as possible. But DWDM is now migrating into MAN and LAN applications where the economics of installing DWDM systems are more attractive that upgrading the entire installed fiber plant.

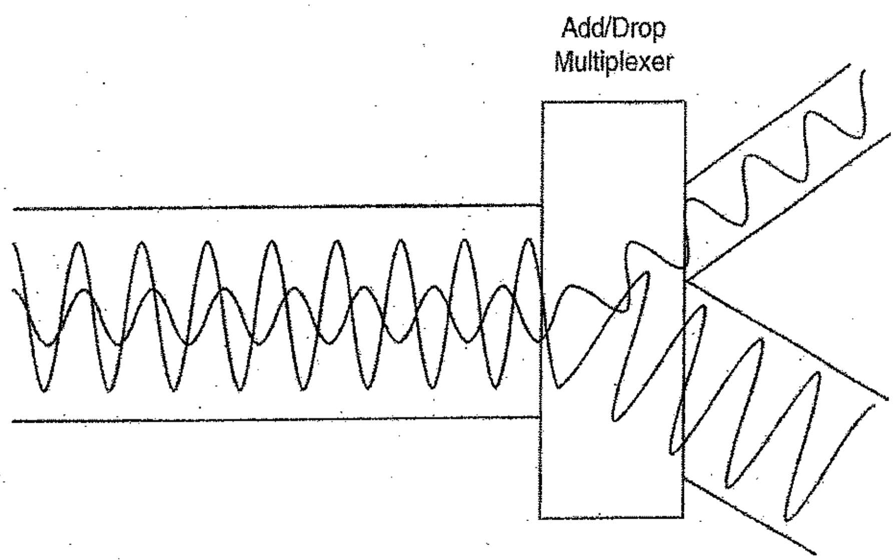


Figure 3-9: Optical add/drop multiplexer

A DWDM system employs at least four multiplexing devices: one mux and one demux for each direction channels. Several devices may be cascaded to multiplex the desired number of channels. These systems require that the laser wavelength be extremely stable. The drawback to DWDM is that it only functions point to point. So if a point fails, all calls on the path are lost until an alternative path can be set up an individual sessions are reestablished. However, add/drop multiplexers circumvent the need to demultiplex transmission into electronic signals prior to rerouting or amplifying them, as shown in Figure 3-9. Furthermore, optical routers switch DWDM traffic wavelength by wavelength as it comes into optical hub sites, allowing carriers to establish meshed optical nets. The result is a more reliable network with virtually no downtime because the routers share network data and are smart enough to quickly route around failures. The ability to optically multiplex each light path, or wavelength, has also given rise to DWDM systems that can transparently bring in heterogeneous data formats. DWDM has been compared to a multilane highway for carrying data, in contrast to a single line in the case of a traditional TDM implementation. Rather than trying to pack more vehicles into one lane at increasingly higher speeds, DWDM makes full use of all the lanes. Perhaps more importantly, it also enables the various highway lanes to move at different speeds and to carry different types of information. But abundant bandwidth DWDM affords is a growing challenge.

3.4 Code Division Multiple Access (CDMA)

Introduction

A CDMA system uses a spread-spectrum technology platform, which enables multiple users to occupy the same radio channel, frequency spectrum, at the same time. CDMA technology has been and is being utilized for microwave point -to-point communication and satellite communication in both military and civilian applications. With a CDMA system, each of the subscribers, or users, utilizes his or her own unique code to differentiate himself or herself from the other users.

A CDMA system offers many special features including its ability to thwart interference and to improve immunity to multipath effects due to its bandwidth.

The benefits associated with CDMA technology are the following:

- More system capacity than analog and TDMA systems.

- Improve protection from interference.
- No frequency Planning required between CDMA channels.
- Improve handoffs with MA HO and soft handoffs.
- Fraud protection due to encryption and authentication capabilities.
- Compatibility with new wireless features.

CDMA technology is based on the principle of *direct sequence* (DS) and is a wideband spread-spectrum technology. The (DMA channel utilized is reused in every cell of the system and is differentiated by the pseudo *random number* (PN) *code* that it utilizes.

However, despite the apparent advantages CDMA technology offered for cellular systems, there are several implementation concerns regarding its use in an existing system. The introduction of a CDMA system into existing AMPS requires the establishment of a guard band and guard zone. The guard band and the guard zone ane required to ensure that the interference received from the AMPS does not negatively impact the ability for the CDMA system to perform well.

Code Division Multiple Access (CDMA), in layman's terms, is like a full of people talking in different languages at the same time. However, you can pick out the conversation in the language you understand and ignore all the others. CDMA is a method of sharing a single frequency among users by encrypting each signal with a different code. As a result, it supports many callers along the same carrier. Transmission signals are broken up into coded *packets* of information that hop available frequencies and are reassembled at the receiving end. Each earth station transmits coded information to the satellite, regardless of any overlap with other stations that may be transmitting simultaneously. At the receiver end, the separation of the transmitted information by each station is achieved through the detection of the individual earth station's transmitted identification code.

Like TDMA, CDMA operates in the 1900 MHz band as well as the 800 MHz band. CDMA has three times the capacity of TDMA and ten times that of FDMA. In 1992, the TIA published the first CDMA standard, IS-95. The *third generation* CDMA systems provide both operators and subscribers with significant advantages over first (analog) and second (TDMA-based) generation systems. The advanced methods utilizes in CDMA technology improve capacity, coverage, voice quality, and immunity from interference by other signals introducing a new generation of wireless networks.

Compared with the conventional systems, the CDMA system makes frequency assignment easy, as the same frequency can be used for each cell. Moreover, this system allows high-quality communications, since no frequency switching is required when moving from one cell to another. CDMA is the first technology to use a technique called **soft handoff**, which allows a handset to communicate with multiple base stations simultaneously; the system chooses the best signal in order to provide the user with the best audio at all times. In contrast to CDMA users, TDMA users can experience an interruption in the audio when the signal is handed off from one base station to another, resulting in higher interference during handoff and increased dropped calls.

CDMA employs a technique originally created by the military called a **spread spectrum**, which is significantly different from AMPS and TDMA technologies. Rather than dividing RF spectrum into separates user channels by frequency slices as in FDMA or time slots as in TDMA, spread spectrum users by assigning them digital codes within a broad range of the radio frequency. The system chops up the signal into data packets and spreads it over a wide band of frequencies. The data packets are then reassembled at the relieving end. This makes it very difficult for hackers to grab orderly, meaningful data, and ensures secure communications. In addition, the signal appears like low-level noise to conventional radio receivers. Since the receivers are designed to eliminate noise, the signed goes undetected. Also, this technology is ideal for densely populated areas because it prevents signal interference among mobile units, say cellular phones and portable computers. The spread spectrum techniques can be divided into two families: Frequency Hopping spread spectrum (FHSS) and Direct Sequence Spread Spectrum (DSSS).

3.4.1 Frequency Hopping spread spectrum (FHSS)

Frequency Hopping spread spectrum (FHSS) resists interference by jumping rapidly from frequency to frequency in a pseudo-random way. The relieving system has the same pseudo-random algorithm as the sender and jumps simultaneously. These systems are considered less susceptible because of the constant shifting of the frequency, which also makes FHSS Systems difficult to intercept and gives them an added layer of security. Another advantage of the reduced interference inherent in FHSS systems is that multiple hopping sequences can typically be assigned within the same physical area, thereby increasing the total amount of available bandwidth. In large facilities, especially those with multiple floors, it is necessary to space antennas, or access points, in an overlapping array. This ensures adequate coverage for wireless devices that are moving from cell to cell. With FHSS, system users can roam between access points on different channels. This makes FHSS technology more flexible than DSSS, in this respect.

3.4.2 Direct Sequence Spread Spectrum (DSSS)

Direct Sequence Spread Spectrum (DSSS) resists interference by mixing in a series of pseudo-random bits with the actual data. The receiver, using the same pseudo-random algorithms, strips out the extra bits. A redundant bit pattern (also called a chip) is produced for each bit of data can be transmitted. If one or more bits are damaged in transmission, the original data can be recovered as opposed to having to be retransmitted process. DSSS can be used as a substitute for leased lines or fiber optic cables to bridge LAN segments in point-to-point or multi-point connectivity between buildings. In DSSS, roaming can only be done between access points on the same channel, which makes the roaming capabilities of DSSS less robust than those of FHSS.

3.4.3 Cellular Digital Packet Data (CDPD)

Cellular Digital Packet Data (CDPD) has been combined with spread-spectrum radio transmission in several wireless communications products. CDPD allows a packet of information to be transmitted in between voice telephone calls. Even if all the subscribers in a particular area are using their cellular telephones, 30 percent of the frequency is still available for data transfer. For signal transmission, the system starts channel hopping. Anytime it finds an opening, it transmits the information in packets. This approach not only spreads the packets in a predetermined manner, but also causes them to hop frequencies.

CDPD enables data specific technology to be tacked onto the existing cellular telephone infrastructure. It supports wireless access to the internet and other public packet-switched networks. CDPD is an open specification that adheres to the layered structure of the OSI model and therefore, has the ability to be extended in the future. CDPD supports the internet's IP protocols for broadcasting and multicasting, including IPv6, and the ISO Connection Network Protocol (CLNP).

For mobile users, CDPD support for packet-switching means that a persistence link is no longer a requirement. The same broadcast channel can be shared among a number of users at the same time. The user's modem recognizes the packet intended for its user. As data such as e-mail arrives, it is forwarded immediately to the user without a circuit connection having to be established. There is a circuit-switched version, called CS CDPD, which can be used where traffic is expected to be heavy enough to warrant a dedicated connection. Cellular telephone and modem providers that offer CDPD support make it possible for mobile users to get access to the internet.

3.5 CDMA-2000

CDMA2000 is a family of third-generation (3G) mobile telecommunications standards that use CDMA, a multiple access scheme for digital radio, to send voice, data, and signaling data (such as a dialed telephone number) between mobile phones and cell sites. It is the second generation of CDMA digital cellular.

CDMA (code division multiple access) is a mobile digital radio technology that transmits streams of bits and whose channels are divided using codes (PN sequences). CDMA permits many radios to share the same frequency channel. Unlike TDMA (time division multiple access), a different technique used in GSM and D-AMPS, all radios can be active all the time, because network capacity does not directly limit the number of active radios. Since larger numbers of phones can be served by smaller numbers of cell sites, CDMA-based standards have a significant economic advantage over TDMA-based standards, or the oldest cellular standards that used frequency division multiple access (FDMA).

CDMA2000 has a relatively long technical history, and remains compatible with the older CDMA telephony methods (such as CDMA-One) first developed by Qualcomm, a commercial company, and holder of several key international patents on the technology.

CDMA2000 can support mobile data communications at speeds ranging from 144 Kbps to 2 Mbps. Versions have been developed by Ericsson and Qualcomm. As of March 2006, the CDMA Development Group reports more than 250,300,000 subscribers worldwide.

The figure 3-10 shows an overview of a CDMA2000 radio system. This diagram shows that there are several types of devices that can be used in the CDMA2000 system. These include CDMA2000 multiple bandwidth radios and IS-95 CDMA compatible mobile telephones. The CDMA2000 mobile telephone devices are usually capable of operating as IS-95 CDMA and CDMA2000 mobile radios. This diagram also shows that the CDMA2000 system can mix and combine standard 1.25 MHz wide IS-95 channels into the 3.75 MHz CDMA2000 channels.

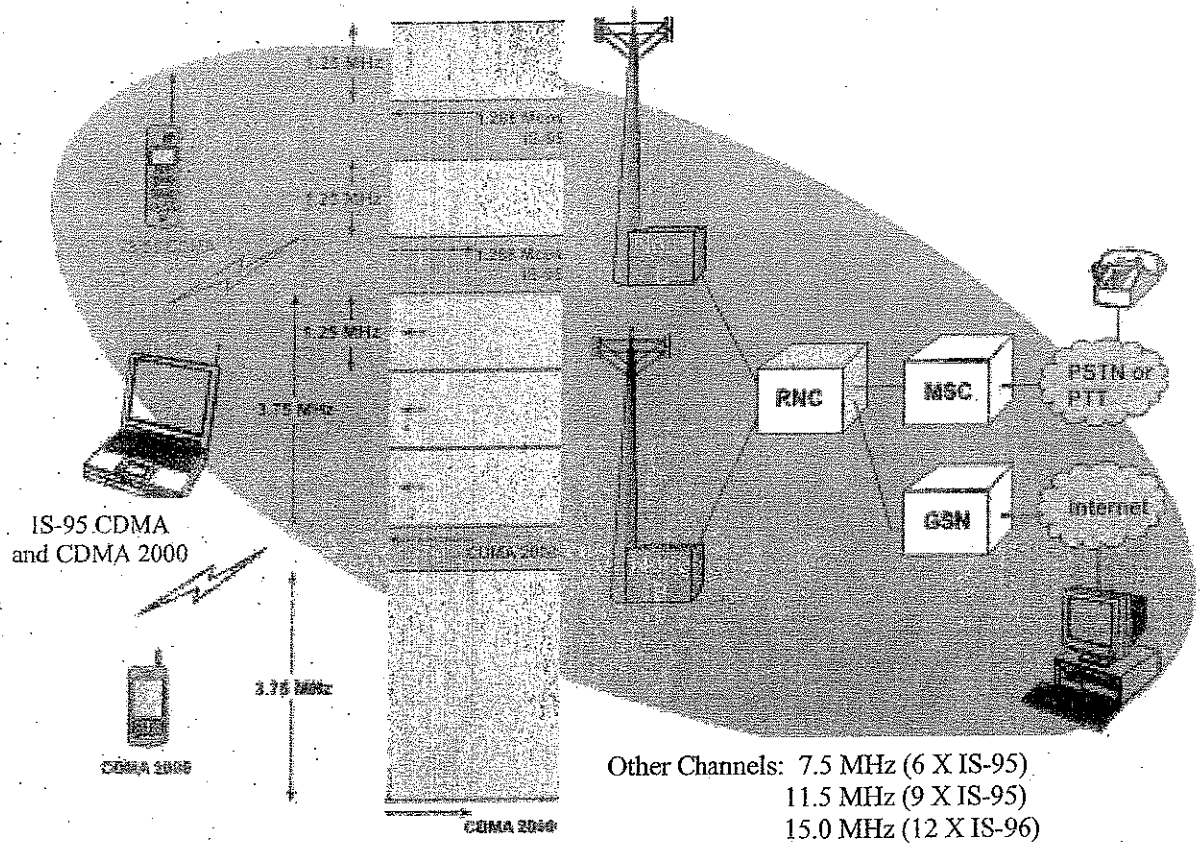


Figure 3-10: Overview of a CDMA2000 radio system

The CDMA2000 standards **CDMA2000 1x**, **CDMA2000 1xEV-DO**, and **CDMA2000 1xEV-DV** are approved radio interfaces for the ITU's IMT-2000 standard and a direct successor to 2G CDMA, IS-95 (CDMA-One). CDMA2000 is standardized by 3GPP2.

CDMA2000 is a registered trademark of the Telecommunications Industry Association (TIA-USA) in the United States, not a generic term like CDMA. (This is similar to how TIA has branded their 2G CDMA standard, IS-95, as CDMA-One.)

CDMA2000 is an incompatible competitor of the other major 3G standard UMTS. It is defined to operate at 400 MHz, 800 MHz, 900 MHz, 1700 MHz, 1800 MHz, 1900 MHz, and 2100 MHz.

Below are the different types of CDMA2000, in order of increasing complexity:

3.5.1 CDMA2000 1x

CDMA2000 1x, the core CDMA2000 wireless air interface standard, is known by many terms: **1x**, **1xRTT**, **IS-2000**, **CDMA2000 1X**, **1X**, and **cdma2000** (lowercase). The designation "1xRTT" (1 times Radio Transmission Technology) is used to identify the version of CDMA2000 radio technology that operates in a pair of 1.25-MHz radio channels (one times 1.25 MHz, as opposed to three times 1.25 MHz in 3xRTT). 1xRTT almost doubles voice capacity over IS-95 networks. Although capable of higher data rates, most deployments have limited the peak data rate to 144 Kbps. While 1xRTT officially qualifies as 3G technology, 1xRTT is considered by some to be a 2.5G (or sometimes 2.75G) technology. This has allowed it to be deployed in 2G spectrum in some countries which limit 3G systems to certain bands.

The main differences between IS-95 and IS-2000 signaling are: 64 more traffic channels on the forward link that are orthogonal to the original set. Some changes were also made to the data link layer to accommodate the greater use of data services—IS-2000 has media and link access control protocols and Quality of Service (QoS) control. In IS-95, none of these were present, and the data link layer basically consisted of a "best effort delivery" RLP—this arrangement is still used for voice.

In the United States, Verizon Wireless, Sprint PCS, MetroPCS, Alltel, Cellular South, and U.S. Cellular use 1x, also it is in use in Canada by Bell Mobility and TELUS Mobility, and in Mexico by Iusacell and Unefon.

In India, BSNL, Reliance and Tata Teleservices are major wireless services providers on CDMA 2000 1x.

In Indonesia, Mobile-8 is the major mobile wireless service provider on CDMA 2000 1x (with EVDO in Western Java). The other CDMA providers are fixed wireless (such as Bakrie Telkom, Telkom Flexi, and Indosat's Starone).

In People's Republic of China, China Unicom is the major mobile wireless service provider on CDMA 2000 1x.

CDMA2000 3x

CDMA2000 3x utilizes a pair of 3.75-MHz radio channels (i.e., 3 X 1.25 MHz) to achieve higher data rates. The 3x version of CDMA2000 is sometimes referred to as Multi-Carrier or MC. The 3x version of CDMA2000 has not been deployed and is not under development at present.

3.5.2 CDMA2000 1xEV-DO

CDMA2000 1xEV-DO (1x Evolution-Data Optimized, originally **1x Evolution-Data Only**), also referred to as **1xEV-DO, EV-DO, EVDO**, or just **DO**, is an evolution of CDMA2000 **1x** with High Data Rate (HDR) capability added and where the forward link is time-division multiplexed. This 3G air interface standard is denoted as **IS-856**.

CDMA2000 1xEV-DO in its latest revision, Rev. A, supports downlink (forward link) data rates up to 3.1 Mbps and uplink (reverse link) data rates up to 1.8 Mbps in a radio channel dedicated to carrying high-speed packet data. 1xEV-DO Rev. A was first deployed in Japan and will be deployed in North America in 2006. The Rev. 0 that is currently deployed in North America has a peak downlink data rate of 2.5 Mbps and a peak uplink data rate of 154 kbps.

3.5.3 CDMA2000 1xEV-DV

CDMA2000 1xEV-DV (1x Evolution-Data/Voice), supports downlink (forward link) data rates up to 3.1 Mbps and uplink (reverse link) data rates of up to 1.8 Mbps. 1xEV-DV can also support concurrent operation of legacy 1x voice users, 1x data users, and high speed 1xEV-DV data users within the same radio channel.

In 2005, Qualcomm put the development of EV-DV on an indefinite halt, due to lack of carrier interest, mostly because both Verizon Wireless and Sprint are using EV-DO.

3.6 Advantages of CDMA-2000

CDMA2000 benefited from the extensive experience acquired through several years of operation of CDMA-One systems. As a result, CDMA2000 is a very efficient and robust technology. It delivers the highest voice capacity and data throughput using the least amount of spectrum, and it can be used to provide services in urban as well as remote areas cost effectively.

The unique features, benefits, and performance of CDMA-2000 make it an excellent technology for high-voice capacity and high-speed packet data. Since CDMA2000 1X supports both voice and data services on the same carrier, it allows operators to provide both services cost efficiently. CDMA2000 1xEV-DO is optimized for data and is capable to support large volumes of data traffic at broadband speeds. 1xEV-DO is well suited to provide high-speed data services to its mobile subscribers and/or broadband access to the Internet.

Due to its optimized radio technology, CDMA-2000 enables operators to invest in fewer cell sites and deploy them faster, ultimately allowing the service providers to increase their revenues with faster Return On Investment (ROI).

The CDMA-2000 evolutionary path was designed to minimize investment and the impact to an operator's network without service interruption for the end-user. This has been achieved through backward and forward compatibility, hardware reuse, in-band migration and hybrid network configuration. This unique feature of CDMA-2000 technologies has provided operators a significant time-to-market advantage over other 3G technologies.

Key advantages of CDMA-2000 technologies include:

- Increased Voice Capacity
- Higher Data Throughput
- Multicast Services

1- Increased Voice Capacity:

The spectral efficiency of CDMA2000 1X permits high traffic deployments in a small amount (1.25 MHz channel) of spectrum. CDMA2000 1X can provide voice capacity of nearly three times that of CDMA-One systems with Selectable Mode Vocoders (SMV) and antenna diversity techniques. CDMA2000 delivers 4-8 times higher voice capacity than TDMA-based technologies.

CDMA2000 1X supports 35 traffic channels per sector per RF (26 Erlangs/sector/RF) using the EVRC vocoder. Voice capacity improvement in the forward link is attributed to faster power control, lower code rates (1/4 rate), and transmit diversity (for single path Rayleigh fading). In the reverse link, capacity improvement is primarily due to coherent reverse link.

2-Higher Data Throughput:

Today's commercial CDMA2000 1X networks support a peak data rate of 153 kbps (Release 0) or 307 kbps (Release 1). CDMA2000 1xEV-DO enables peak rates of up to 2.4 Mbps (Rev. 0) or 3.1 Mbps on the downlink, and 1.8 Mbps on the uplink (Rev A). 1xEV-DO networks deliver the highest data speeds commercially available today.

Average Data Throughput	
CDMA2000 1X	60-100 kbps
CDMA2000 1xEV-DO	400-800 kbps

3-Multicast Services:

With the introduction of EV-DO Release 0 and followed by EV-DO Revisions A and B, operators have the ability to offer multicast services, "one to many" delivery, which allows transmitting the same information to an unlimited number of users without the need to rebroadcast the information multiple times. Multicast functionality offers significant advantages to operators and users. For operators, it allows a vast range of high-revenue generating services with minimum network resources at low cost. For the end-user, multicast services provide access to multimedia content, such as TV broadcasts, MP3 audio files, movies, etc., and a higher quality of services. For 1xEV-DO Release 0, the multicast functionality is referred to as Gold Multicast and for 1xEV-DO Rev A it is called Platinum Multicast.

Additional advantages of CDMA-2000 include:

- Frequency Band Flexibility
- Migration Path
- Serves Multiple Markets
- Supports Multiple Service Platforms
- Full backward compatibility
- Increased Battery Life
- Synchronization
- Power Control
- Soft Hand-off
- Transmit Diversity
- Voice and Data Channels
- Traffic Channel
- Supplemental Channels
- Turbo Coding

3.7 CDMA Applications Overview

CDMA2000 was designed with the Internet in mind, making it the ultimate platform on which to build innovative applications. CDMA2000 users are experiencing a host of advanced services, including web browsing, m-commerce, MMS (multimedia messaging services), streaming video, games, enterprise solutions and email. To meet future demand for data services, CDMA2000 operators are building their portfolios at a rapid pace, creating valuable opportunities for applications developers and content providers.

Demand for wireless data services and applications is taking off around the world and CDMA2000 operators are leading the industry in the range of services they offer, number of users they serve and revenue they generate from data. Capitalizing on the high-speed data capabilities and flexibility of CDMA2000 technologies, operators have introduced a variety of applications which enable their consumer and enterprise customers to access information, surf the Web, download music and video, send pictures and play games. As a result, operators have seen a phenomenal uptake in data services and significant increase in data revenue.

CDMA2000 is an ideal platform for building advanced applications. It delivers high data speeds of up to 3.1 Mbps - faster than fixed-line broadband solutions and it supports multiple application platforms, including BREW and Java. There is a wide range of devices with advanced data functionality for consumers and enterprise to choose from.

- **Enterprise Solutions**

The inherent high-speed data capabilities and security aspects of CDMA2000 make it an ideal technology for workers to stay connected while mobile. Gaining access to email, the Internet and corporate Intranets via handsets or laptops equipped with modem cards are key drivers for companies looking to take advantage of mobile high-speed data services.

Wireless data services offer opportunities across both horizontal and vertical markets. Horizontal segments include field sales, field services and mobile office workers who can benefit from anytime, anywhere access to mobile data. Many CDMA2000 operators offer a wide range of services for the enterprise sector. Some specific industries already making use of wireless data are public safety, insurance, healthcare and pharmaceutical.

Location-Based Services

Location-based services (LBS), which encompass a broad range of applications specific to a user's position, represent an important source of differentiation and revenue for operators. CDMA2000 operators have introduced a wide variety of innovative services, such as traffic information, navigation and tracking. With these services, their consumer and enterprise users can obtain updates on traffic flow and directions to the nearest bank or restaurant, find their friends and manage their fleets.

Multimedia Messaging Services (MMS)

MMS enable graphics, pictures, video or music to be attached to text messages and sent to mobile devices or computers. These services are widely deployed by CDMA2000 operators in Korea and Japan and are gaining momentum elsewhere around the world, including the U.S. Carriers are seeing good uptake of MMS and enjoying the resulting revenue.

Push-to-talk

Push-to-Talk IPRS (IP Radio Service) always-on instantaneous communication technology enables one-to-one and one-to-many, two-way voice and data sessions that do not require dial-up – subscribers push a button and begin speaking.

3.8 CDMA-450 (Additional Topic)

CDMA450 is a TIA-EIA-IS-CDMA2000 (CDMA-MC) system deployed in 450 MHz which includes a family of standards developed by 3GPP2, published by TIA and approved by ITU for IMT-2000: CDMA2000 1X, CDMA2000 1xEV-DO and CDMA2000 1xEV-DV. Currently, CDMA2000 1X and CDMA2000 1xEV-DO are commercially available for the 450 MHz band and CDMA2000 1xEV-DV is being developed.

Advantages of CDMA450

CDMA450 advantages derive from the spectral efficiency and high-speed data capabilities of CDMA2000 and the expanded coverage afforded by a lower frequency band.

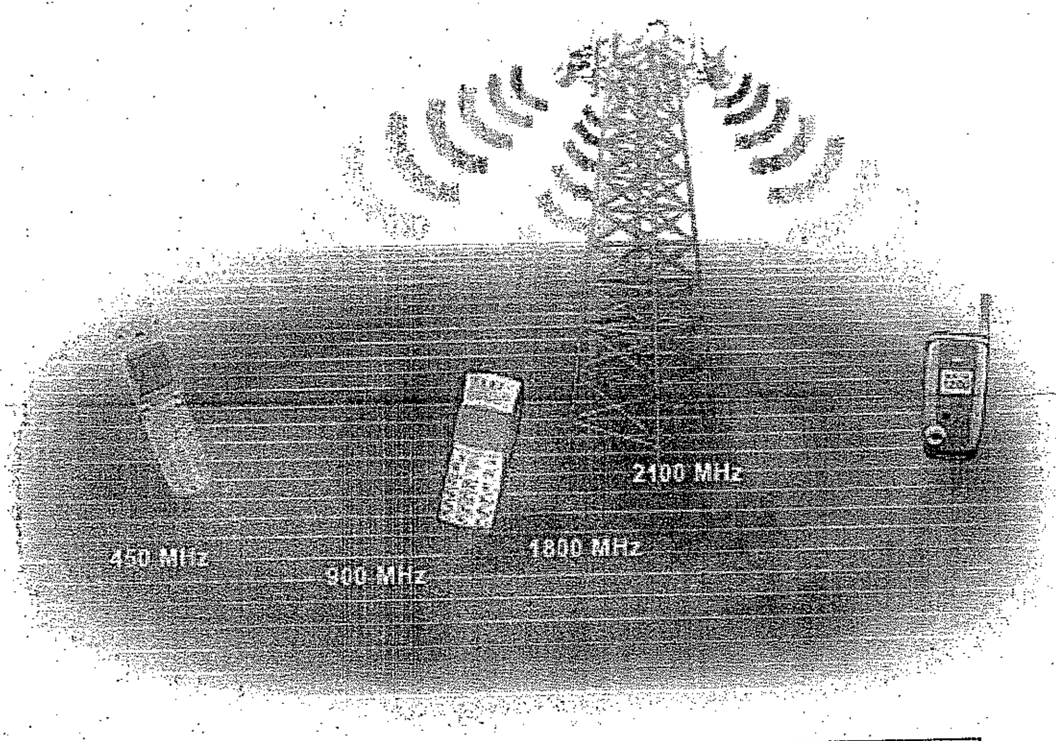


Figure 3-11: CDMA2000 expanded coverage at lower frequency band

- CDMA450 provides a larger cell size compared to cell sizes in other bands, which translates to fewer cell sites and significantly lower capex and opex to service vast coverage areas.

- CDMA450 offers IMT-2000 services: high-quality voice and high-speed data access.
- CDMA2000 1X allows for voice capacity of up to 20 enlarges per sector/carrier.
- CDMA2000 1X supports high-speed data up to 153 kbps and CDMA2000 1xEV-DO offers broadband access up to 2.4 Mbps.
- Clear evolution path to advanced 3G services.
- CDMA450 requires only a small amount of spectrum (1.25 MHz), a significant consideration for NMT450 operators who have 4-5 MHz allocated to them.
- CDMA450 allows for a phased evolution.

Review questions :
Q1 : Compare between AMPS ; TDMA ; and CDMA Technologies as the following table :

Technology	System	Operating frequency	Capacity	Security
AMPS				
CDMA				
TDMA				

Q2 :

Write down the advantages of CDMA over others ?

-
-
-
-

FIGURES TABLE KEY

APPENDIX I

1. What is the Power?

Power is defined as the quantity of energy dissipated per unit of time. **Electric power** is defined as the amount of work done by an electric current, or the rate at which electrical energy is transmitted. The SI unit of power is the watt.

1.1 The units of Power:

The power expressed at the unit of "Watt", KW, and mW.

Where all are an absolute power values, where:

Watt = 1000 mW

KW = 1000 Watt

The **watt** (symbol: W) is the SI derived unit of power, equal to one joule per second. A human climbing a flight of stairs is doing work at the rate of about 200 watts; a highly trained athlete can work at up to approximately 2000 watts for brief periods. An automobile engine produces 25 000 watts (approximately 30 horsepower) while cruising. A typical household incandescent light bulb uses 40 to 100 watts.

The **watt** is named after James Watt for his contributions to the development of the steam engine, and was adopted by the Second Congress of the British Association for the Advancement of Science in 1889 and by the 11th Conférence Générale des Poids et Mesures in 1960.

SI multiples:

Multiple	Name	Symbol	Multiple	Name	Symbol
10^0	**watt**	W			
10^1	decawatt	daW	10^{-1}	deciwatt	dW
10^2	hectowatt	hW	10^{-2}	centiwatt	cW
10^3	kilowatt	kW	10^{-3}	milliwatt	mW
10^6	megawatt	MW	10^{-6}	microwatt	µW
10^9	gigawatt	GW	10^{-9}	nanowatt	nW
10^{12}	terawatt	TW	10^{-12}	picowatt	pW
10^{15}	petawatt	PW	10^{-15}	femtowatt	fW
10^{18}	exawatt	EW	10^{-18}	attowatt	aW
10^{21}	zettawatt	ZW	10^{-21}	zeptowatt	zW
10^{24}	yottawatt	YW	10^{-24}	yoctowatt	yW

1.2 Decibel:

The **decibel (dB)** method of calculation, that uses a logarithm to allow very large or very small relations to be represented with a conveniently small number (similar to scientific notation). It is a dimensionless unit of ratio which is used compares two usually-variable quantities. When used to compare a variable quantity to a known reference quantity the measurement is qualified with a suffix such as with dBm where the reference quantity is one milliwatt. Decibels are useful for a wide variety of measurements in acoustics, physics, electronics and other disciplines.

The decibel is not an SI unit, although the International Committee for Weights and Measures (CIPM) has recommended its inclusion in the SI system. Following the SI convention, the *d* is lowercase, as it is the SI prefix *deci-*, and the *B* is capitalized, as it is an abbreviation of a name-derived unit, the *bel*, named for Alexander Graham Bell. Written out it becomes *decibel*. This is Standard English capitalization.

1.3 Radio Power:

It is convenient to express the power level in (dB_m) unit, where:

$$\text{Power } (dB_m) = 10 \log \text{Power (mW)}$$

Example:

Convert the power level that equals to 1.5 watt to (dB_m) units?

Solution:

The power (mW) $= 1.5 \times 1000$

$= 1500$ mW

» Power (dB_m) $= 10 \log$ Power (mW)

$= 10 \log 1500 =$ dB_m

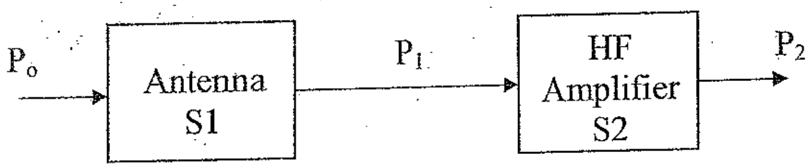
Note: Remember that the dB_m is an expression of power level.

i.e. You can say that this transmitter delivers power at level equal to 30 dB_m.

» The power $= \log^{-1} \frac{30}{10} = 1000$ mW $= 1$ watt

1.4 Gain and Attenuation units:

Assume that you intend to analyze the following stages of a communication system:



Where:

P_o = the received power in (mW)

P_1 = the output power of antenna stage S1 (mW)

P_2 = the output power of HF amplifier S2 (mW)

Assume that $P_o = 10$ mW

$P_1 = 20$ mW

$P_2 = 2$ KW

So, the Gain (G) is defined as the ratio of output power to the input power of that is: $G_1 = \frac{P_1}{P_o}$ and $G_2 = \frac{P_2}{P_1}$

i.e. $G_1 = \frac{20}{10} = 2$

$$G_2 = \frac{2 \times 1000}{20} = 100$$

This means that in stage 1 (S1) amplify the input power 2 times.

(i.e. P_1 = double P_o)

In stage 2 (S2) P_2 is 100 times amplified of P_1 (i.e. $P_2 = 100\ P_1$)

Due to the large difference in the stages amplification within the over all system, it is convenient to express a most popular unit decibel (dB).

Decibel (dB) is defined as the relative unit of power. This means that the level of power is related to a reference value.

Thus, we can say that:

$$G_1\ (\text{dB}) = 10 \log \frac{P_1}{P_o} \qquad \text{or} \qquad G_2\ (\text{dB}) = 10 \log \frac{P_2}{P_1}$$

And the signal power (P_s) level with the respect to the Power of Noise (PN) is:

$$\text{PN} = 10 \log \frac{P_S}{P_N}\ (\text{dB})$$

APPENDIX II
LIST OF ABBREVIATIONS

TDM	Time Division Multiplexing
ATM	Asynchronous Transfer Mode
PSTN	Public Switched Telephone Network
SNR	Signal-to-Noise Ratio
dB	Decibels
NF	Noise Figure
NR	Noise Ratio
BER	Bit error rate
AM	Amplitude Modulation
FM	Frequency Modulation
PM	Phase Modulation
FSK	Frequency Shift Keying
PSK	Phase Shift Keying
BPSK	Binary Phase Shift Keying
QAM	Quadrature Amplitude Modulation
PAM	Pulse Amplitude Modulation
PPM	Pulse Position Modulation
PWM	Pulse Width Modulation
PCM	Pulse Code Modulation
SDH	Synchronous Digital Hierarchy
FDM	Frequency Division Multiplexing
PDH	Plesiosynchronous Digital Hierarchy
TWN	Telecommunication Management Network
AMPS	Advance Mobile Phone System
TACS	Total Access Communication System
NMT	North Mobile Telephone

PCS	Personal Communication System
TDMA	Time Division Multiple Access
CDMA	Code Division Multiple Access
FDMA	Frequency Division Multiple Access
SDMA	Space Division Multiple Access
CSMA	Carrier-sense Multiple Access
RACH	Random Access Channel
TDD	Time Division Duplex
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communication System
DECT	Digital Enhanced Cordless Telecommunications
ITU	International Telecommunication Union
ADC	Analog-to-Digital Converter
DAC	Digital-to-Analog Converter
STDM	Statistical Time Division Multiplexing
SMDS	Switched Multi-Megabit Data Service
TASI	Time Assignment Speech International
PBX	Private British Exchange
DSI	Digital Speech Interpolation
WDM	Wavelength Division Multiplexing
DWDM	Dense Wavelength Division Multiplexing
DS	Direct Sequence
FHSS	Frequency Hopping Spread Spectrum
DSSS	Direct Sequence Spread Spectrum
CDPD	Cellular Digital Packet Data
UMTS	Universal Mobile Telecommunication System
SMV	Selectable Mode Vocoders
MMS	Multimedia Messaging Service
LSB	Location-Based Services

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معــعــهــد الاتــصـــالات و المـــلاحــة

Telecommunication & Navigation Institute

الخطة الدراسية للمقرر النظري

Theoretical course Lesson Plan

Transmission	Dept. القسم	Telecommunication - 11	Course Name اسم المقرر
4	Weekly Hrs. عدد الساعات الأسبوعية	TR 151	Course Code رمز المقرر

Course Objectives أهداف المقرر

On completion this course the trainee will know :

- The different types of modulation and principles
- Analogue and digital access technology
- The Types of noise affect the signal
- Multiplexing Technology
- CDMA-1 and CDMA-2000 Systems

Weeks	Content
1	Introduction noise effect
2	Analogue modulation (AM & FM & PM)
3	Digital modulation (ASK & FSK & QPSK)
4	Pulse modulation
5	PCM system & SDH
6	Multiple access in radio
7	**MID –TERM EXAMINATION**
8	TDMA &SDMA & FDMA technology
9	CDMA-1 (IS-95)
10	CDMA -2000 technology
11	CDMA-2000 advantages
12	CDMA-2000 1x and EV-1x Applications
13	Radio structure using CDMA technology

Weeks	Content
14	**Final Review**
15	موعد بداية الاختبارات النهـــائيـــة **Final Exams**
16	

موعد اختبار منتصف الفصل

Mid-term Exam Date

20 / /

يبلغ المدرب/المدرس جميع المتدربين بالموعد

توزيع الدرجات الكلية والتي مجموعها **100** درجة

Grading Policy (Total of 100)

الاختبار الفصلي Final Exam	% 60
اختبار منتصف الفصل Mid-term Exam	% 20
الاختبارات القصيرة و التطبيقات Quizzes & Applications	% 10
النشاط Classroom/Lab Activities	% 5
الانتظام Attendance	% 5

ختم القسم

Dept. Seal