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Electronic Communications Systems
and the Frequency Domain:
An Illustrated Primer for C3 Students

by

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19 Abstract *(continue on reverse if necessary and identify by block number*

This thesis is a tutorial for Command, Control, and Communications (C3) students and synthesizes the essence of electronic communications systems and related frequency spectrums into an integrated set of illustrations. The objective is to help the C3 student focus his or her attention on the performed operations and transformations in order to acquire a better understanding for the processes involved. The first major illustration shows how the various components of a communication system are related. These components include analog and digital signals, multiplexing, modulation, and various transmission mediums. Central to any communicxations system is the limitations imposed by bandwidth and noise. The concept of bandwidth is developed throught Fourier analysis. An integrated set of graphics shows the realtionship between the time and frequency domains and illustrates how the bandwidth increases as the pulse width decreases. Transmitting information often requires higher data rates which, in turn, require higher frequencies. Radio wave propagation is frequency dependent and a chart is developed showing the different categories of radio wave propogation as they relate to atmospheric layers and frequency. Finally, a chart relating transmission medium attenuation, noise sources, and various radio wave terminology is given.

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Electronic Communications Systems and the Frequency Domain: An Illustrated Primer for C3 Students

by

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Submitted in partial fulfillment of the requirements for
the degree of

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from the

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June 1990

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ABSTRACT

This thesis is a tutorial for Command, Control, and Communications (C3) students and synthesizes the essence of electronic communications systems and related frequency spectrums into an integrated set of illustrations. The objective is to help the C3 student focus his or her attention on the performed operations and transformations in order to acquire a better understanding for the processes involved. The first major illustration shows how the various components of a communication system are related. These components include analog and digital signals, multiplexing, modulation, and various transmission mediums. Central to any communications system is the limitations imposed by bandwidth and noise. The concept of bandwidth is developed through Fourier analysis. An integrated set of graphics shows the relationship between the time and frequency domains and illustrates how the bandwidth increases as the pulse width decreases. Transmitting information often requires higher data rates which, in turn, require high frequencies. Radio wave propagation is frequency dependent and a chart is developed showing the different categories of radio wave propogation as they relate to atmospheric layers and frequency. Finally, a chart relating transmission medium attenuation, noise sources, and various radio wave terminology is given.

I. INTRODUCTION

A. PURPOSE

This thesis illustrates the key features of electronic communication systems for individuals studying Command, Control, and Communications (C3). The fundamental elements are shown from a relational perspective by using graphics rather than wordy explanations. The objective is to focus the reader's attention on the performed operations and transformations to nurture an intuitive feel for the processes involved. This approach will help mitigate the tendency of students to "lose the forest through the trees" due to the vastness of the subject and confusing terminology when first exposed to communications.

The focus here is primarily on the frequency spectrum and bandwidth concepts. Information theory and channel coding (i.e., error correcting codes) are very important in electronic communications, but are left to development in another thesis. This thesis is structured to supplement coursework material at the Naval Postgraduate School.

B. STRUCTURE

The thesis is split into six sections. Requirements - The Command and Control Connection, Communications Systems Overview, Information and Bandwidth, Encoding, Modulation, and Channels-Transmission Mediums.

The Requirements section emphasizes communications systems exist for meeting the needs of the commander. Clearly defining the requirements is the most important aspect of designing any new communication system.

The next section gives a general overview of a communications system. A chart illustrating the "big picture" is introduced. It shows how the main elements of communication systems are related and serves as the road map for the remaining sections.

Communications systems pass various types of information requiring different data rates and throughput. Nevertheless, the bandwidth and noise of the channel limits the throughput any system can handle. This requirement motivates the section entitled Information and Bandwidth. Information and Bandwidth lays the foundation for understanding the relationship between a signal's time and frequency domain via Fourier analysis.

Encoding modifies the information signal in a way that provides for more efficient transmission. Converting an analog signal to a digital signal improves the signal's performance in the presence of noise. Multiplexing makes for efficient channel use.

Modulation is a form of encoding that makes long-haul communications feasible. It does this by impressing the information signal onto a high frequency continuous wave (sinusoidal signal) that "carries" the information signal to its destination.

Transmitting information quickly is often desired. However, higher data rates require wider bandwidths. The bandwidth limiting characteristics of the channel constrains the information transfer rate. The section entitled Channel-Transmission Mediums discusses the guided (copper, fiber optics) and unguided media (air, space) as well as transmission impairments. Figure 1-1 illustrates the thesis structure.

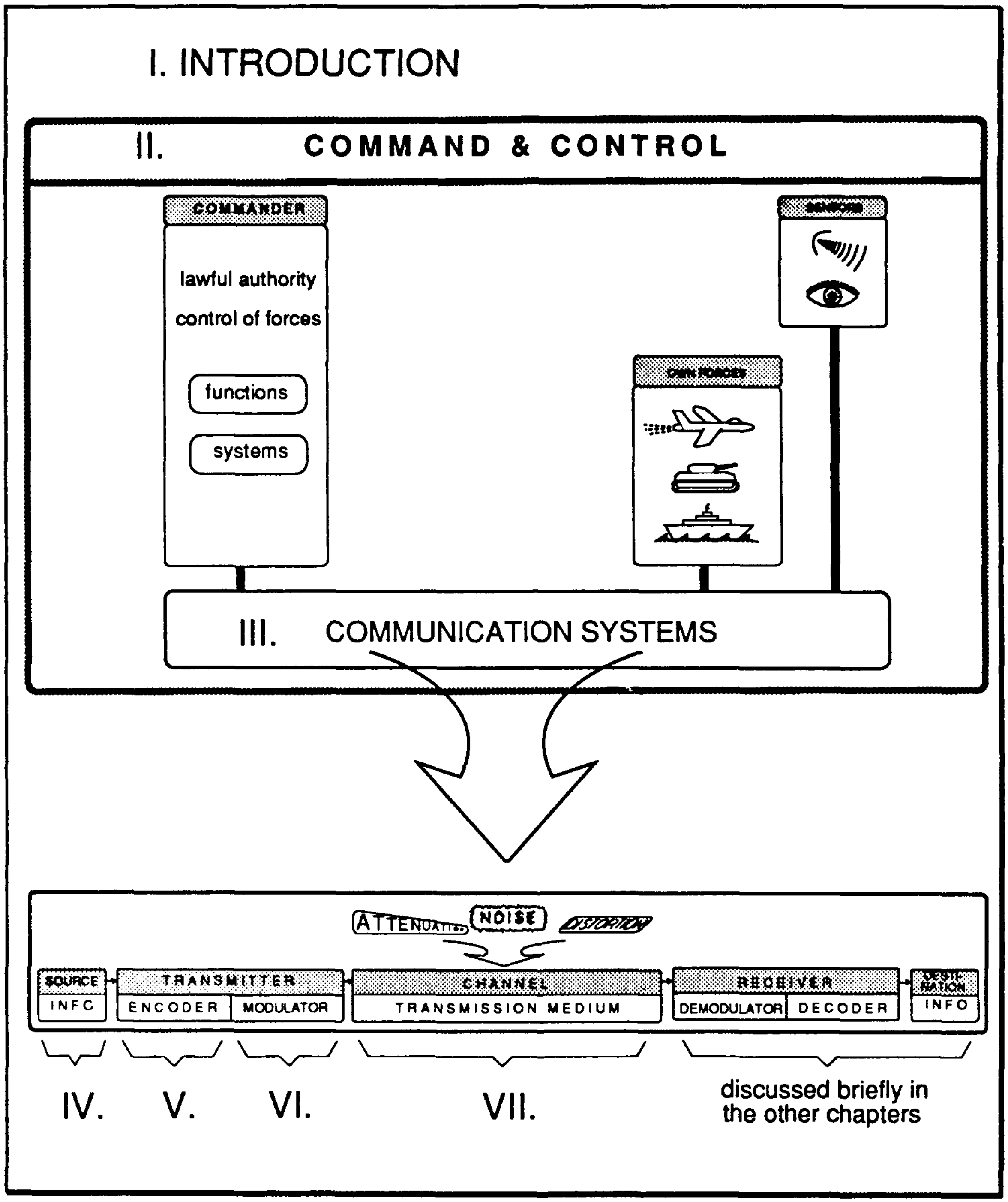


Figure 1-1. Thesis Structure

II. REQUIREMENTS - THE COMMAND AND CONTROL CONNECTION

This chapter provides a perspective of the role communications plays in the Command and Control (C2) arena and places the technical material in this thesis in perspective. The chapter emphasizes the need to keep the commander at the center of thought when defining requirements or designing a communication system.

A. COMMAND AND CONTROL

C2 is a very broad topic and means various things to different people. The result is a myriad of terms defining Command and Control that causes confusion. These terms include: Command, Control, and Communications (C3); Command, Control, Communications, and Intelligence (C3I); Command, Control, Communications, Computers, and Intelligence (C3I); Command, Control, Communications, Computers, Intelligence, and Interoperability (C3I2).

The author views C2 as the fundamental descriptor, with all other C#I# terms simply a subset of C2. In other words, C2 contains the other subcategories. For example, a commander needs communications (C3) to give direction to his troops. He also needs intelligence (C3I) to make informed decisions for directing his assigned forces.

The official Joint Chiefs of Staff Publication 1 (JCS Pub 1) definition of Command and Control is included here as a common reference point. It is:

> The exercise of authority and direction by a properly designated commander over assigned forces in the accomplishment of a mission. Command and control functions are performed through an arrangement of personnel, equipment, communications, facilities, and procedures employed by a

> commander in planning, directing, coordinating and controlling forces and operations in the accomplishment of the mission.

Personnel, equipment, communications, facilities, and procedures are collectively considered a Command and Control *system*. The JCS Pub 1 definition is:

> The facilities, equipment, communication, procedures, and personnel essential to a commander for planning, directing, and controlling operations of assigned forces pursuant to the missions assigned.

Figure 2-1 provides a graphical representation of the C2 definition with C2 systems providing the foundation. The primary purpose of the C2 system is to meet the needs of the commander. A command and control system can be divided into four parts for clarifying the primary C2 functions (Hoever 88):

- recognized point of authority
- resource which can be controlled by an authority
- means to control resources by authority
- means to perceive environment directly or indirectly

If an official JCS definition exists, why are other terms needed? The additional terms are used in order to emphasize a particular aspect of a C2 system. Communications (C3) and Intelligence (C3I) were added to emphasize the fundamental role they have in the command and control process. Computers (C4I) were added because they are essential to modern communications and commander's decision aides. Interoperability (C4I2) was added after the Goldwater-Nichols DoD Reorganization Act of 1986 emphasized "jointness" and the need for more efficient use of defense resources. (Hoever 88)

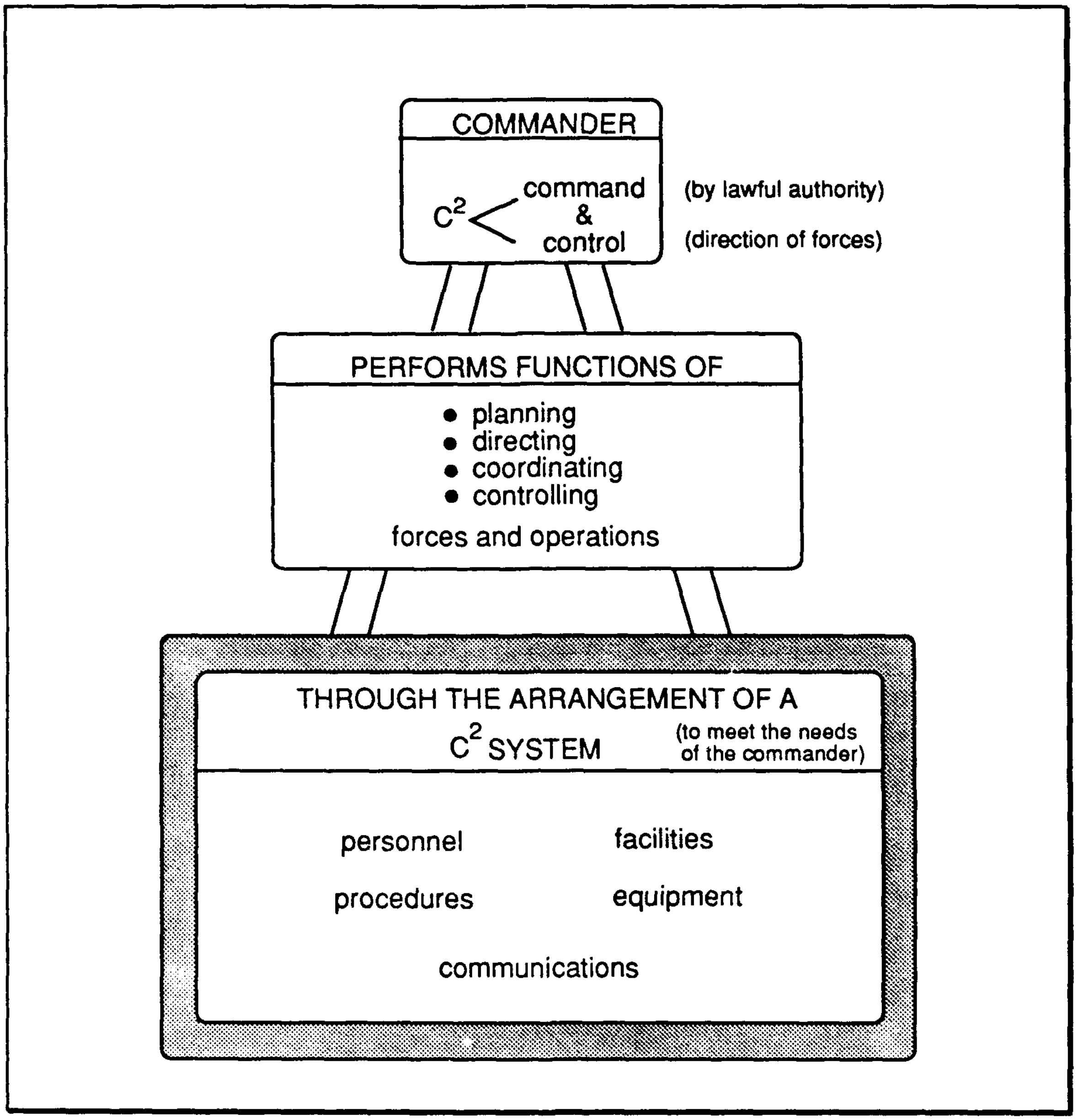


Figure 2-1. Command and Control

Figure 2-2 shows a more liberal description of a C2 system and identifies key elements that are added to further enhance the C2 definition (Bethman 89, p. 20). Note, however, that a communication system is only one aspect of command and control. A communication system is frequently the center of attention while procedures, doctrine, and personnel are pushed to the back seat. Perhaps this is because communications are a tangible and quantifiable entity people can see, use, test, and easily identify when failure occurs. (Jones 89)

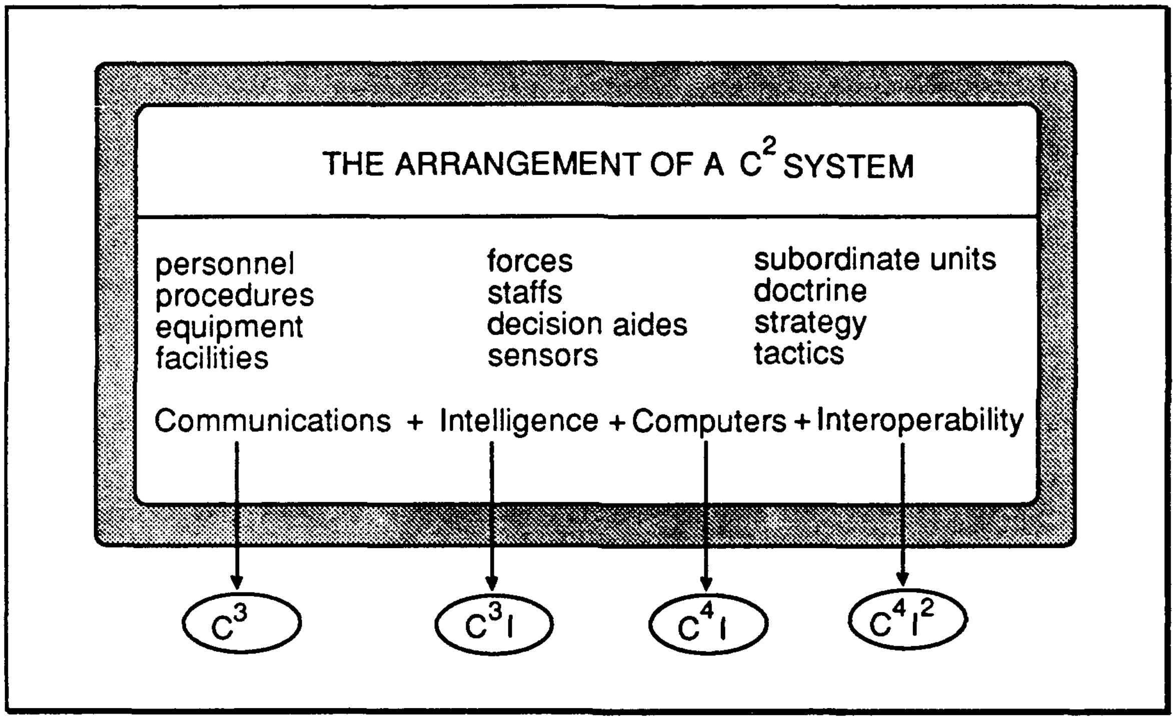


Figure 2-2. Command and Control System Detailed

Figure 2-3 illustrates a link between a commander and his forces and sensors. The functions inside the commander's block (sense, process, compare to desired state, decide, act) are part of a commander's decision-cycle loop and is known as the *Lawson C2 model.* These are processes for reaching and making a decision. Other terms may also be used, such as observe, orient, decide, and act. This loop is also know as the *Boyd OODA loop*. (Orr 83, p. 26)

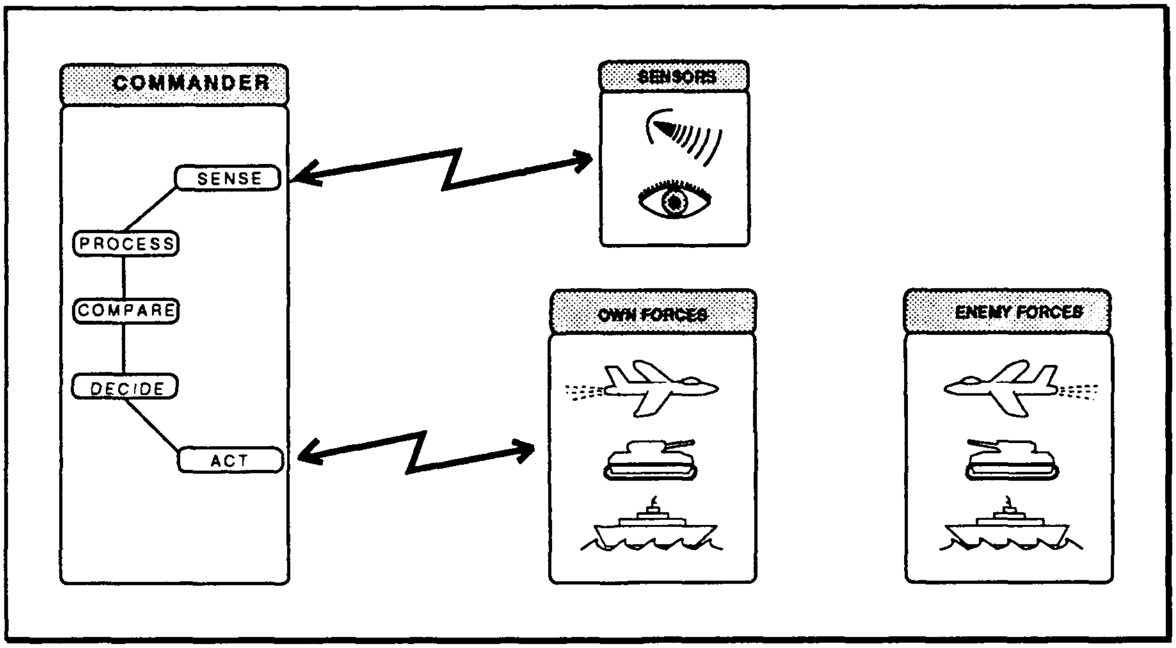


Figure 2-3. Commander, Forces, Sensor Linkage

B. REQUIREMENTS

The second focus of this chapter is on the requirements. Clear and functional communication requirements are needed to ensure the communication system will meet the commander's needs. Functions are broad activities to support the commander's assigned missions. Therefore, requirements should first be viewed at the top level from the commander's perspective. Once the major functions are identified, the following broad requirements can be addressed: What type of information needs to be passed? How much? How fast does it have to get there? Who is to receive the information and where does it go? This is not intended to be a checklist, but to stress that a great deal of thinking needs to take place before technical specifications are set. It cannot be assumed that "obvious" requirements will be met. The following real world situation is an example of how a simple requirement can be miscommunicated between user and developer.

A requirement was levied on a contractor to develop a secure voice system. The system was designed and demonstrated for the commander. It provided the "successful" transmission of secure voice, but the system was immediately rejected because the voice sounded like Donald Duck. The Commander required that he be able to recognize the person on the other end of the line. (Jones 89)

One must ask how such an "obvious" requirement could be overlooked. Perhaps the developers and action officer did not think it was important, or perhaps they were so busy trying to solve specifics that they failed to perceive what was really needed. They may have focused on the digitization process, encryption schemes, line attenuation, or impedance matching characteristics. All of these are necessary components to build a physical device, but they are worthless if the system is not used. Therefore, a considerable amount of time should be spent on understanding and defining clearly the requirements before any specific design work is initiated. (Star 89)

Human cognitive capabilities are another factor to consider when one is defining requirements. Cognition is a broad term for how humans perceive their environment and mentally process data. One major finding of cognitive studies reveals that humans are able to process about five to nine "chunks" of new information at a time. (Jones 89)

Given too much data the commander will be overwhelmed and the result is a negative impact on the mission. Moreover, funds can be wasted in designing a high data-rate system when an austere system would have sufficed. On the other hand, given too little data the commander is unable to make an informed decision. Data may also be presented in such a way that the commander latches onto a specific piece of information while paying little attention to other pieces and arriving at a wrong conclusion (Edwards 86, p. 247). Therefore, one needs to understand the communication system truly as a *system*, not a mere collection of equipments or library of technical specifications. The human element must always be considered.

The systems designer should keep cognition factors in mind when deciding upon what data and how much of it to give the commander. In the author's opinion, there is a strong advantage to presenting any idea with a picture, if possible. When listening to a verbal description a listener creates a mental image. However, the mental image could be wrong and depends on how the listener interprets the verbal description. Differences in meaning often occur because people filter words and apply their own meaning based on their personal frame of reference.

There is a delicate balance between providing the commander with too little vice too much information. However, once the basic requirements are understood, one can proceed to technical requirements. Technical requirements may include the following (Roden 88, p.445):

- required bit transmission rate
- maximum allowable bit error rate

- maximum system bandwidth
- maximum transmitted signal power (and signal-to-noise ratio)
- maximum construction cost (complexity of detector)
- maximum acquisition time of detector

Design engineers take the above details under account. However, it is the responsibility of the C3 professional to act as the bridge between the commander's functional requirements (to accomplish command and control functions) and the engineer's technical design requirements (to build the equipment). Tradeoffs are inevitable, but the focus must be on what is really needed. Figure 2-4 focus on the fundamental communication link questions and the resulting requirements. The next chapter discusses the fundamental components of communication systems.

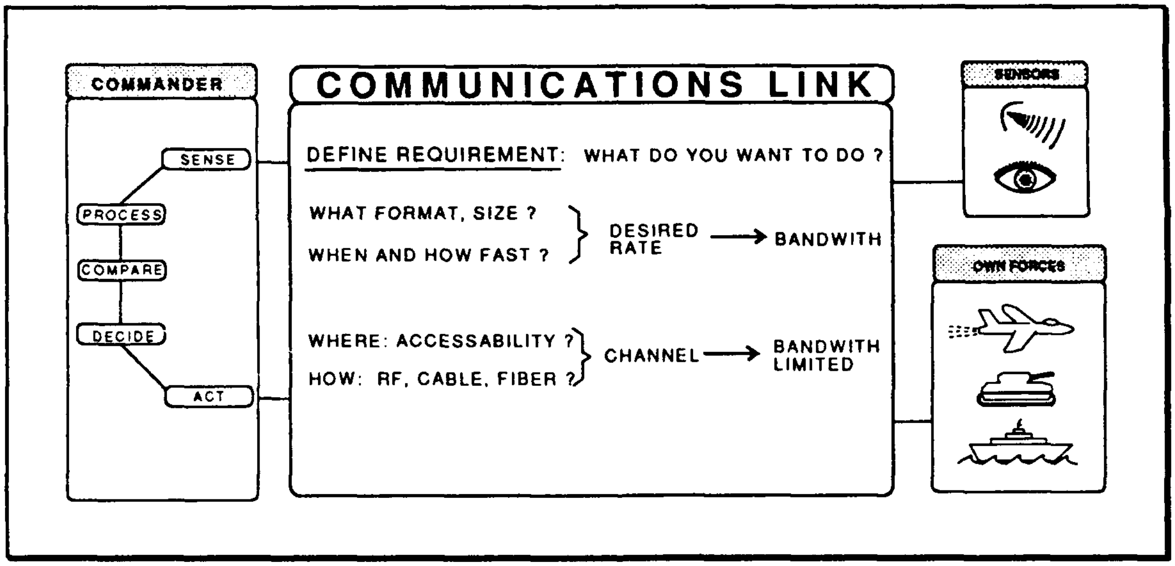


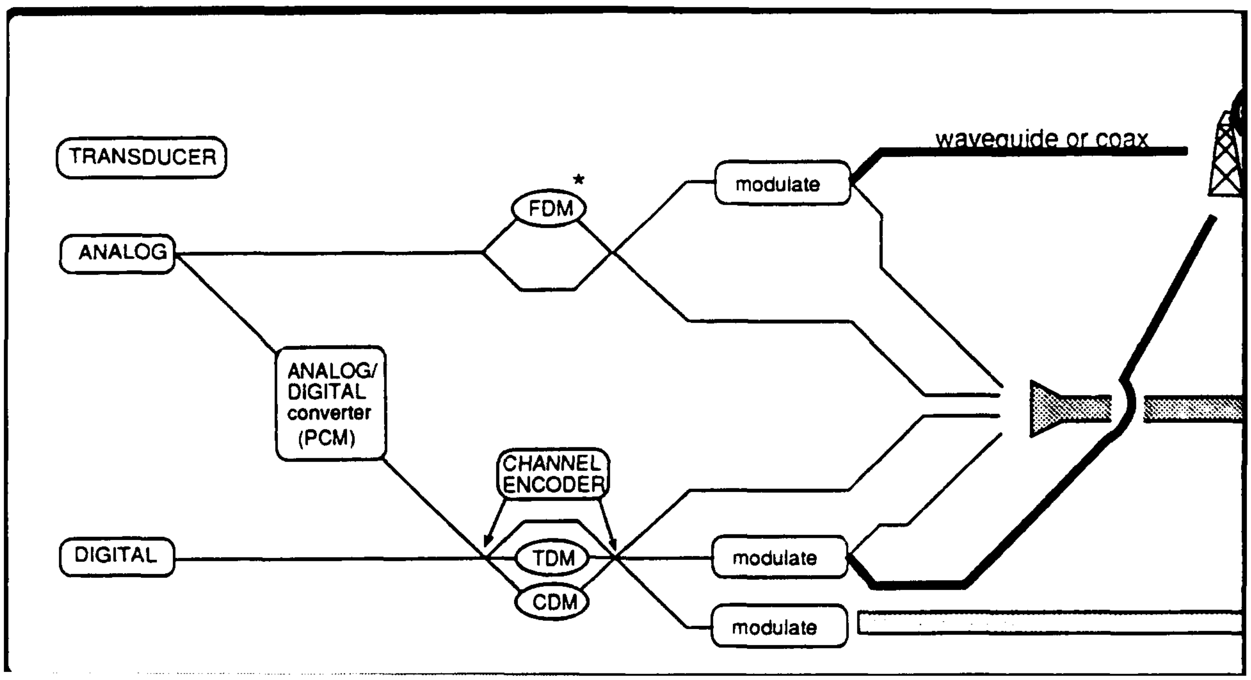
Figure 2-4. Determining Fundamental Requirements for the Communications Link

III. COMMUNICATION SYSTEM OVERVIEW

The previous chapter emphasized that a communication system (as part of the C2 system) exists to meet the needs of the commander. This chapter provides a general overview of communication systems.

A comprehensive view of a communication system is shown in Figure 3-1. The objective is to show how various components of a communication system are related. When classroom or textbook discussion focuses on a specific communication detail in the system, it is useful to refer to this chart to determine where the specific function is taking place in the communication process, what is being done to the information, and why. The figure is split into three charts. Chart (a) is a breakdown of basic communication subsystems. Chart (b) shows a communications road map. Chart (c) displays a functional detail of the major components. Many of the terms in the chart will be explained later in the thesis.

Keeping a proper perspective also pertains to equations. When any equation is being applied, one should know where it applies in the communications process and how it is applied. The reader should view any equation as simply a model representing an approximation of real world behavior. The model may be a linear or nonlinear device or system. Simplifying assumptions concerning the real world behavior are inherent in the model and should be examined. These assumptions are critical in understanding the limitations of the model's applicability. It is easy to get lost in a maze of mathematics and lose sight of what is actually going on. (Giordano 85, p. 32)



TRANSDUCER
ANALOG
DIGITAL
FDM
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ANALOG/
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converter
(PCM)
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ENCODER
TDM
CDM
modulate
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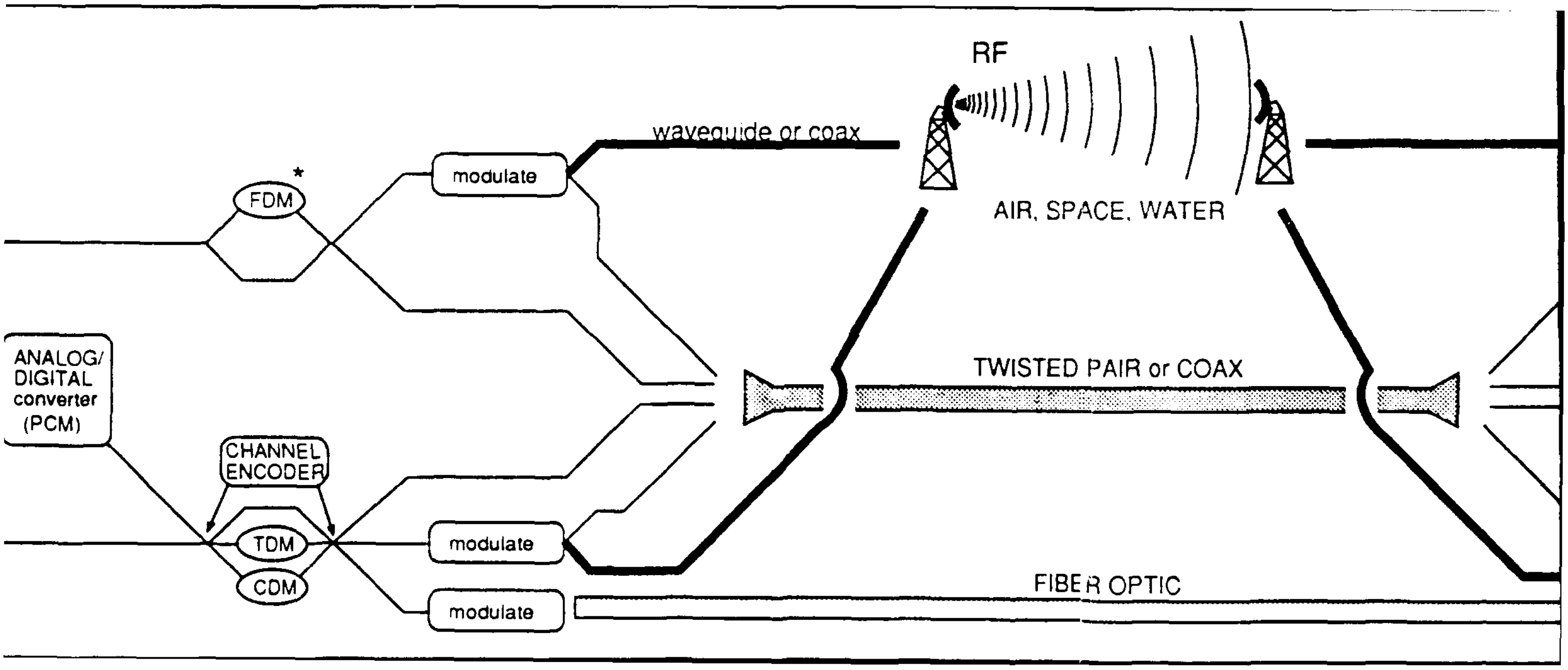
modulate
waveguide or coax



RF
waveguide or coax
modulate
FDM
*
AIR, SPACE, WATER
ANALOG/
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converter
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TWISTED PAIR or COAX
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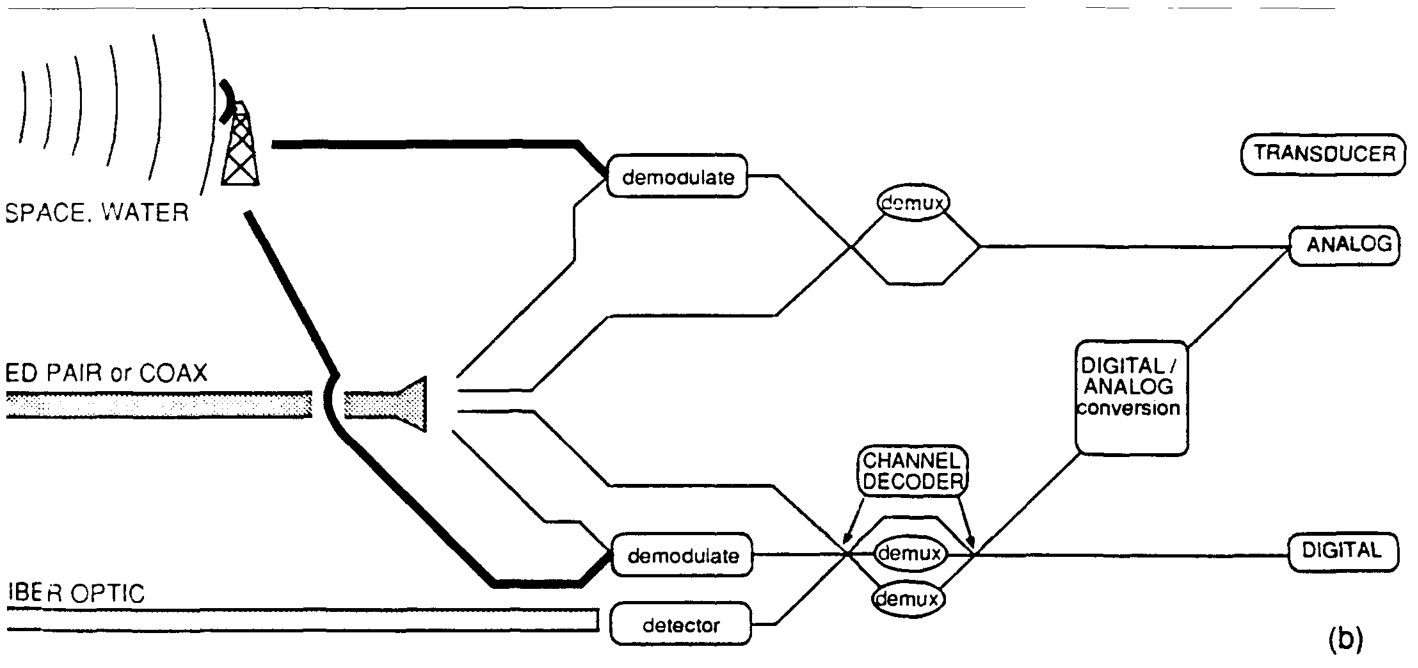
TDM
modulate
CDM
modulate
FIBER OPTIC



SPACE. WATER
ED PAIR or COAX
IBER OPTIC
demodulate
demodulate
detector
demux
demux
demux
CHANNEL DECODER
DIGITAL / ANALOG conversion
TRANSDUCER
ANALOG
DIGITAL

(b)

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The purpose of a communication system is the transmission of information from one place to another. The goal is to "reproduce at the destination an acceptable replication of the source message." (Carlson 86, p. 2) Communications are usually thought of as an immediate "from here to there" process. This interpretation has a very broad application and meaning. For example, "from here to there" could apply at the level of the computer chip when the arithmetic logic unit sends results to a register. It could also apply to the case of vast distances, such as transmitting pictures of Neptune from the Voyager spacecraft back to Earth. Another interpretation is communications as a "from now till then" process. For example, a tape recorder contains a magnetic tape on which the information is stored for later reception. (Hamming 90) This thesis deals with comn ınication in the first, more immediate sense of "from here to there."

A. BASIC COMMUNICATION SUBSYSTEMS

Most books on electronic communications have introductions showing some type of transmitter, a transmission channel, and a receiver. (Some authors use the terms encoder and decoder for transmitter and receiver, respectively). The transmitter can also be subdivided into an encoder and modulator. Likewise the receiver can be subdivided into a demodulator and decoder. (Stremler 90, p. 3)

1. Channel

The channel provides the link between source and destination. However, the channel attenuates, distorts and adds noise to the signal. The channel is the limiting factor in the performance of any well-designed communication system. Therefore, the transmitter's role is to prepare the signal in a manner that minimizes the limitations imposed by the channel. The receiver then inverts the transmitter operations and (ideally) recovers the information with a minimum amount of error. (Stremler 90, p. 3)

The limitations imposed by the channel restrict the information rate. The maximum theoretical rate a given channel can support is called *channel capacity* and was modeled by Claude Shannon. The following channel capacity formula assumes a bandlimited channel operating in the presence of additive white gaussian noise:

$$C = B \log_2 (1 + S/N)$$

where C is the capacity in bits per second, B is the bandwidth in cycles per second (hertz), and S/N is the signal to noise ratio. From the formula it is clear that one could increase the signal power, decrease the noise, or increase the bandwidth to increase data rate. However, these factors cannot be increased beyond certain amounts. The amount of power generated, or the amount of power antennas can handle, is limited. Also, as the bandwidth increases, more noise is capable of entering the receiver and the magnitude of the noise increases as well, thus reducing S/N. One can increase channel capacity by raising the bandwidth until the rate of change of log S/N exceeds the rate of the bandwidth. Although theoretical limits have not been obtained by practical systems, they provide a good bound on the problem. (Roden 88, p. 159)

2. Transmitter/Receiver

Note that the transmitter is split into two components: encoder and modulator. An encoder modifies a signal to improve transmission efficiency. This term encoding is broad and has several meanings, including source coding and channel coding. *Source coding* encompasses formatting and data redundancy reduction or data compression. Formatting makes an analog source compatible with digital processing (Pulse Code Modulation-sample, quantize, encode). Data reduction is the act of representing source symbols so the average code length is minimized. (Sklar 88, p. 52; Hamming 86, p. 2)

Channel coding, improves performance by allowing the transmitted signal to better withstand the effects of various channel impairments, such as noise, fading, and jamming (Sklar 88, p. 246). Channel coding is encompasses error detection and correction, or encryption. For example, the pictures from the Voyager spacecraft were extremely weak and contained hundreds of errors. However, coding allowed for correction of these errors on earth and one could see these amazing pictures quite clearly. (Hamming 90)

Source and channel coding are based on information theory and are not discussed in this thesis. Other forms of encoding such as Pulse Code Modulation and multiplexing are discussed.

Modulation can also be considered as a form of encoding since it prepares the signal for efficient transmission. The difference is that coding optimizes error free detection whereas modulation impresses information on a carrier to optimize compatibility with the channel. In other words, modulation shifts or translates the information signal to a higher frequency to make possible the propagation of electromagnetic waves. Modulation is discussed in greater detail in Chapter VI. (Schwartz 90, p. 203; Stanley 82, p. 6)

Demodulation and decoding are the inverse operations of modulation and encoding and occur at the receiver end.

B. COMMUNICATIONS ROAD MAP

Communications is such a vast subject that a road map was developed to give the reader a perspective on the entire communications process. The road map includes the major elements, but leaves out small (yet critical) components, such as filters and equalizers. These details would simply clutter up the diagram and defeat its purpose. The student should try to locate where any components left out of the road map actually fit into the communications process. The communication system depicted is only good for one way communication and is termed *simplex transmission.* The figure illustrates simplex transmission for purposes of discussion and graphical representation. Nevertheless, most systems, provide for two way transmission. A system is termed *full duplex* if information flows in both directions simultaneously; it is called *half duplex* if information can only flow in one direction at any instant in time (Stremler 90, p. 4). Full and half duplex transmissions are attained by having transmitters and receivers on both sides of the channel.

1. Information

Information must be in the form of an *analog signal* (continuously varying voltage levels, such as voice or video) or *digital signal* (discrete voltage levels, typically the binary digits 0 and 1) before it can be transmitted via an electronic communications system. A *transducer* converts one form of energy into another and is required for voice/video (microphone/camera) to change sound/light waves into an analog electrical signal. Computer generated data are already in digital form and do not need a transducer.

The author has deliberately not defined the term "information" up to this point. Information comes in a myriad of types and forms. To avoid the ambiguity of the term "information," some authors use the terms *analog data* and *digital data.* Another usage is *analog message* and *digital message* or *analog signal* and *digital signal.* Still others refer to "data" only if it was originally digital, such as the output of a computer. This thesis will use the broader term *information* specified as an analog or digital signal. (Carlson 86; Stallings 88; Alisouskas 85)

Information transfer requires varying amounts of bandwidth depending on the allowable errors and desired transfer rate (i.e., throughput rate). *Bandwidth* can be thought of as a measure of how fast a signal can change (Stremler 90; p. 1). A rapidly varying signal requires more bandwidth than does a slower varying signal. The associated bandwidth is critical since the two basic limitations on the performance of a communication system are bandwidth and noise. (Schwartz 90, p. 565; Carlson 86, p. 4) The relationship between pulse width (data rate) and frequency (bandwidth) is explained in detail in the next chapter.

2. Pulse Code Modulation

An analog signal can be transmitted by analog techniques (top path of Figure 3-1 b), or it can be converted into a digital signal and transmitted digitally (bottom path). Analog to digital conversion has several advantages. The major advantage is the error correcting capability of digital communications (Roden 88, p. 5). Also, *repeaters* clean up a noisy signal and reproduce an exact replica of the signal. The process is called *signal regeneration.* The major disadvantage is the increase in bandwidth for the digital signal. Analog signals use amplifiers but these amplify the noise as well as the information signal as shown in Figure 3-2.

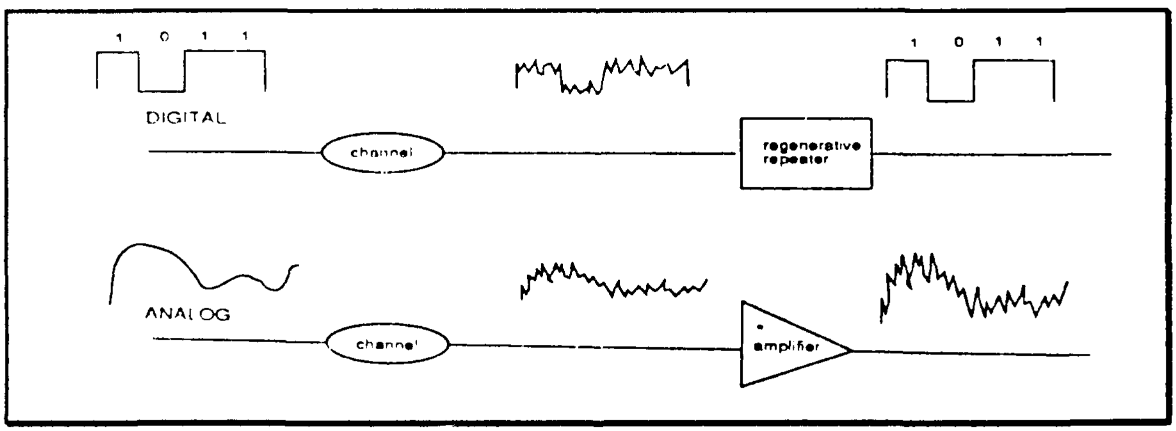

Figure 3-2. Digital Repeater and Analog Amplifier

3. Multiplexers

Analog or digital signals can be sent directly to the transmitter (note the branching paths in Figure 3-1b) or first combined with other signals, in a process called *multiplexing*, and then transmitted over one transmission channel. The multiplexing process makes more efficient use of the channel by using more of the channel's available capacity.

Frequency, time, and code division are three types of multiplexing. They are illustrated in Figure 3-3. Frequency division multiplexing is used with analog signals. Each signal is shifted to a particular frequency slot and occupies that frequency slot continuously (Couch 90, p. 357). Shifting individual signals to different frequencies is itself a form of modulation. However, the signals are still loosely considered to be baseband because they have not yet modulated the final carrier frequency.

Time division multiplexing is the interleaving of sampled data from the different sources. Each signal occupies a discrete time slot while occupying the allocated frequency spectrum. (Couch 90, p. 194)

Code division multiplexing is used with a special kind of digital signal called *direct sequence spread spectrum* (DS-SS). DS-SS is generated when a code at a much higher data rate than the information signal is added to the information signal. In code division multiplexing, each signal occupies all of the frequency band all of the time. The signals do not interfere with one another because each code is different. Only the receiver that has the code can process the spread signal. (Dixon 84, p. 8)

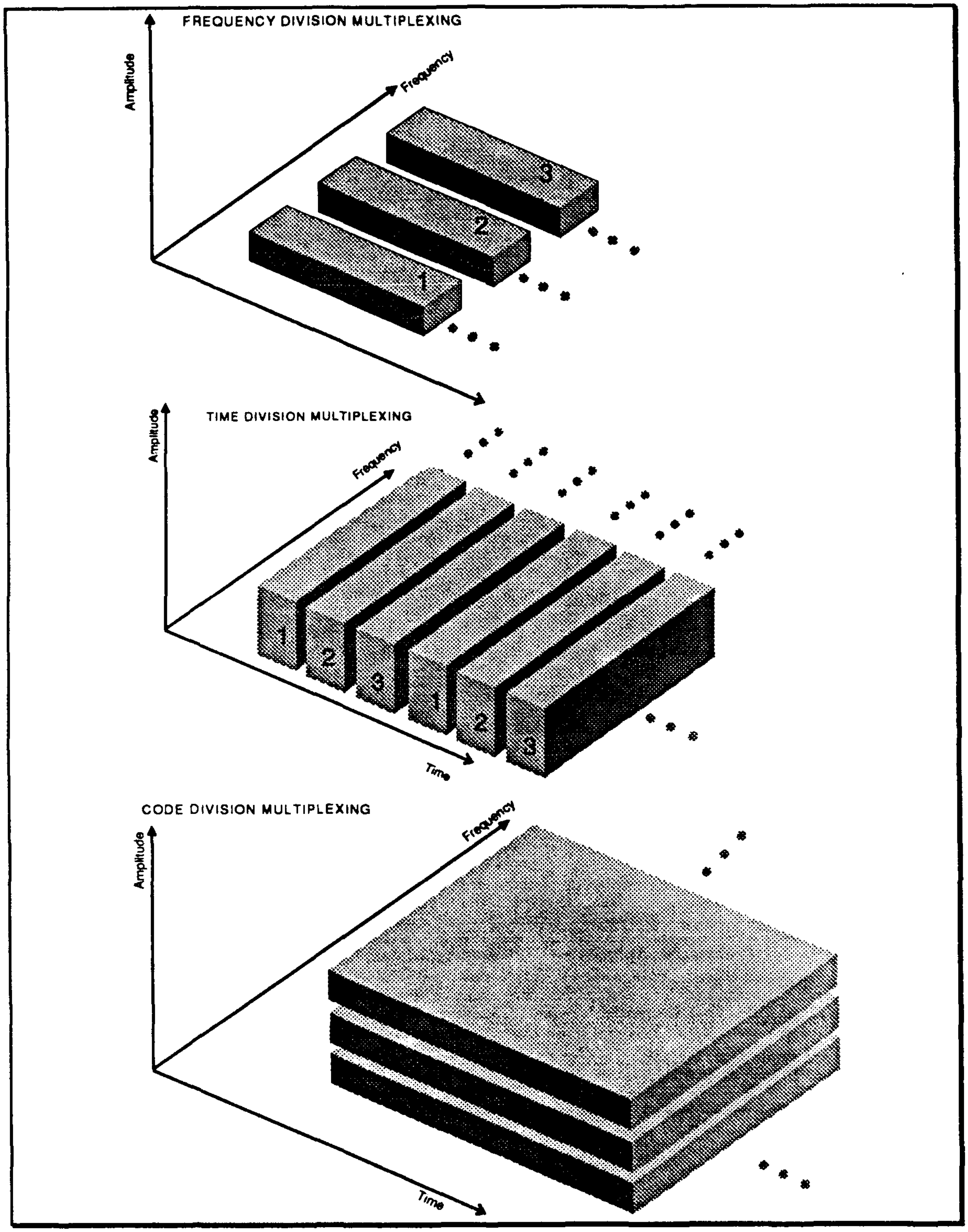

Figure 3-3. Three Types of Multiplexing

4. Modulators

Up to this point signals are considered baseband because they have not yet modulated a carrier frequency. Baseband signals can be transmitted directly over a metallic path for short distances only. The primary purpose of modulation in a communication system is to generate a signal suited to the characteristics of the transmission medium (Carlson 86, p. 7). It makes long-haul communications feasible.

The following analogy may be helpful in understanding the concept of the term carrier. Suppose you and a friend are on opposite sides of a raging river with no means of crossing it. You want to communicate something to him and start yelling your message, but he cannot hear you because there is too much noise emanating from the river. You then write the message on a piece of paper and try throwing it across, but the air does not "propagate" it and it falls at your feet. You now try wadding it up and throwing it again, but it only goes about 10 feet and falls into the water. You look around and see some rocks on the bank. The light bulb finally turns on in your head and you write another message and wrap it around a rock. You now throw it successfully across the river. The rock in a sense "carried" the information to the other side (Lathi 89, p. 13). Also, the rock was large enough to be "propagated" through the air. The other techniques failed because they were overcome by "noise" from an external source or were "attenuated" too much.

Modulation provides for signal/channel optimization. Optimization includes the tradeoffs between authorized frequency allocation, bandwidth, power, transfer rate, antenna size, and cost, among other factors. High frequencies allow for wider bandwidth signals, which means greater data rates can be used. Also, high frequencies allow reasonably sized antennas to be used. The antenna is actually part of the transmitter subsystem, but is shown in Figure 3-1b under the channel subsystem due to drawing constraints.

5. Antennas

Antennas act as an interface between the electronic circuit and the propagating medium. The size of the antenna is directly related to the wavelength of the signal it is designed to transmit or receive. An antenna's dimension should be a minimum of one tenth of the signal's wavelength for efficient transmission. For example, a carrier signal at 30 kHz has a wavelength of 10 kilometers. This means the antenna would have to be a minimum of 1 kilometer long. A 30 MHz signal has a wavelength of 10 meters which requires a minimum antenna size of 1 meter. In practice, antennas are designed for a quarter to a half of the signal's wavelength. Therefore, higher-carrier frequencies allow for smaller antennas. (Carlson 86, p. 7; Stanley 82, p. 511)

6. Propagation

The following description of propagation gives one of the clearest explanations found by the author and is included in its entirety (Smith 89):

> When energy is emitted into space, whether it is heat, sound, or light, it travels away from the source at a speed that is governed by the medium in which it is travelling. This travelling of energy is called propagation. In the case of radio waves this propagating energy exists in the form of fields. There are two types of fields which carry this energy, the electric or E field, and the magnetic or H field. The radio energy is carried partly in the electric field and partly in the magnetic field, hence these propagating waves are called electromagnetic.
>
> While these fields cannot be seen, we have seen the effects of them. Electric fields exist between the clouds and the earth and are the cause of lightning. E fields cause electrons to move, which is called current flow. H fields create magnetism and therefore attract magnetizable materials such as iron. This is the principle used in stereo speakers. The coil sets up magnetic fields, because of the current flowing through it, and causes a metal bar attached to the fabric

of the speaker to vibrate, creating sound waves. An interesting fact about these electromagnetic fields is that if one of them exists and it is changing, it creates the other. That is, a changing electric field creates a magnetic field, and a changing magnetic field produces an electric field. This phenomenon **allows the propagating wave to sustain itself** (emphasis author's) as it travels through its transmission medium. As the fields travel through space, they build and collapse at right angles to each other, and at right angles to direction of propagation. See Figure 3-4 (copied from Miller 88, p. 463).

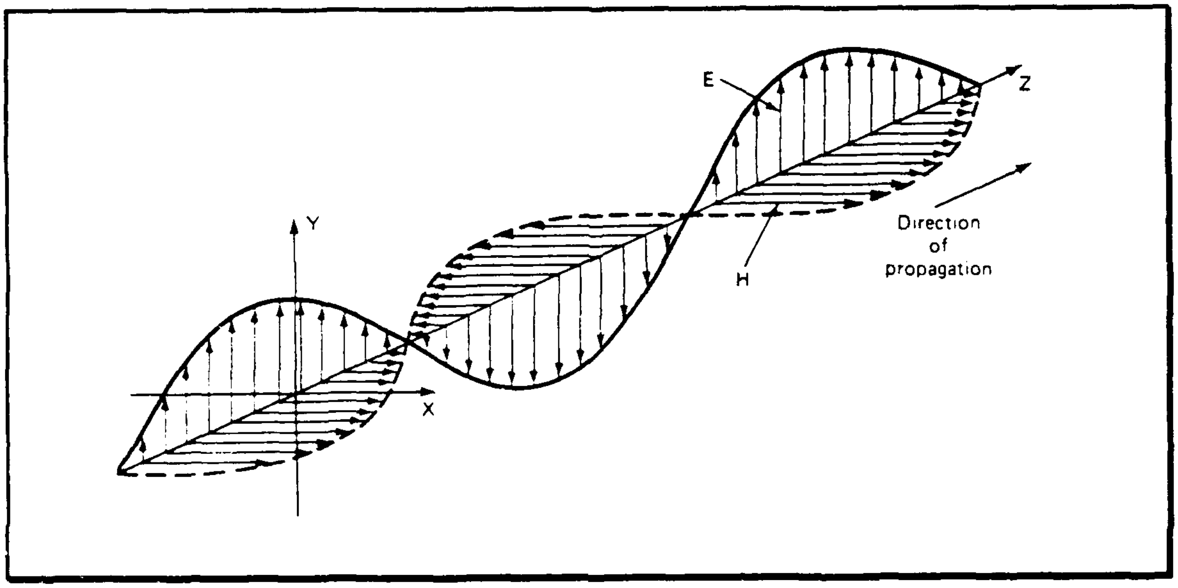

Figure 3-4. Electromagnetic Wave Propagation

7. Channel

Getting information from "here to there" requires some type of path or medium. The three main transmission mediums are free space (including the troposphere, ionosphere, outer space), metallic path (copper: twisted pair and coax) and fiber optics. Waveguides or coax between the modulator and antenna can be thought of as a medium and are shown as heavy black lines in Figure 3-1b. The three basic mediums are explained in detail in Chapter VII.

The chart directly below the medium characteristics in Figure 3-1c shows the relative attenuation (decibels per kilometer) of the various mediums as a function of frequency. The lower the attenuation the better. The chart shows that different mediums pass a range of frequencies more easily than others which is the reason why certain operating frequencies are chosen. A larger chart showing the attenuation is explained in more detail in Chapter VII. There are also other factors associated with mediums, such as refraction, distortion, dispersion, and noise. These factors are also discussed in Chapter VII.

C. FUNCTIONAL BREAKOUT

The key points of this thesis are given on the bottom line of Figure 3-1c. The information signal, digitization, and multiplexing all have a bandwidth associated with them and all are limited by the transmission medium. The modulator carries the signal at a frequency compatible with the medium. The line from the modulator to frequency bands symbolizes an arbitrarily chosen carrier frequency.

IV. INFORMATION AND BANDWIDTH

A. OVERVIEW

The bandwidth of the information is critical since the two basic limitations on the performance of a communication system are the noise and the bandwidth-limiting characteristics of the channel (Schwarz 90, p. 565). The channel limits the rate at which information can be sent. Therefore, it is important to understand the inversely proportional relationship between pulse width and bandwidth. Decreasing the pulse width is often desired to allow more data to be passed in a given period of time. However, decreasing pulse width requires the bandwidth be increased.

Wider bandwidths are subject to limitations imposed by the channel, including distortion, dispersion, and attenuation. There are also legal constraints, such as FCC and international frequency assignments. Channel limitations are discussed in Chapter VII.

A methodology is needed for characterizing a signal to determine its frequency components (bandwidth). Fourier analysis is the basic mathematical tool that permits conversion from the *time domain* (pulse width) to the *frequency domain* (bandwidth). Fourier analysis gives the insight for understanding the frequency spectrum. It is assumed the reader has already taken a mathematics course in Fourier analysis. Fourier series and Fourier transforms are reviewed briefly to show the applicability to communication systems.

B. BANDWIDTH-FREQUENCY CONTENT

Figure 4-1 shows the interrelationship of the concepts and tools required for developing the frequency spectrum. Note that the boxes either contain a light bulb or a screw driver. The light bulb represents a fundamental idea/concept that needs to be understood before related subjects make sense. The screwdriver represents mathematical tools that provide a working foundation for the concepts.

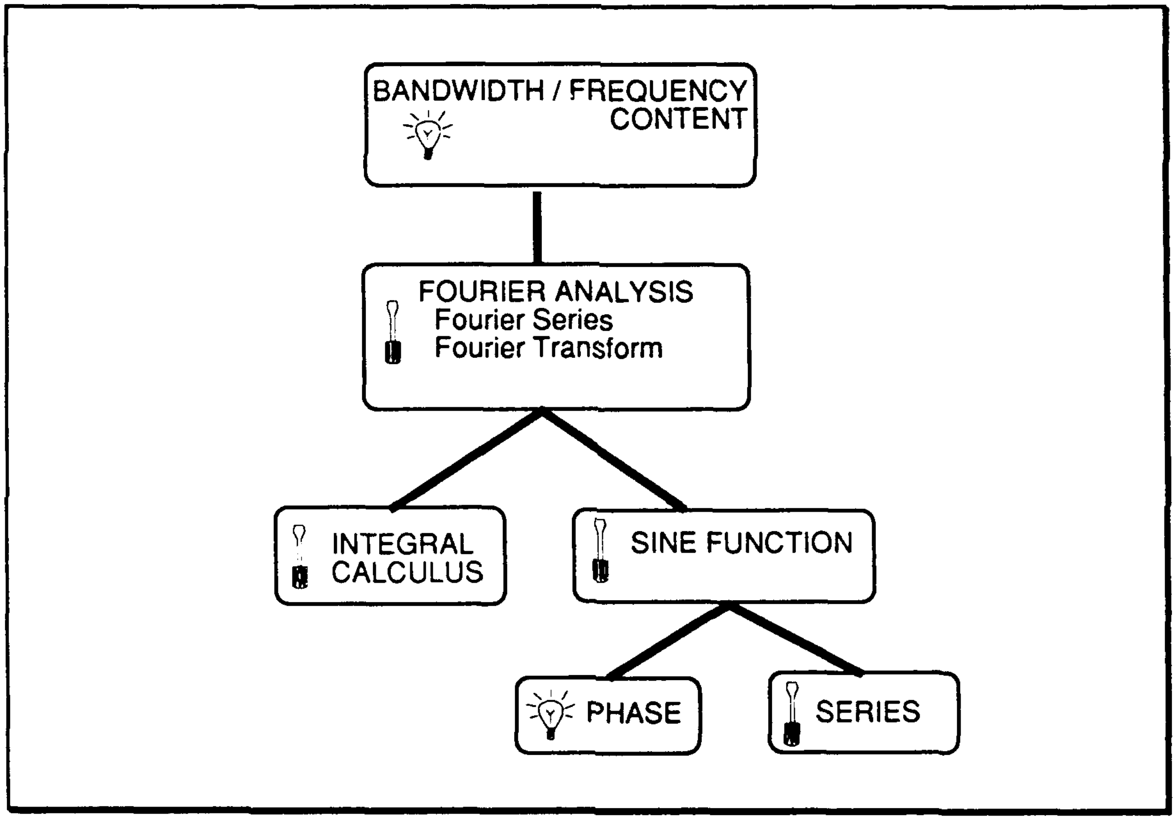

Figure 4-1. Frequency Spectrum Building Blocks

A hierarchical relationship is shown in Figure 4-1 to provide a sense of applicability of the needed mathematics. Questions often posed by a student are "Why are we learning this? Who uses it? Where is it applied?" The intent is to shed light on these questions.

1. Frequency Domain

The time domain is usually easier to comprehend than the frequency domain because we experience events in a time continuum. However, in communications the frequency domain provides a more convenient means of representing signals. Therefore, the chapter focuses on the relationship between the time domain and frequency domain.

Figure 4-2 shows a three-dimensional depiction of the relationships between time, frequency, and amplitude. Figures 4-2 (a) and (b) show typical time domain plots with periods every T_1 and T_2 seconds, respectively. Figure 4-2 (c) is a three-dimensional illustration encompassing both time and frequency.[1] Viewing Figure 4-2 (c) from the amplitude and time plane, one sees the time domain depicted in (a) and (b). By rotating the perspective so one now looks at the signal "head on" down the time axis (i.e., looking at the amplitude and frequency plane) one sees the frequency domain as depicted in (d). Note how frequency is inversely related to the period. The reader should do this several times to become comfortable with the time and frequency domains.

The time and frequency domains are realizable with physical devices. The time domain can be seen by using an oscilloscope, whereas a spectrum analyzer is used for the frequency domain.

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[1] This figure is for illustration purposes only and is not a plot of a function of two variables. The box around the cosine wave is for helping visualize the 3-D effect.

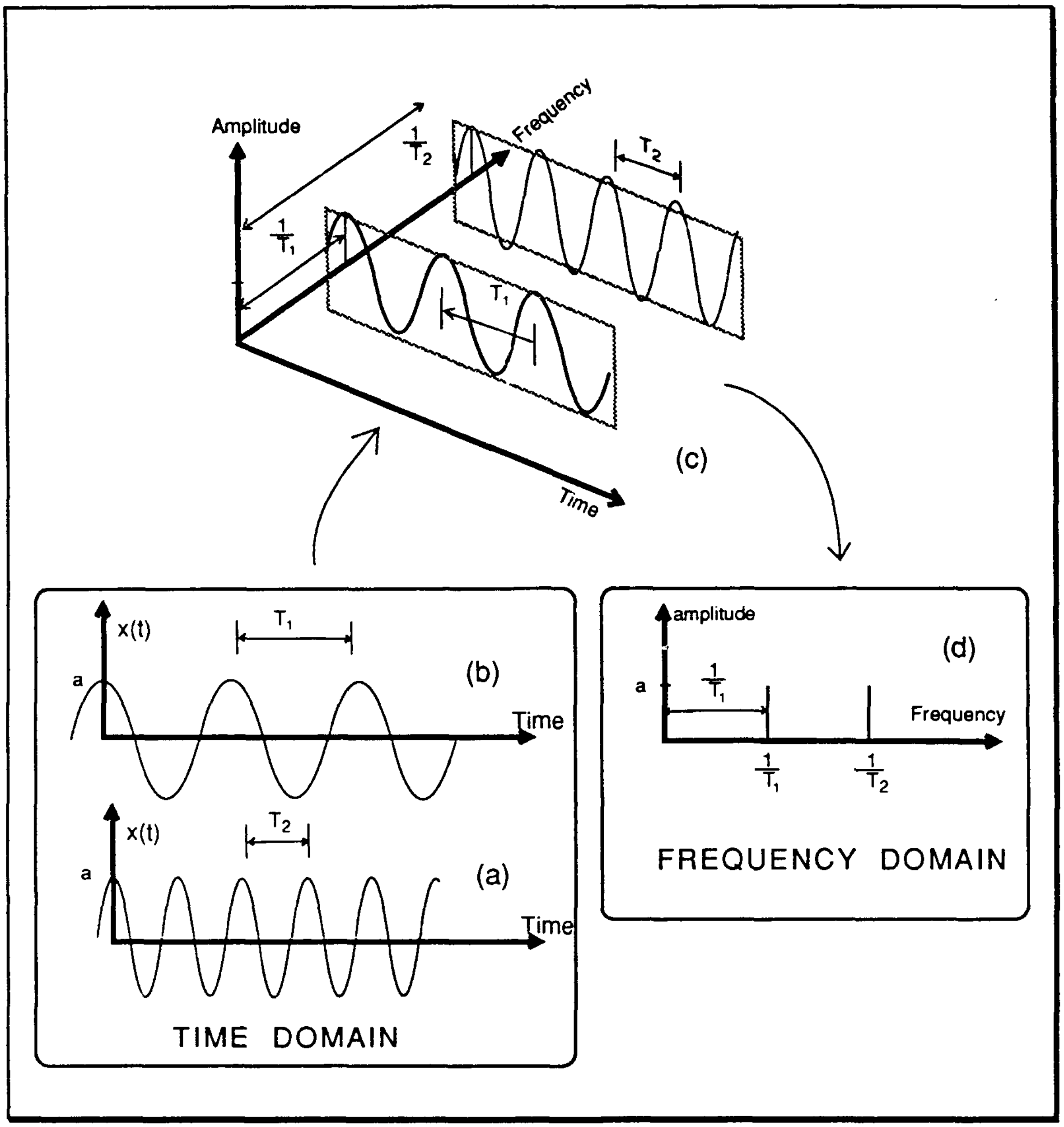


Figure 4-2. Relationship Between Time, Frequency and Amplitude

2. Fourier Series

The spectrum (or frequency content) of a periodic signal is obtainable mathematically via the Fourier series. Recall that the Fourier series is able to represent a periodic function[2] or waveform/signal as an infinite sum of sinusoids of varying amplitude, frequency and phase. The terms function and waveform are used interchangably throughout the discussion. (Stremler 78)

A function is said to be *periodic* if it repeats itself in a finite interval. Examples of periodic waves are square waves and the sinusoidal waves. The minimum distance (or time interval) for which it repeats itself is called the *period* and is usually denoted by 2L or T. Since communcation signals are the focus of this chapter, the time period T is used. Mathematically we write $x(t) = x(t+T)$. The reciprocal of the period (1/T) is the *fundamental frequency*[3] (f_1) and is expressed in hertz (cycles per second). Thus the fundamental frequency is the number of times the signal repeats itself in one second. (Brigham 74; Giordano 91; Stremler 78)

The Fourier trigonometric series is expressed as:

$$x(t) = a_0 + \sum_{n=1}^{\infty} a_n \cos n\omega_1 t + \sum_{n=1}^{\infty} b_n \sin n\omega_1 t \qquad (4\text{-}1)$$

where x(t) is the periodic function and $\omega_1 = 2\pi/T$ (radian frequency). The terms a_o, a_n, and b_n are called the *Fourier coefficients* and are determined by the waveform to

---

[2] Providing it meets the four *Dirichlet* conditions. The function must (1) be single valued (2) have a finite number of discontinuities (3) have a finite number of maxima and minima (4) be absolutely integratable (Haykin 78, p. 15).

[3] The fundamental frequency is sometimes denoted by f_o in textbooks, whereas others use f_1. The subscript 1 is used in this thesis so there is consistency between the terminology for the fundamental frequency and the first harmonic. Also, the subscript "o" is avoided to prevent confusion with zero frequency.

be represented. The frequencies of the sinusoids in the series are all integer multiples of the fundamental frequency of the waveform x(t). The integers, represented by n, are called *harmonic numbers* and the corresponding terms in the series are called *harmonics*. The harmonics give the frequencies that compose the waveform (frequency content). The first harmonic, where n = 1, represents the fundamental frequency. The Fourier coefficients a_n and b_n determine the amplitudes of the harmonics. (Stremler 78)

The Fourier coefficients are defined as follows:

General case:

$$a_0 = \frac{1}{T}\int_0^T x(t)dt$$

$$a_n = \frac{2}{T}\int_0^T x(t)\cos n\omega_1 t\, dt$$

$$b_n = \frac{2}{T}\int_0^T x(t)\sin n\omega_1 t\, dt$$

Even function:

$$a_0 = \frac{2}{T}\int_0^{T/2} x(t)dt$$

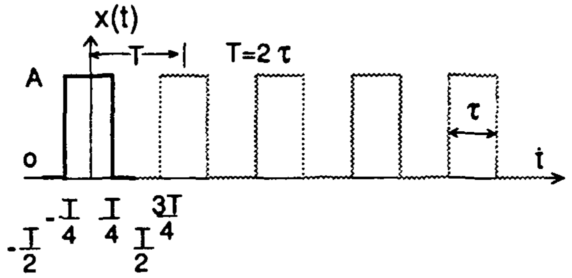
$$a_n = \frac{4}{T}\int_0^{T/2} x(t)\cos n\omega_1 t\, dt$$

$$b_n = 0$$

The coefficient a_0 is the *average* value of the function x(t) over the period $0 \le t \le T$. In waveform/signals terminology it is the *dc (direct current) voltage*. The dc term is also called the *dc bias*, and it raises or lowers the entire waveform above or below the time axis. The dc component does not change with respect to time (no period). Therefore, it is said to have no frequency or *zero frequency*. (Stanley 82, p. 25)

The waveforms chosen for all the examples to follow are very simple and similar to one another. Only one parameter is changed at a time to emphasize the similarities between the chosen waveforms. The first Fourier series example is the square wave where $T = 2\tau$ is the period. Here τ is the pulse width. The waveform is described by the even function:

$$x(t) = \begin{cases} A, & 0 < t < \dfrac{T}{4} \\ 0, & \dfrac{T}{4} < t < \dfrac{3T}{4} \\ A, & \dfrac{3T}{4} < t < T \end{cases}$$



Solving for the Fourier coefficients we have:

$$a_0 = \frac{2}{T}\int_0^{\frac{T}{4}} A\,dt + \frac{2}{T}\int_{T/4}^{T/2} 0\,dt = \frac{2}{T}At\Big|_0^{\frac{T}{4}} = \frac{A}{2}$$

$$b_n = 0$$

$$a_n = \frac{4}{T}\int_0^{\frac{T}{4}} A\cos n\omega_1 t\,dt + \frac{4}{T}\int_{T/4}^{T/2} 0\cos n\omega_1 t\,dt$$

$$= \frac{4}{T}\frac{A}{n\omega_1}\sin n\omega_1 t\Big|_0^{T/4} + 0 = \frac{4}{Tn\omega_1}\sin n\omega_1\frac{T}{4} - \frac{4A}{\tau n\omega_1}\sin 0$$

Substitute $2\pi/T$ for ω_1 to simplify the equation, and a_n becomes

$$a_n = \frac{4A}{Tn\dfrac{2\pi}{T}}\sin n\frac{2\pi}{T}\frac{T}{4} = \frac{2A}{n\pi}\sin\frac{n\pi}{2}$$

Substituting a_0 and a_n into the general equation (4-1) gives the following Fourier expansion for the squarewave form x(t):

$$x(t) = \frac{A}{2} + \sum_{n=1}^{\infty}\frac{2A}{n\pi}\sin\frac{n\pi}{2}\cos n\omega_1 t$$

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Approximating x(t) by the first 6 harmonics, n = 1 to 6 (all even harmonics are zero) gives:

$$x(t) \approx \frac{A}{2} + \frac{2A}{\pi}\left(1\cos\omega_1 t - \frac{1}{3}\cos 3\omega_1 t + \frac{1}{5}\cos 5\omega_1 t\right)$$

Figure 4-4 depicts the first and third harmonics added together to approximate the square wave. The average value a_0 and the fifth harmonic are added later in another Figure 4-5.

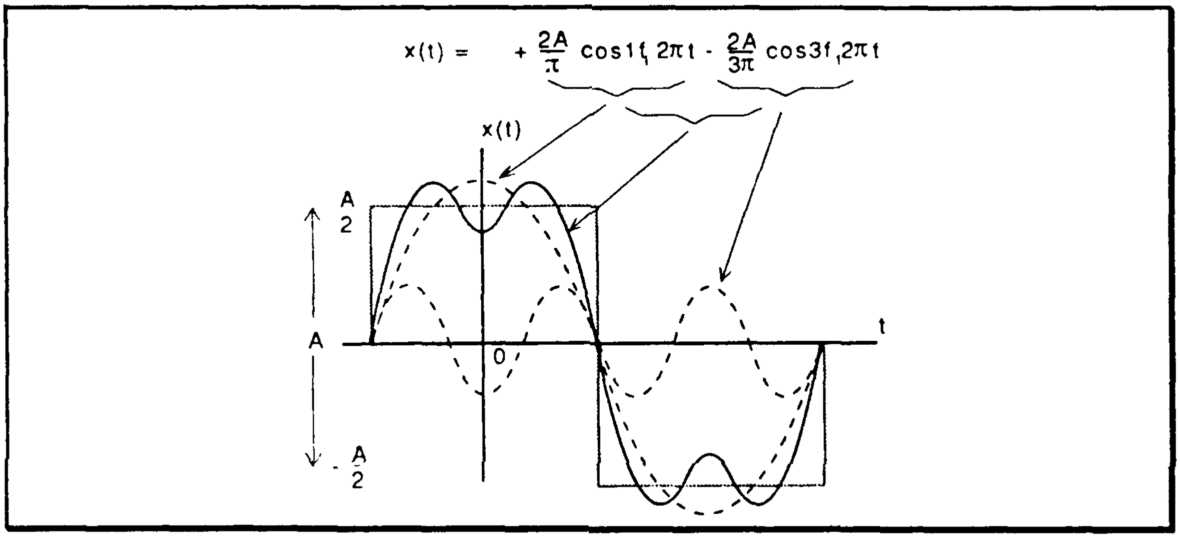

Figure 4-3. First and Third Harmonic Summed

The *discrete spectrum*[4] is obtained by plotting each Fourier coefficient amplitude (a_n) versus its harmonic frequency. Generally, b_n is also included. However, the above example uses only even functions and b_n is 0. If b_n had not been zero, then the *magnitude* of the sine and cosine coefficients, $M_n = \sqrt{a_n^2 + b_n^2}$ is plotted. Also, the absolute value of the coefficients is displayed rather than the magnitude M_n to remind the reader that there are negative amplitude coefficients

[4] Discrete since only integer values of n are possible.

(phase shift of π, see the third harmonic of Figure 4-3). Similar terminolgy is used later in the chapter and positive spectrums are similar to spectrum analyzer displays.

The discrete spectrum amplitude axis is usually labeled X_n, C_n, c_n, or M_n depending on the author's development.[5] Nevertheless, the reader should be aware that different notations do exist. (Haykin 78, p. 9; Stanley 82, p. 27)

Figure 4-4 depicts a_o (dc component) and the first six harmonics for the frequency domain plot. The fundamental frequency notation f_1 is used to maintain the more familiar use of the term frequency. The radian frequency ω_1 could have been used as well. The light gray line is the envelope of the absolute value of the sinx/x function which will be discussed later.

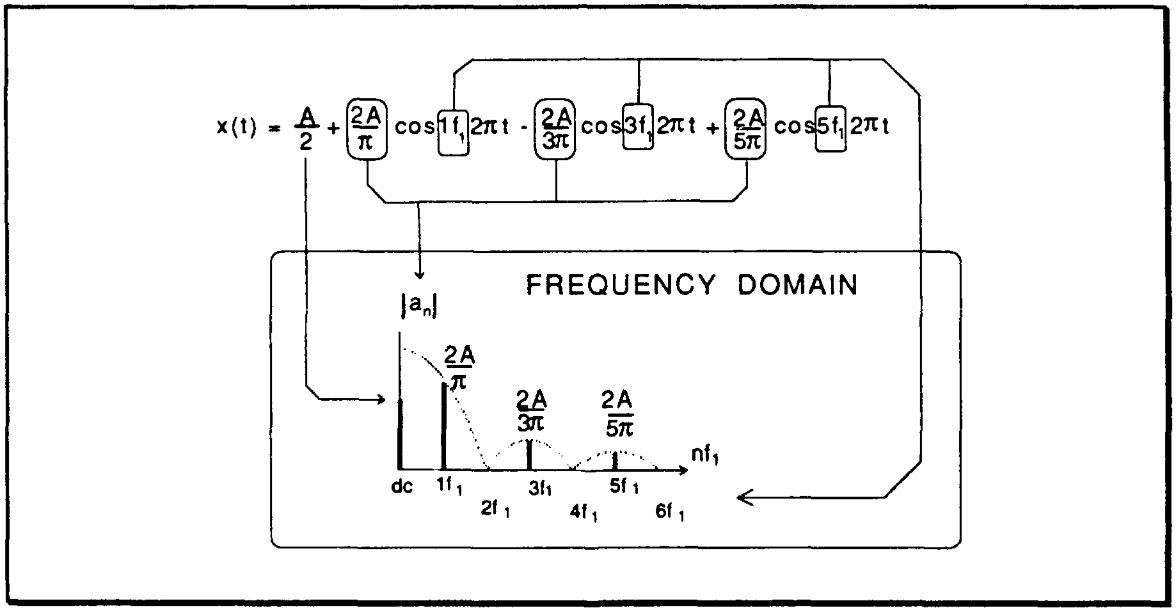

Figure 4-4. Discrete Frequency Spectrum

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[5] A magnitude spectrum and phase spectrum exists for an amplitude phase form of the Fourier series. Also, there is a Fourier series form using complex exponentials, giving a two-sided vice one-sided spectrum. These other forms are not needed since all the b_n coefficients in the examples given are zero.

A technique for illustrating the summation of sinusoids is developed in Figure 4-5. Only the harmonic number is given below each spectral line to keep the amplitude spectrum diagram relatively clutter free. The first column of boxes shows the time domain of each individual harmonic. The middle column shows the waveforms to be summed (but are not yet summed). This view is provided to show the relationship between the harmonics. The far right column is the summation of the harmonics from the middle box. For example, the top two boxes of the first column are to be summed (first and third harmonic). They are shown together in the top box of the middle row. The summation is shown in the top right-hand box. Next, the fifth harmonic is to be added to the summation of the first and third harmonic. This is shown in the middle box of the middle column. The sum of these two are shown to the immediate right. Finally, the dc component is added, which effectively raises the sum of the first, third, and fifth harmonics. The dc component is added last allowing easier visualization of the summation process. Notice how the final plot, in the last box of the right-hand column of the figure, begins to resemble the square wave x(t) in the example.

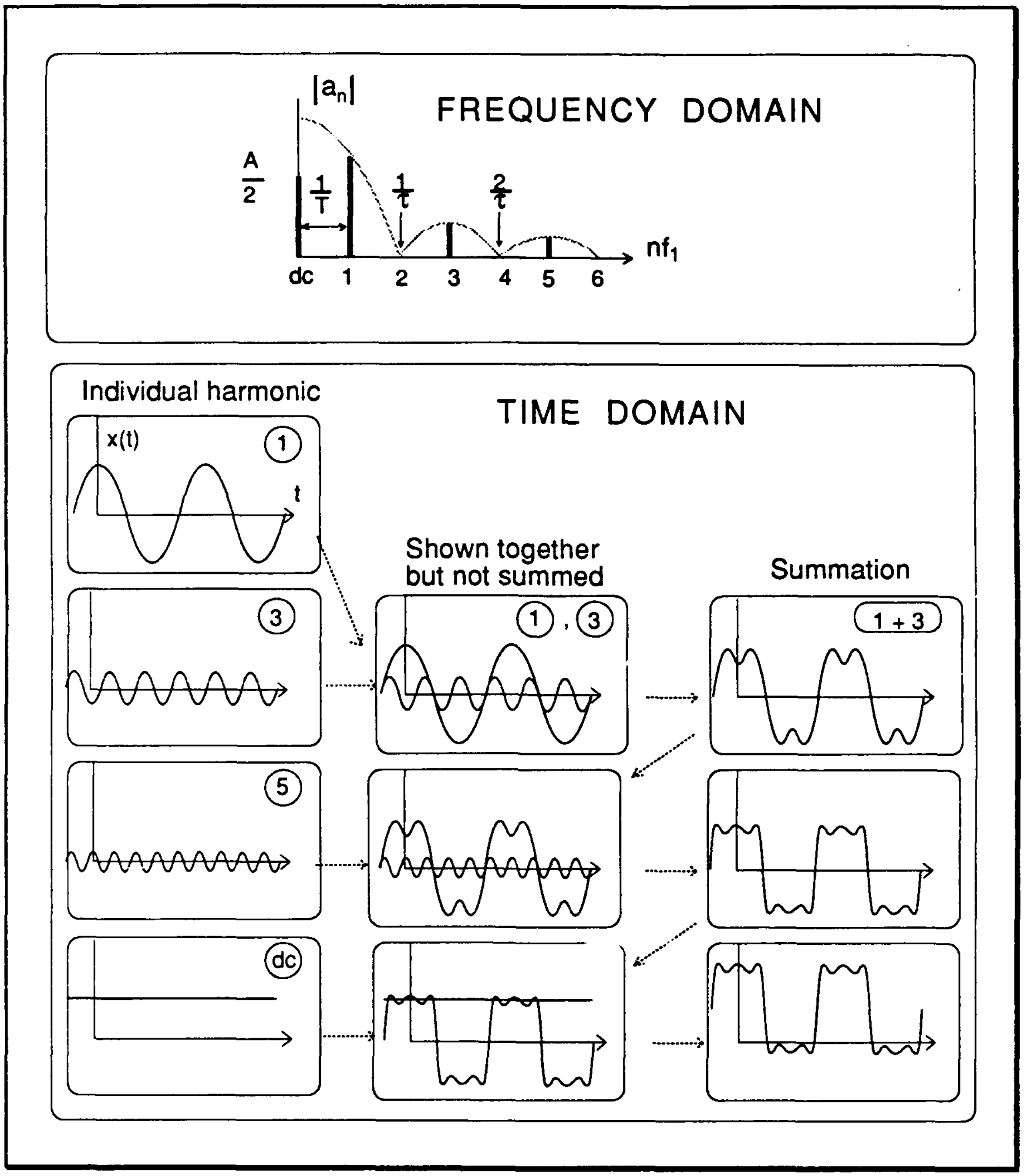
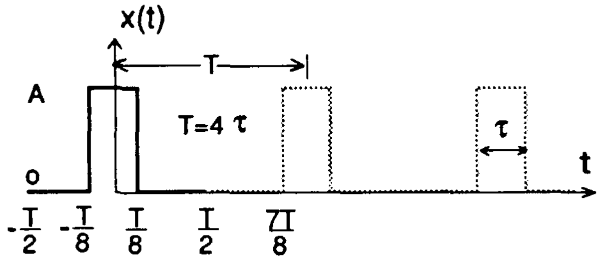


Figure 4-5. Methodical Technique for Summing Harmonics

The presentation continues with "digital-type signals," that is, signals with only two levels (on/off or 0/1), because they are the easiest to manipulate. Moreover the resulting spectra are very common in communications.

The next example keeps the pulse width the same, but doubles the period for our preceding example: $T = 4\tau$. Thus, the waveform is now defined by the function:

$$x(t) = \begin{cases} A, & 0 < t < \dfrac{T}{8} \\ 0, & \dfrac{T}{8} < t < \dfrac{7T}{8} \\ A, & \dfrac{7T}{8} < t < T \end{cases}$$



Now calculate for the Fourier coefficients for x(t):

$$a_0 = \frac{2}{T}\int_0^{\frac{T}{8}} A\,dt + \frac{2}{T}\int_{T/8}^{T/2} 0\,dt = \frac{2}{T}At\Big|_0^{\frac{T}{8}} = \frac{A}{4}$$

$$b_n = 0$$

$$a_n = \frac{4}{T}\int_0^{\frac{T}{8}} A\cos n\omega_1 t + \frac{4}{T}\int_{T/8}^{T/2} 0\cos n\omega_1 t$$

$$= \frac{4}{T}\frac{A}{n\omega_1}\sin n\omega_1 t\Big|_0^{T/8} = \frac{4A}{Tn\omega_1}\sin n\omega_1\frac{T}{8} - 0$$

Substitute $2\pi/T$ for ω_1 to simplify the expression for the a_n coefficients:

$$a_n = \frac{4A}{Tn\frac{2\pi}{T}}\sin n\frac{2\pi}{T}\frac{T}{8} = \frac{2A}{n\pi}\sin\frac{n\pi}{4}.$$

Next substitute a_o and a_n into the general equation (4-1) for the new waveform x(t):

$$x(t) = \frac{A}{4} + \sum_{n=1}^{\infty} \frac{2A}{n\pi} \sin\frac{n\pi}{4} \cos n\omega_1 t$$

Selecting the first 12 harmonics to approximate x(t), (n =1 to 12) noting that every other even harmonic (n = 4, 8, 12) is zero:

$$x(t) \approx \frac{A}{4} + \frac{2A}{\pi}\begin{pmatrix} + \frac{.707}{1}\cos\omega_1 t & + & \frac{1}{2}\cos 2\omega_1 t & + & \frac{.707}{3}\cos 3\omega_1 t & + & \frac{0}{4}\cos 4\omega_1 t \\ - \frac{.707}{5}\cos 5\omega_1 t & - & \frac{1}{6}\cos 6\omega_1 t & - & \frac{.707}{7}\cos 7\omega_1 t & - & \frac{0}{8}\cos 8\omega_1 t \\ + \frac{.707}{9}\cos 9\omega_1 t & + & \frac{1}{10}\cos 10\omega_1 t & + & \frac{.707}{11}\cos 11\omega_1 t & + & \frac{0}{12}\cos 12\omega_1 t \end{pmatrix}$$

The summation of the first 12 harmonics is shown in Figure 4-6 using the same methodology presented earlier (Figure 4-5). This figure includes the actual time domain waveform at the top of the right-hand column for purposes of comparison against the summed waveform at the bottom of that column. Notice also that we added the dc component at the beginning of forming the sum of the first 12 harmonics, rather than at the end (as in Figure 4-5). The dc component can be added at any stage of the summation process.

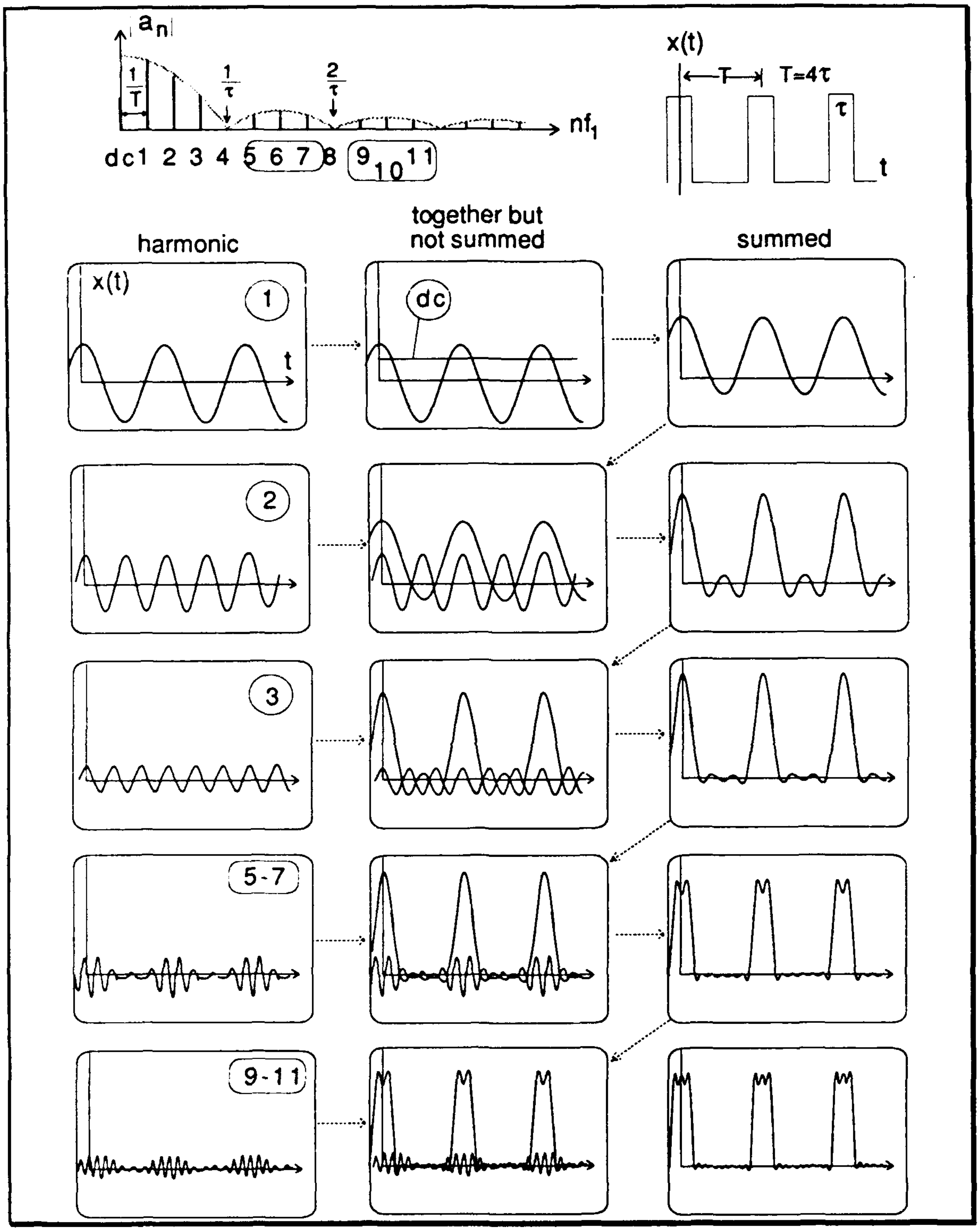


Figure 4-6. Summation of Harmonics

For purposes of comparison, we double the period one more time[6]. The author will forgo the Fourier analysis since it is similar to the previous two examples, but the results are presented at the bottom of Figure 4-7. Note that the spacing between the harmonics in the frequency domain is inversely proportional to the period. Also, although the amplitude of the harmonics decreases with an increasing period, the envelope of the discrete frequency spectrum is similar in each case. Furthermore, the $1/\tau$ frequency remains exactly the same because the same pulse width is used for all three cases.

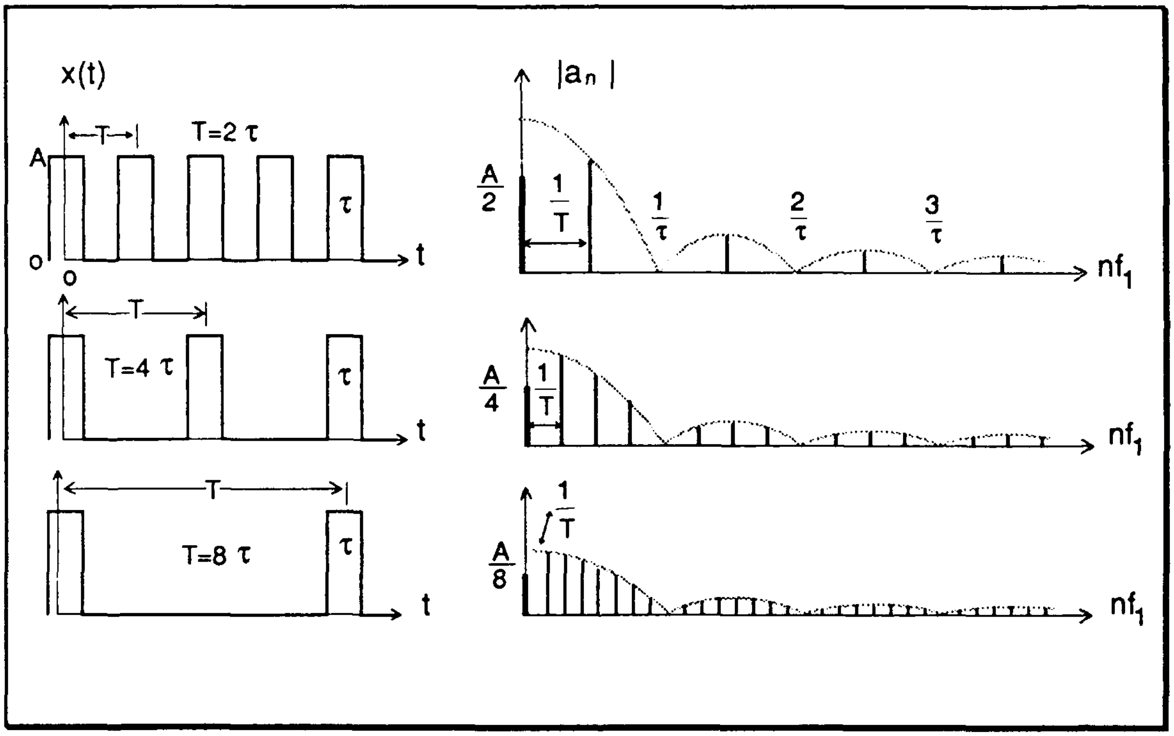


Figure 4-7. Pulse Width Constant, and Period Increased

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6 The period does not have to be an even mutiple of the pulse width. The doubling of the periods are chosen for illustration purposes. It makes the frequency components line up nicely and makes the n^{th} harmonic associated with the period to pulse width ratio coincide with the $1/\tau$ point.

3. Fourier Transform

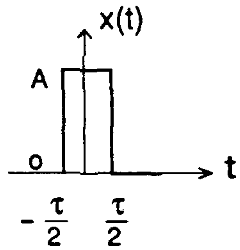
The preceding Fourier examples had a finite period T where $T = n\tau$. Now consider the waveform as n is increased to infinity. Then there is no period and the signal is *nonperiodic*. A random signal would also be considered a nonperiodic signal. Communications signals are considered random in the sense that it is not known what precise sequence of bits is going to be received. Therefore, the Fourier series is not applicable in the nonperiodic case.

The *Fourier transform* is used for nonperiodic signals. It employs a complex-valued exponential function[7] to transform the time domain to the frequency domain according to the following equation:

$$X(f) = \int_{-\infty}^{\infty} x(t)\, e^{-j\omega t}\, dt \qquad (4\text{-}2)$$

Let us consider an example of a single pulse defined by:

$$x(t) = \begin{cases} A, & -\frac{\tau}{2} < t < \frac{\tau}{2} \\ 0, & \text{elsewhere} \end{cases}$$



Determining the Fourier transform we have:

$$X(f) = \int_{-\tau/2}^{\tau/2} A e^{-j\omega t} dt = \frac{A}{-j\omega}\left[e^{-j\omega t} \Big|_{-\tau/2}^{\tau/2} \right] = \frac{A}{-j\omega}\left[e^{-j\omega\tau/2} - e^{+j\omega\tau/2} \right]$$

[7] The complex exponential actually makes the analysis easier. See the texts by Stremler and Lathi for excellent discussions on the use of complex notation (Stremler 90, p. 23; Lathi 89, p. 28).

$$X(f) = \frac{A}{\omega} \bullet 2\left[\frac{e^{j\omega \tau/2} - e^{-j\omega \tau/2}}{2j}\right]$$

Note the expression in brackets resembles the complex form of the sine:

$$\sin\theta = \left[\frac{e^{j\theta} - e^{-j\theta}}{2j}\right]$$

Substituting this form gives:

$$X(f) = \frac{2A}{\omega}\sin\left(\omega\frac{\tau}{2}\right)$$

Now substitute $2\pi f$ for ω and rewrite the transform as:

$$X(f) = \frac{\tau}{\tau}\frac{2A}{2\pi f}\sin\left(2\pi f\,\frac{\tau}{2}\right) = A\tau\frac{\sin\pi f\tau}{\pi f\tau}$$

Note that the expression on the right-hand side of the last equation is a function of frequency (continuous vice discrete). The Fourier transform results in a *two-sided frequency spectrum* (the frequencies can assume negative, zero, and positive values). The spectra given in the sine/cosine series representations were *one sided* (the average value and only positive integer values of n). There is a complex form of the Fourier series which reveals a two-sided discrete spectrum. The amplitude of the two-sided spectrum is exactly half of the amplitude of the single-sided spectrum[8]. The average value, or dc component, stays the same. The curious

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[8] Briefly the one sided spectrum is composed of real componets, where as the two sided spectrum inolves frequencies in the complex coordinate system and the *complex conjugate* is required in the complex expontial form to get back to the real axis. Full development of the complex notaion is beyond the scope of this theis. The books by Stremler and Lathi do an excellent job describing the concepts. The important point is becoming familiar with the general shape of the frequency spectrum.

reader should read the texts by Stremler and Lathi for details (Stremler 90, p. 23; Lathi 89, p. 28).

The expression for X(f) derived above is also of the form *sinx/x* where $x = \pi f \tau$. The sinx/x is termed the *sinc* function and occurs frequently in communications and radar systems (Haykin 78, p. 12). The sinx/x is used interchangably with the derived function X(f) for ease of discussion.

Figure 4-8 shows a plot of sinx/x. Note that when the frequency equals $1/\tau$, the $\sin(\pi f \tau)$ term is simply $\sin(\pi)$ or zero. This zero point is also known as the *first zero crossing* (*FZC*). When the absolute value is plotted, the FZC is called a *null.* The amplitude at zero frequency is $A\tau$ since the limit of sinx/x is one as the frequency approaches zero.

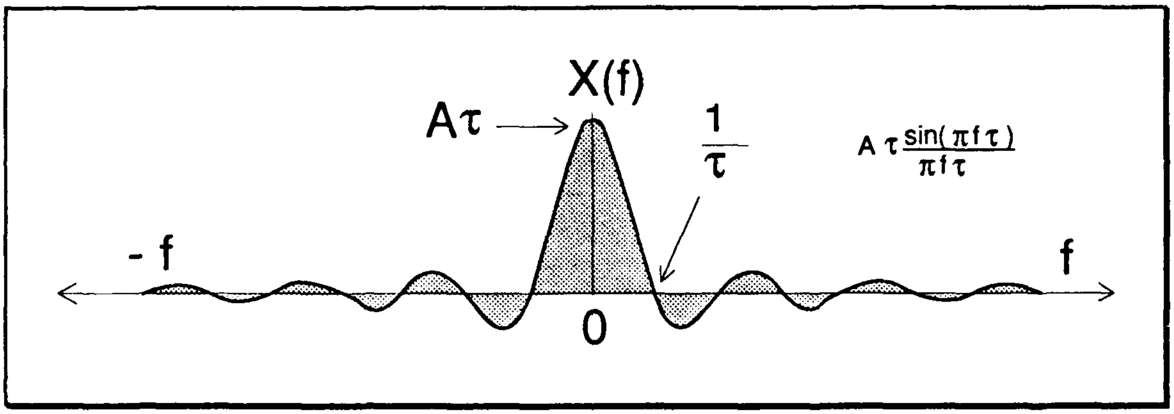

Figure 4-8. Typical Sinx/x Plot

The reader may come across various forms of the sinx/x plots without recognizing they are really looking at different representations of the same function. The author includes the absolute value and the logaritmic amplitude (vertical axis) of sinx/x in Figure 4-9 to illustrate some of the variations possible. Additionally, three different values of τ are used to show the inverse relationship between the pulse width and the first zero crossing. In effect, decreasing the pulse width (so a higher data rate is possible) increases or spreads out the the frequency spectrum (wider bandwith).

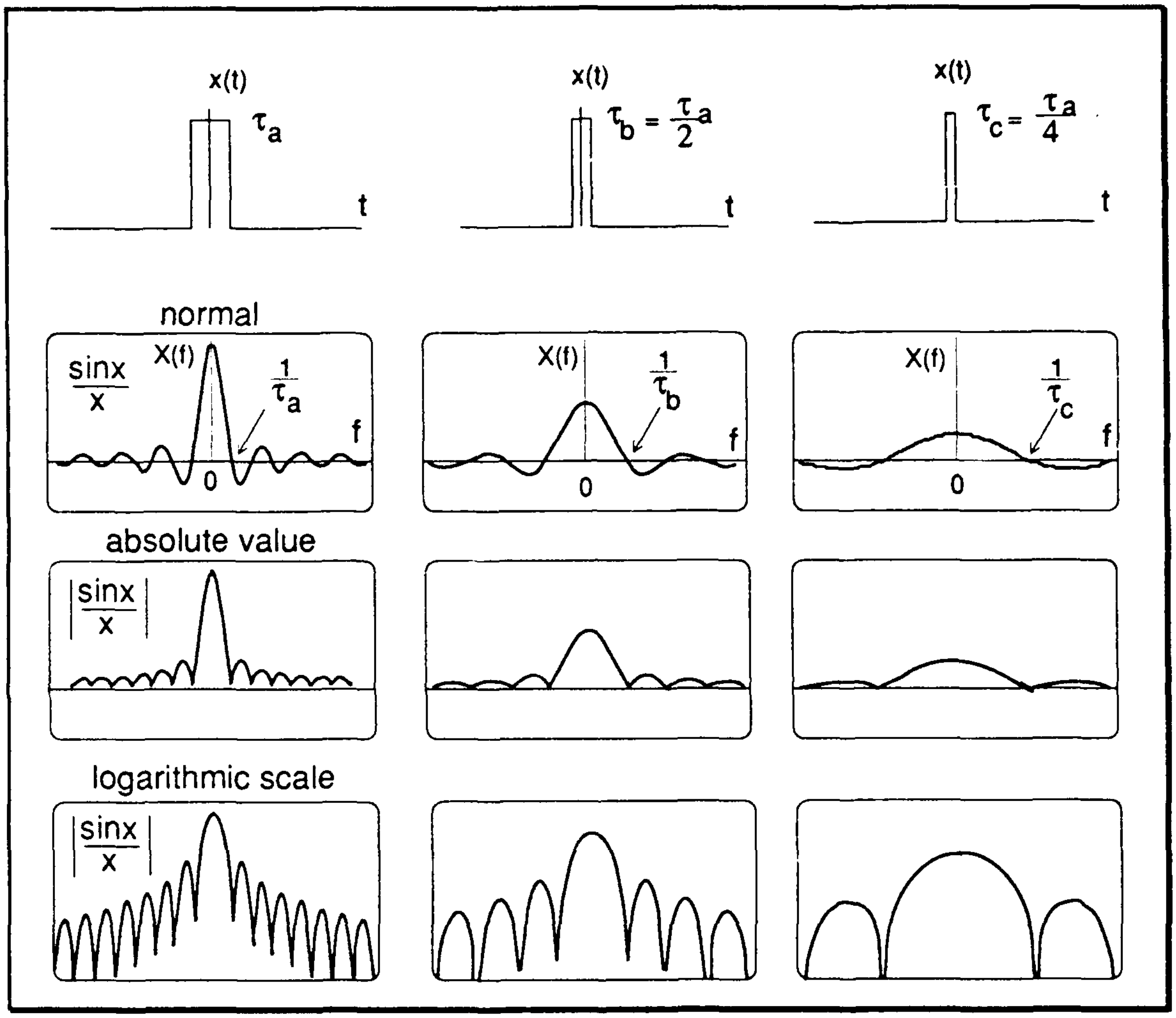

Figure 4-9. Various Sinx/x Plots

The author now ties together the main concepts into a single chart for both periodic and nonperiodic waveforms (see Figure 4-10). The amplitudes and harmonic numbers are removed to reduce chart clutter. Also, only the absolute values of the one-sided spectrum are shown in order to conserve space.

Furthermore, the spectrums are similar to baseband signals displayed on spectrum analyzers (see Chapter VI).

Figure 4-10 reveals the effects of varying the pulse width τ or the period T. Plots (a) thru (c) illustrate that spacing between the harmonics in the frequency domain is inversely proportional to the period for fixed pulse width. Also, the envelopes of the discrete frequency spectrum are similar. The $1/\tau$ point remains in exactly the same position because the pulse width is held constant.

Plot (d) is nonperiodic but the pulse width is the same as for (a), (b) and (c). Note that the frequency spectrum is continuous rather than discrete, and the $1/\tau$ nulls occur at the same frequency levels as before.

Plot (e) shows the effect of decreasing the pulse width by half, and reveals a doubling of the frequency content (bandwidth). Plot (f) decreases the pulse width by half again with the result of doubling again the frequency content. Finally, the pulse width is decreased until it is a "spike" as shown in (g). This is the *Dirac delta* or *unit impulse* function. To be technically correct, the unit impulse is a generalized function defined as having zero amplitude everywhere except at $t = 0$ and such that it contains unit area under the total waveform (Haykin 78, p. 46). Note that in plot (g) the frequency spectrum is perfectly flat since the $1/\tau$ null approaches infinity as τ approaches zero.

Plots (h) and (i) show that the harmonic frequencies remain the same if the period T is held constant, but that the pulse width τ is decreased in the same manner as (e) and (f). Also, the null crossings are increased in the same proportions as in plots (e) and (f). Finally, the pulse width is decreased to a "spike" as it was in (g). Note how the envelopes of (g) and (j) are entirely similar. Since plot (j) has the same period as (b), (h), and (i), the harmonic spacing is the same. The function in plot (j) is termed the *Dirac comb* (Haykin 78, p. 52).

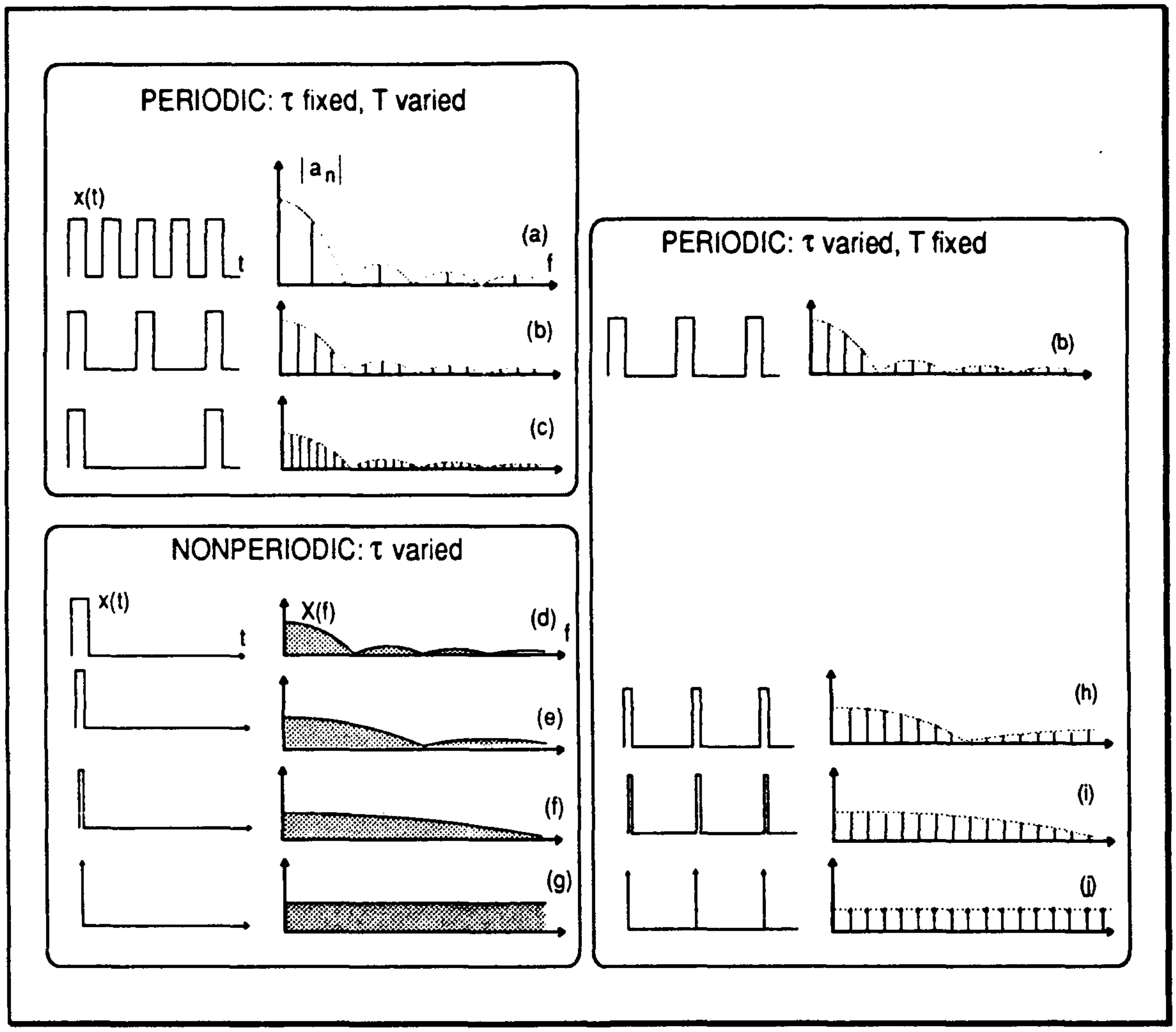

Figure 4-10. Spectrum as Tau and Period Vary

V. ENCODING

A. DIGITAL AND ANALOG SIGNALS

Analog communications have been around for over a century and started with the invention of the telephone by Alexander Graham Bell in 1876 (Roden 88, p. 2). Analog signals are characterized by continuously varying voltages. On the other hand, digital signals are characterized by discrete voltage levels. Once encoded there are usually only two possible discrete voltage levels; they are associated with ones and zeros.

Digital communications became available commercially in 1962 with the introduction of Bell System's T1 transmission system (24 digital phone lines at 1.544 Mbps). "By 1976, digital communications were replacing analog communications in many areas traditionally dominated by the analog format. This explosion of interest in digital communications was made possible by revolutionary advances in computers and solid state electronics." (Roden 88, p. 2)

There are several advantages of digital over analog communications (giving rise to the increased use of digital signals). These advantages include (Roden 88, p. 5; Stallings 88, p. 39):

- *Ease of signal regeneration - better noise immunity.* Repeaters clean up a signal contaminated by noise and reproduce an exact replica of the signal. This is possible since the repeater must decide only between two levels: 1 and 0. On the other hand, analog signals use amplifiers which amplify the noise as well as the information signal.
- *Ease of error detection/correction.* The binary data stream can be encoded with additional bits for error detection and correction. These bits can be processed by the receiver and used to request a retransmission if an error is detected. Other codes allow for the correction of an error bit at the receiver end. This procedure is called *forward error correction* and retransmission is not necessary.

- *Ease of encryption.* Encryption allows for secure communication so that only an authorized sender can transmit the message and only the intended receiver can understand it.
- *Integration of different data types.* Video, voice, and data can be transmitted over the same communication medium. Integrated Services Data Network is an example of this application.

There are also disadvantages with digital signals. Among these are:

- *Increased bandwidth.* This disadvantage will become apparent in Pulse Code Modulation.
- *Increased complexity.* This occurs because synchronization is required.

B. PULSE CODE MODULATION

Pulse Code Modulation (PCM[1]) is the process of transforming analog signals into digital signals. *Digitizer* is a neme given to devices that perform the analog to digital conversion (Roden 88, p. 101). A decoder is used on the receiving end to convert the digital signal back to analog. *Codec* is the name given to a device that performs both operations. PCM involves sampling, quantizing, and encoding. (Stallings 88, p. 82)

Figure 4-1 illustrates the PCM process. First the signal must be bandlimited (a) to provide boundary conditions on the signal. Otherwise, during sampling undesired higher frequency components will overlap the lower frequency components making it impossible to distinguish between the two (Hamming 89, p. 22). This overlapping is called *aliasing.* Voltage samples of the bandlimited signal are taken at regular intervals at a rate (f_s, see inset b) twice (Nyquist rate) the highest frequency component of the original signal ($f_s > 2f_{max}$). This sampled signal now

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1 Differential PCM and Delta Modulation (DM) are other analog to digital techniques but are are not discussed. DPCM lowers transmission rate by reducing redundancy. DM is a 1 bit version of DPCM where a staircase approximation, based on the difference between the input signal and the latest approximation, is incremented in the direction of the signal. (Feher 83, p. 53)

contains all the information of the original signal. Since filters are not perfect[2], a guard band is included to prevent aliasing and the sampling is usually a little higher than twice f_{max} ($f_s = 2.5f_{max}$ is a common value).

The resultant sampled signal is called *Pulse Amplitude Modulation*.[3] Visualizing how a sampled signal contains the original signal is difficult when viewing only the time domain. However, if the reader looks at the frequency domain (c), it is easy to see that the original signal is multiplied by a constant (dc component, no shift in frequency) and several "carrier" frequencies. The signal remains analog since the sampled signal can assume a continuous range of amplitudes. The quantizer assigns discrete values to the pulses (d). The graph shows two levels of accuracy: one associated with two bits (solid line) and one with four bits (dashed lines). The difference between the (actual) analog value and the discrete (approximation) value is called *quantization error* (circled area in d for the two-bit case). The discrete voltage values can now be line encoded where, in the simplest case, there is a one-to-one correspondence between bits and signal state. This is shown in (e) where a binary 0 is represented by a lower voltage value and a 1 by a higher value. Various line encoding schemes, and the associated frequency spectrums, are contained in Appendix A. (Roden 85, p. 349; Stallings 88, p. 69)

The quantization error can be reduced by using more quantization levels, such as the four-bit case. However, the accuracy gained is at the expense of increased bandwidth, as shown in (f). The quantization error can be minimized for a given set

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2 A filter's bandwidth is measured by the 3 dB point (half power). Therefore, some frequencies lie outside the filter bandwidth.

3 Analog PAM signals were originally used for analog TDM applications. Note, however, the communications road map of Figure 3-1 does not include a provision for PAM TDM.

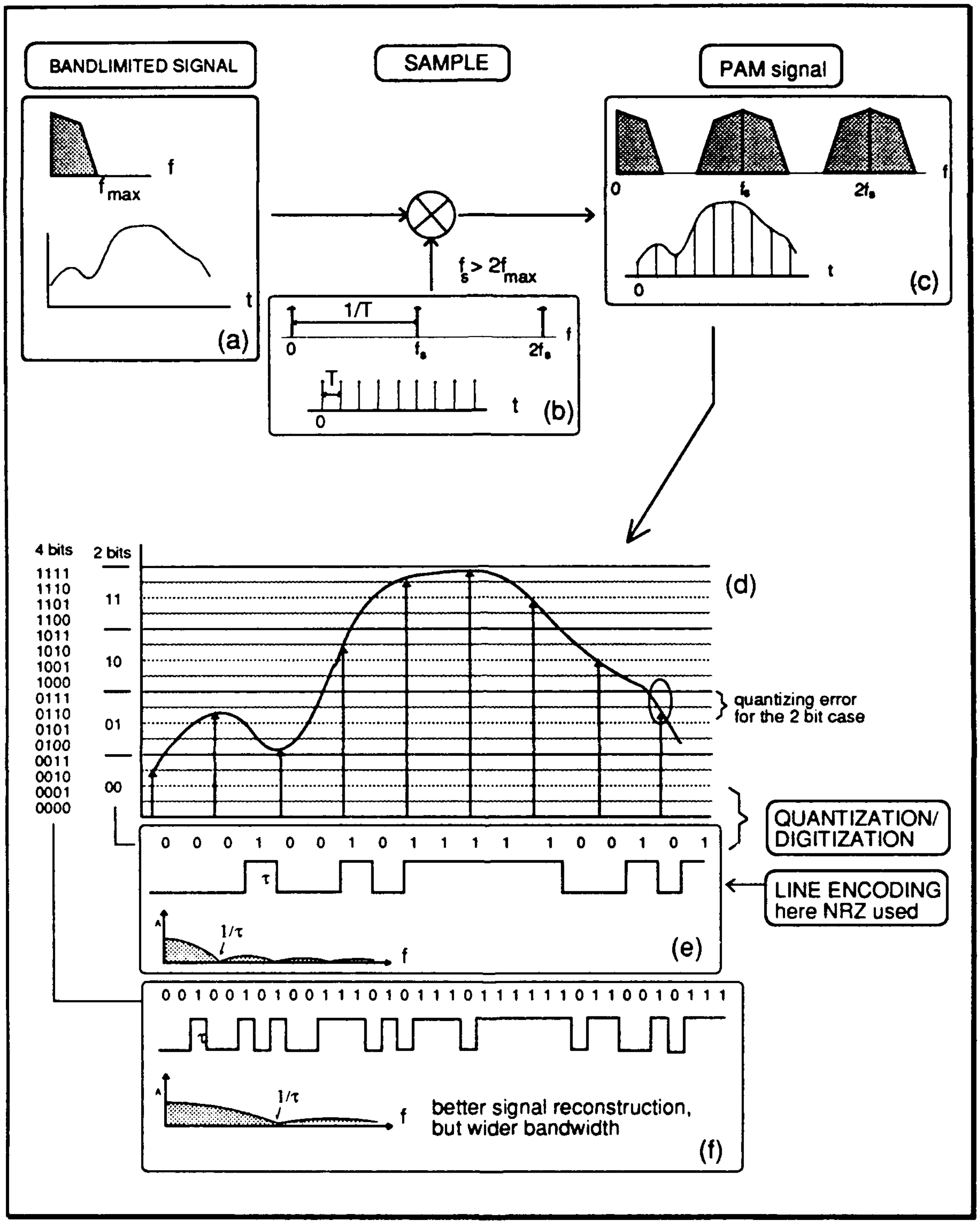

Figure 5-1. Pulse Code Modulation Process

of quantization levels through *nonlinear encoding* or *companding*. The interested reader can turn to Appendix A for examples. The following example gives an idea of numbers used in actual commercial systems.

High-quality commercial digital voice systems use 256 quantization levels: 128 for positive and 128 for negative voltages. This requires 8 information bits for each sample (2^8=256). The bandlimited voice signal is 3.4 kHz with a sampling rate of 8,000 samples per second. Therefore, the transmission rate is 64 kbits/second (8 bits per sample x 8,000 samples per second). If the digital signal is bandlimited to 1/tau, and 1 bit is transmitted per signaling state, the bandwidth is 64 kHz. Therefore, a significant increase in bandwidth is incurred to gain the advantages of digital signaling. (Feher 83, p. 51)

C. MULTIPLEXING

> When several communications channels are needed between the same two points, significant economies may be realized by sending all the messages on one transmission facility, a process called *multiplexing* (Carlson 86, p. 279).

Two types of multiplexing are now discussed: frequency-division multiplexing (FDM) and time-division multiplexing (TDM). A third type of multiplexing is code-division multiplexing (CDM), which involves spread spectrum signals. The interested reader should see Appendix B for CDM details.

1. Frequency-Division Multiplexing

Frequency-division multiplexing is a technique for transmitting several signals (see Figure 5-2(a)) over the same medium by first individually modulating subcarriers with baseband signals (b). The various subcarrier frequencies are relatively low, unsuitable for RF transmission, and form a new "baseband" signal. The modulated subcarriers are combined in a linear summing circuit (c). The composite signal, in turn, modulates a carrier at a higher frequency suitable to the medium (d). (Stanley 82, p. 290; Carlson 86, p. 279)

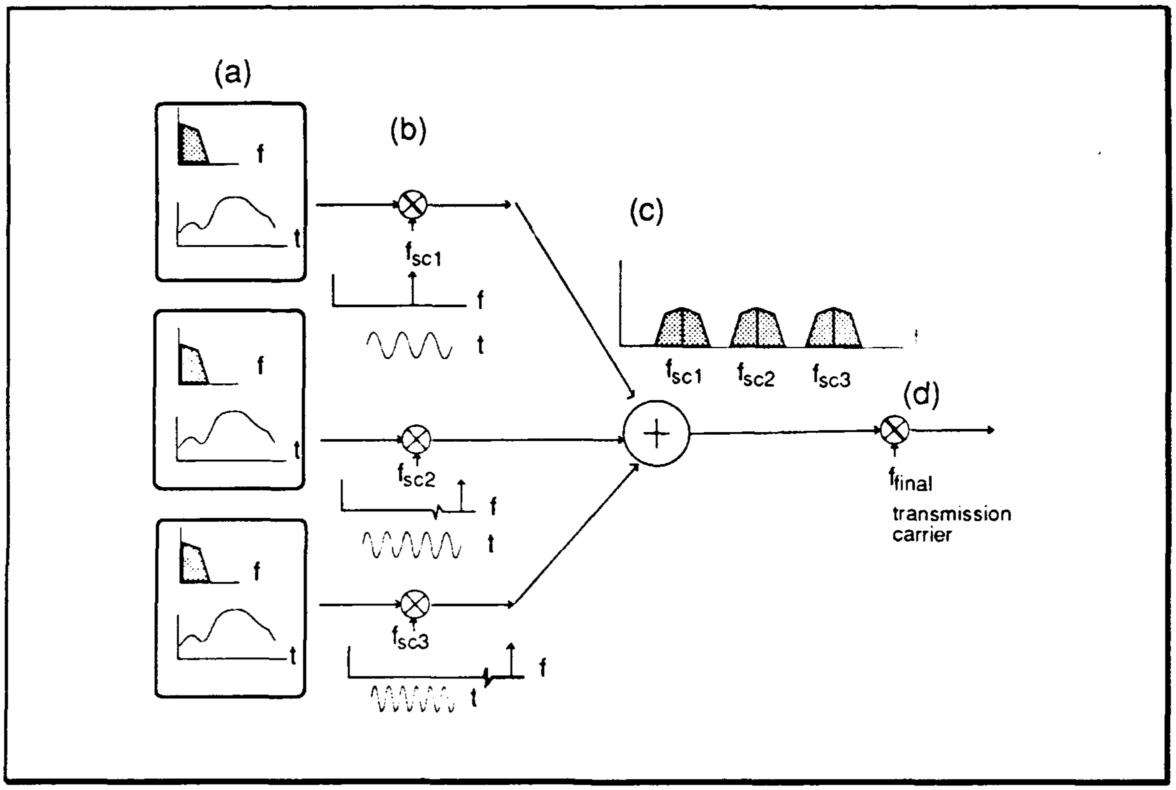

Figure 5-2. Frequency-Division Multiplexing

2. Time-Division Multiplexing

Time-division multiplexing is the interleaving of sampled data from different sources. Each signal occupies a discrete time slot while occupying the total frequency spectrum. Time-division multiplexing can be performed on both analog and digital signals. (Couch 90, p. 194)

The interleaving of analog signals can be visualized by a rotating commutator that samples the bandlimited signal, as shown in Figure 4-3(a). All three input signals are drawn exactly the same, allowing for easier visualization of the interleaved sampled pulses in (b). The sampled signals are then digitized (c) and transmitted as a digital PCM/TDM signal. Also, the PCM/TDM signal can be interleaved[4] with other digital data (d). The widespread Bell System's T1 carrier handles 24 multiplexed voice channels and transmits at 1.544 Mbps (24 channels x 64 kbps plus 8 kbps for synchronization). TDM requires synchronization in order to extract the desired signal at the proper time slot. (Stallings 88, p. 179)

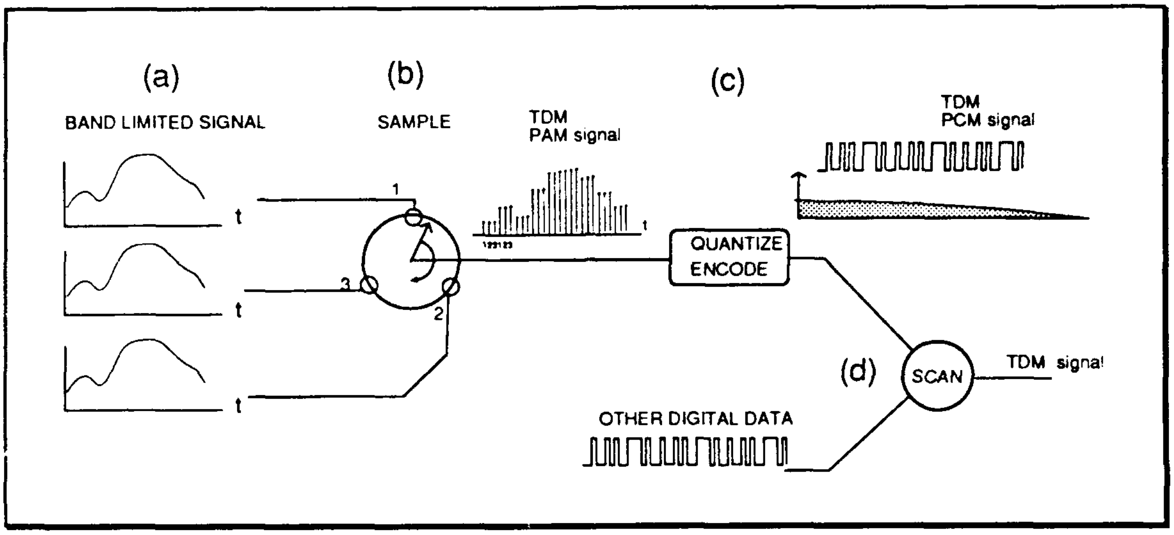

Figure 5-2. Time-Division Multiplexing

4 Interleaving occurs at the bit level or in blocks of bytes or larger quantities.

VI. MODULATION

A. BASEBAND VERSES BROADBAND

The signals discussed up to this point have all been baseband[1] and can be transmitted directly over copper wires. The range, however, is very limited and the signals do not provide access to aircraft, satellites and mobile ground units. On the other hand, modulation allows us to extend the range using guided[2] (coax and fiber optics) and unguided (air, space, water) mediums. Unguided transmission is accomplished by translating the signals to radio frequencies (RF) for electromagnetic wave propagation.

Modulation is the process of changing the amplitude, frequency, or phase of a sinusoid·' carrier (also known as continuous wave, CW) in accordance with a signal containing the information (called the modulating signal).

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1 Baseband signals have frequency spectrums that extend from (or near) dc up to finite value, usually less than a few megahertz. Baseband signals are also called low-pass signals. When a baseband signal modulates a carrier, the resultant signal is called broadband or bandpass. (Sklar 88, p. 52)

2 Modulation is also used for transmission over twisted pair, coax cable and fiber optics. Your home computer modem (modulator/demodulator) modulates your computer data for transmission over the telephone lines (twisted pair). Your cable TV company uses modulation for frequency division-multiplexing the various TV channels and transmits them to your home via coax cable. Modulators for fiber optic transmission convert the electrical signals into infrared pulses (on and off). Fiber optics provides for virtually noise free transmission. Computer local area networks (LANs) use all three mediums, depending on the vendor and application. (Madron 88; Stallings 88; Tannenbaum 88)

Figure 6-1 shows the modulating process (using binary phase-shift keying, BPSK) of impressing a baseband information signal (a) onto a sinusoid carrier (b) and producing a broadband signal (c). The broadband signal provides efficient transmission by propagating a wave via a smaller, physically realizable antenna (see Chapter III). Also, one must consider the channels' attenuating, noise inducing, and distorting effects when choosing a carrier frequency (see Chapter VII).

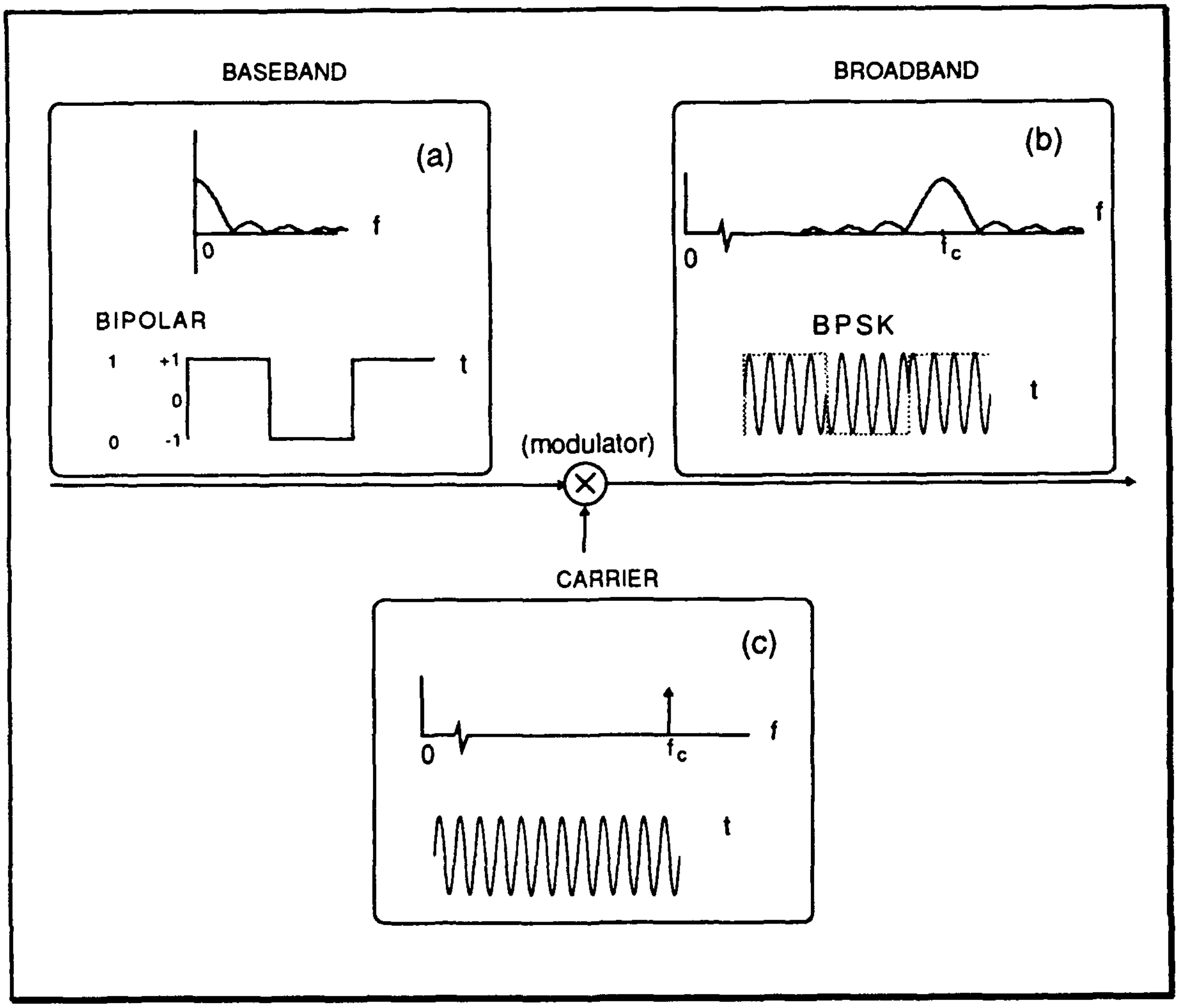

Figure 6-1. Baseband Signal Modulating a Carrier and Producing a Broadband Signal

The signal shown in Figures 6-1 has unlimited bandwidth, characterized by the instantaneous phase shifting at the data transition points. However, signals must be bandlimited to conserve bandwidth, prevent interference with other signals, and reduce the amount of noise entering the system. Therefore, a balance must be found allowing for recognition of data transitions yet minimizing interference and noise. Bandlimiting can be accomplished before or after modulation. Baseband signals can be bandlimited before modulation via a low pass filter (LPF); this is called *premodulation filtering*. The modulated signal can be bandlimited via a *postmodulation* band pass filter (BPF). Both produce the same amplitude spectra.[3] (Feher 83, pp. 79-133)

Figure 6-2 shows the effects of bandlimiting a baseband signal and illustrates why textbooks typically choose bandwidths (BW) of $0.5/\tau$ to $1/\tau$. The left column shows a time domain representation of an unfiltered baseband pulse train. Low pass filtering (frequency domain) is shown in the middle column for various bandwidths. The resultant filtered[4] time domain signal is shown in the right hand column. Note how the bandwidths of $0.5/\tau$ and $1/\tau$ pass sufficient spectral components allowing for recognizable pulses, while bandwidths of $2/\tau$ or greater provide little improvement in signal reconstruction.

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[3] Provided the LPF and BPF transfer functions are the same. This also implies the BPF is symmetrical around the carrier.

[4] Taken from Figure 4-6.

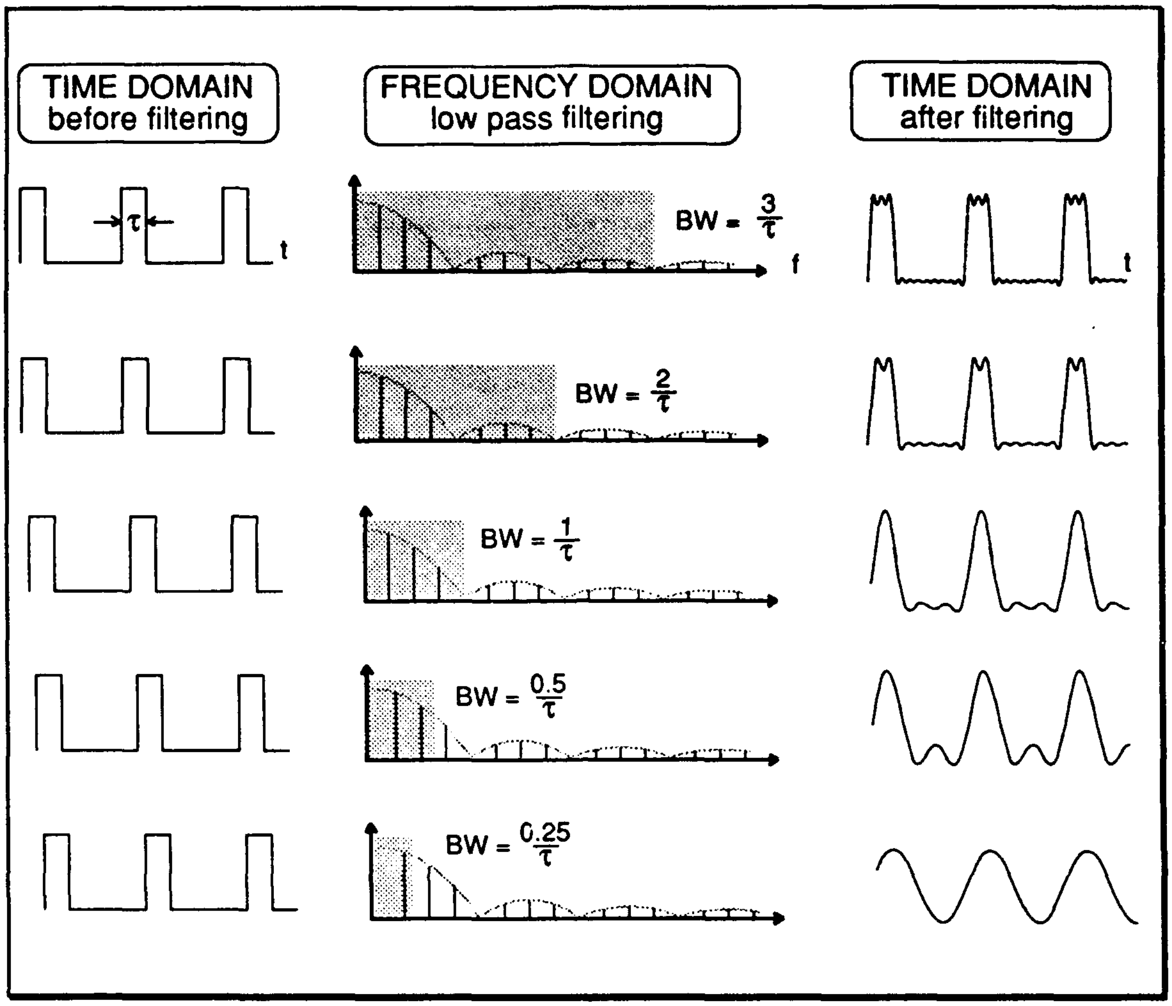

Figure 6-2. Filtering a Baseband Signal

Bandlimited signals, however, produce modulated carriers that do not have constant amplitude and that require linear channels for optimal performance. Unfortunately, satellites and other systems are power limited and tend to operate their amplifiers in a nonlinear range where the best power efficiency is found. Nonlinear channels produce extraneous sidebands and introduce out-of-band interference. There are modulation schemes that reduce these negative effects, but they are beyond the scope of this introductory material. See Appendix C for further details. (Feher 83, p. 29; Pasupathy 79, p. 15)

B. TRANSMISSION/RECEPTION TERMINOLOGY

One can get lost in the terminology used to describe the various modulation techniques. Part of the confusion is due to the use of different terminology for the exact same function. In spite of this, there are only a few basic concepts providing the foundation for modulation and demodulation. The different techniques are just variations of a few fundamental concepts.

1. Modulation

Amplitude modulation (AM) and *frequency modulation* (FM) are two types of analog modulation.[5] Amplitude shift keying (ASK), phase shift keying (PSK), and frequency shift keying (FSK) are three broad categories of digital modulation. All of the above modulation types use a sinusoid as the carrier. Examples of ASK, PSK, and FSK are shown in Figure 6-3.

Only a few carrier cycles are shown for illustrative purposes. In reality, thousand or millions of carrier cycles occur between shifting points. *Amplitude shift keying* changes the amplitude of the carrier. The ASK signal shows on-off keying (OOK), where a 1 is represented by full carrier amplitude (on) and a 0 by no carrier (off). *Phase shift keying* leaves the amplitude the same, but changes the carrier's phase. Phase measurements are made throughout the pulse against the reference and not at the instant phase changes. *Frequency shift keying* conveys information by transmitting discrete frequencies.

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[5] Phase modulation (PM) is another analog transmission technique described in textbooks, but it is seldom used in real world applications (Carlson 86, p. 254).

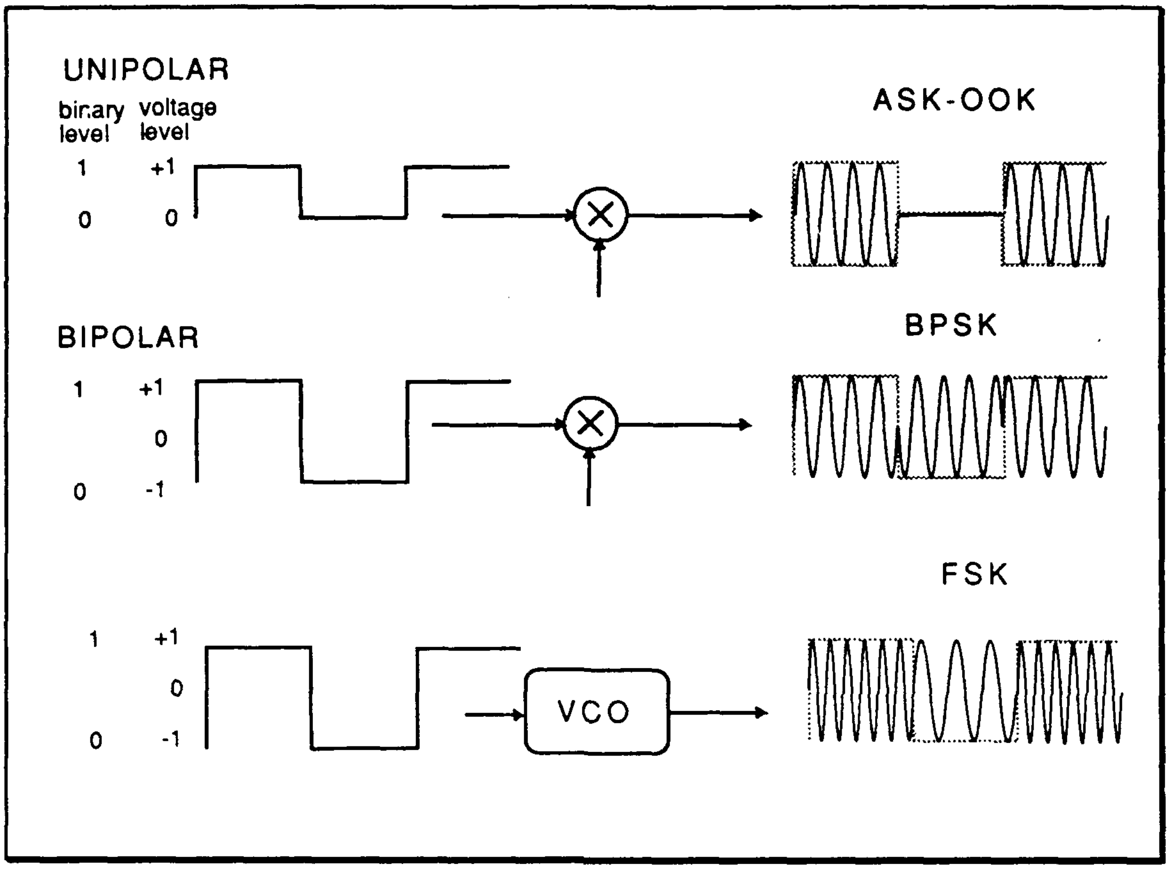

Figure 6-3. Digital Modulation Techniques

AM, ASK, and PSK[6] all use a balanced modulator as the basic modulating device. Modulation occurs by multiplying the input signal with the carrier (hence, another name, *product modulation*). Double-side band AM, ASK and binary PSK can be classified as linear modulation. *Linear modulation* is characterized by shifting the baseband signal to a higher frequency while keeping the message spectrum the same. The message spectrum and its mirror image is centered around the carrier frequency and gives rise to the name *double-side band* as shown in Figure 6-1. (Feher 83, p. 131)

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6 There are various PSK techniques which use two or more balanced modulators, such as QPSK.

Terms used in place of linear modulation are *product modulation, balanced modulation, mixing, and multiplying*. Terms to describe the process of linear modulation are *frequency translation, frequency shifting, frequency conversion*, and *heterodyning*. Conversion is split into two parts describing the act of modulating. *Up conversion* produces a broadband signal and *down conversion* brings the signal down to baseband. (Stanley 82, p. 130 & 325; Carlson 86, p. 193; Feher 83, p. 131)

There are other forms of PSK, such as *quadrature* PSK (QPSK, also called *quaternary* PSK), which transmit two bits per signaling state (phase) vice the one bit for binary PSK. The QPSK bandwidth is ideally half that of BPSK for the same data rate. Therefore, an exact replication of the baseband signal does not occur and it is not classified as true double-side band signal.

The ability to have a signaling state represent multiple bits is the reason why a distinction is made between *bauds* (number of signaling states per second) and number of bits per second (bps). For example, the transmission line may support 1200 baud and BPSK, and NRZ line coding will transmit 1200 bits per second (b/s). On the other hand, the QPSK[7] signal transmits at 1200 baud, but is transmitting twice the number of bits at 2400 bps.

FSK signals are produced by switching between two (or more) carrier sources, or by using a voltage controlled oscillator (VCO). Analog FM signals are also produced with a VCO. Bessel functions are used to model the FM frequency components. FM is sometimes referred to as *exponential CW modulation* because the modulation is nonlinear (nonlinear in the sense that the modulated spectrum

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[7] ASK signals are used in conjunction with PSK signals to produce *amplitude phase keying* (APK).

looks nothing like the baseband spectrum). The VCO output frequencies are still linearly dependent on the input voltages.

Figure 6-4 shows the product modulators (AM, ASK, PSK) and VCOs (FM, FSK) for the modulations indicated. Also shown is FSK generated by two carrier sources. Note the similarities in producing AM, ASK, and PSK signals. The primary difference is the baseband input (analog vice digital).

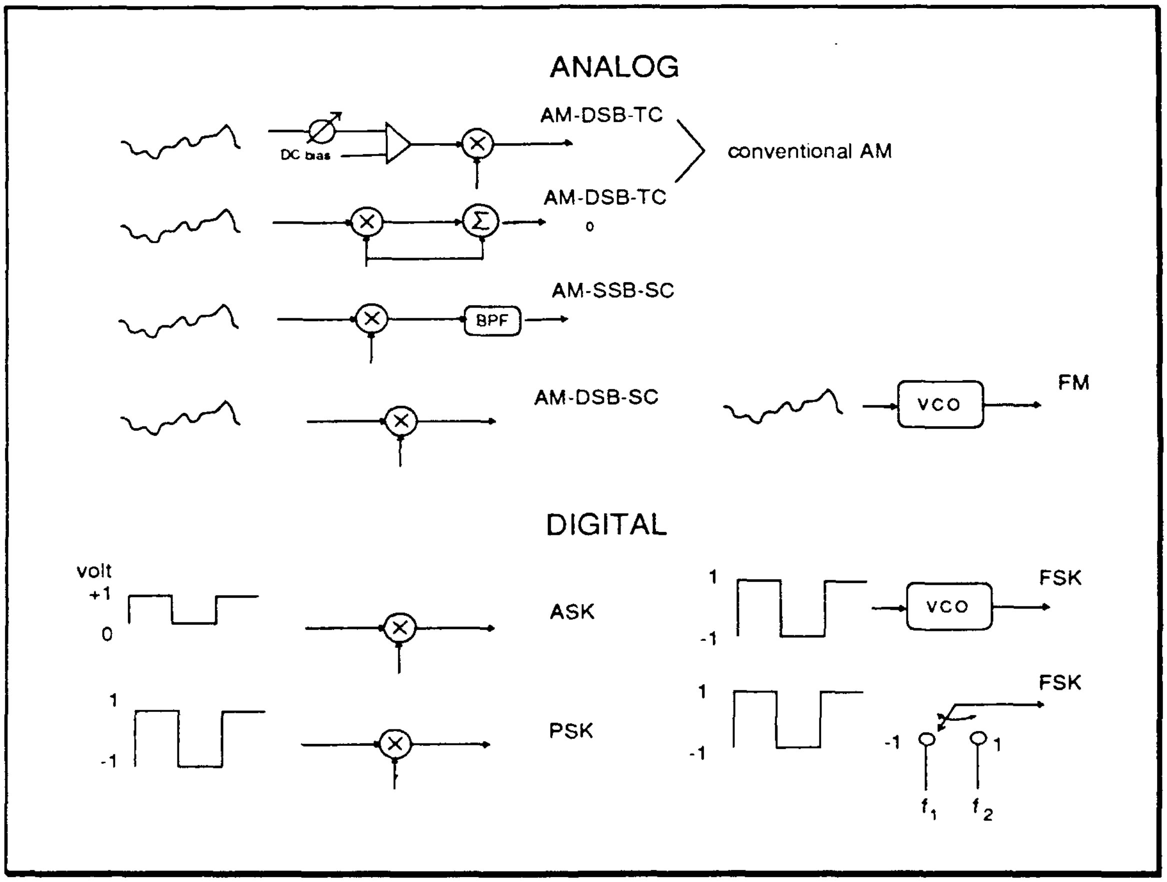

Figure 6-4. Basic Methods for Modulating by a Baseband Signal

2. Demodulation

Demodulation is the inverse process of modulation and falls into two general categories: coherent detection and noncoherent detection. *Coherent detection* requires that a receiver reference carrier be synchronous[8] with the transmitted carrier. Therefore, *synchronous detection* is another term for coherent detection. *Coherent detection* is accomplished by using the same type of multiplier (product) that was used to generate the signal. Therefore, product *detection/demodulation* is synonymous with coherent detection In general coherent systems provide better performance in the presence of noise. (Stanley 82, p. 324)

Noncoherent detectors do not require carrrier coherency and are much simpler to build. *Envelope detection* is one type of noncoherent detection and it is used for AM and ASK (OOK) signals. FSK systems also use envelope detection by filtering the different discrete frequencies through separate bandpass filters; this creates in effect, separate OOK type signals. Coherent FSK is possible with a continuous phase frequency shift-keyed (CPFSK) or minimum shift-keyed (MSK) signals. Analog FM signals are passed through a discriminator to change frequency into voltage. See Appendix C for the relationship between the various modulation techniques.

The channel is the link between the modulator and the demodulator and is the subject of the final chapter.

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[8] Synchronous means having exactly the same frequency and phase as the transmitted carrier.

VII. CHANNEL - TRANSMISSION MEDIUMS

The channel provides the link between the transmitter and the receiver, and it is a limiting factor on practical transmission rates (bandwidth). The channel can be either guided or unguided. *Guided media* propagate the wave along a physical path, such as copper wire and fiber optics. Examples of *unguided media* are air, space, and seawater; they do not guide the wave. Each media imposes a different type of constraint on the signal. Figure 7-1 provides a general overview of the various medias and their impairments on the transmitted signal. (Stallings 85, p. 22)

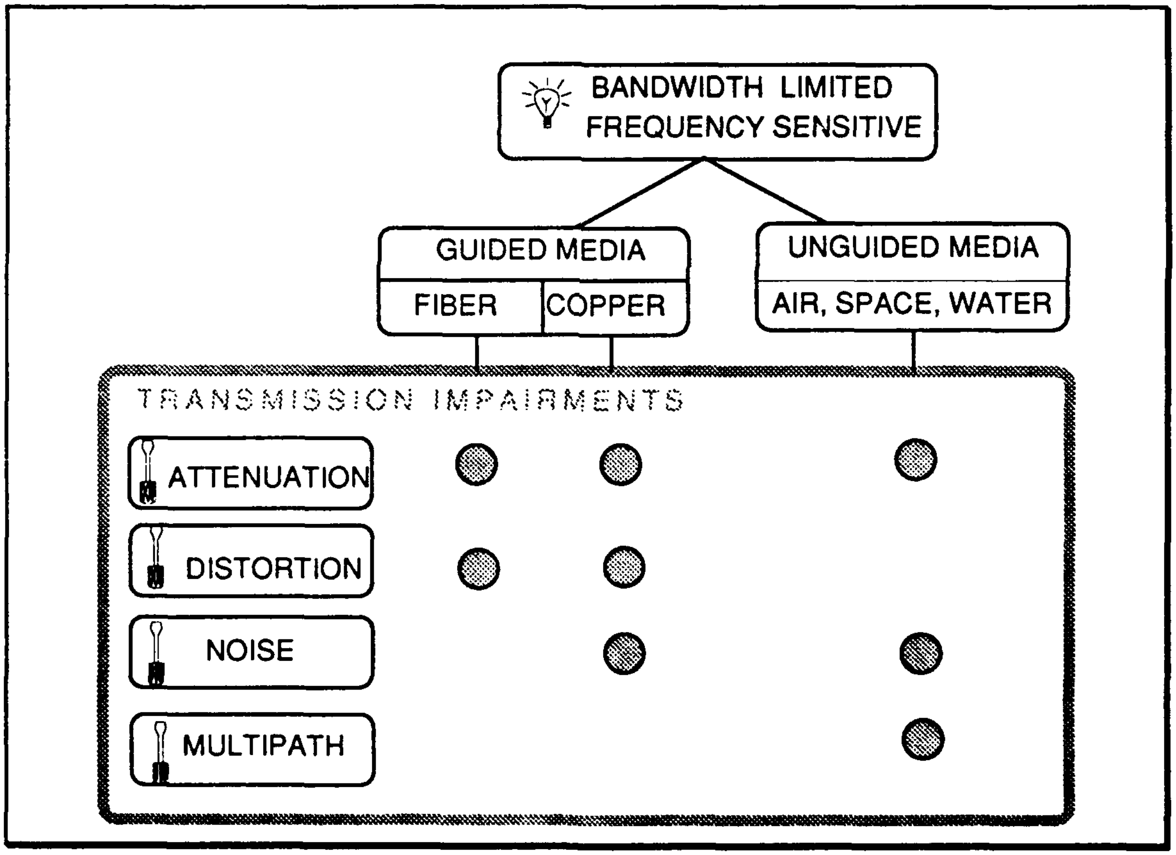

Figure 7.1. Overview of Transmission Mediums and Impairments

A. UNGUIDED PROPAGATION

Unguided propagation is associated with radio frequency transmissions where an electromagnetic wave is propagated in the atmosphere, space, or seawater. There are four general categories of radio wave propagation: ground wave, sky wave, space wave, and scatter (Griffiths 87, p. 28). Each category is a result of natural phenomena affecting some radio wavelengths but not others. The natural phenomena occur at different altitudes, and these in turn affect the maximum achievable communication distances.

Communications are often desired over long distances at high data rates. Unfortunately, natural phenomena support an inverse-type relationship. For example, the low carrier frequencies support world-wide communications, but only support low data rates. On the other end of the spectrum, the higher carrier frequencies support high data rates, but are limited to short line-of-sight terrestrial distances unless expensive relay systems, such as satellites or repeater stations, are used.

Therefore, the author believes it is important to understand the relative vertical height of the atmosphere in relation to space and communication satellites. Figure 7-2 shows a proportionally correct drawing from earth to geosynchronous altitudes; the relative heights of the troposphere, ionosphere, and low earth orbiting vehicles are expanded to the right (Gatland 81, p. 22; Random House Encyclopedia 77, p. 212). The chart shows that our atmosphere is indeed very thin and outer space is not that far away. Also included are relative heights achievable by selected objects. The atmospheric layers are determined by the increasing or decreasing temperature trends.

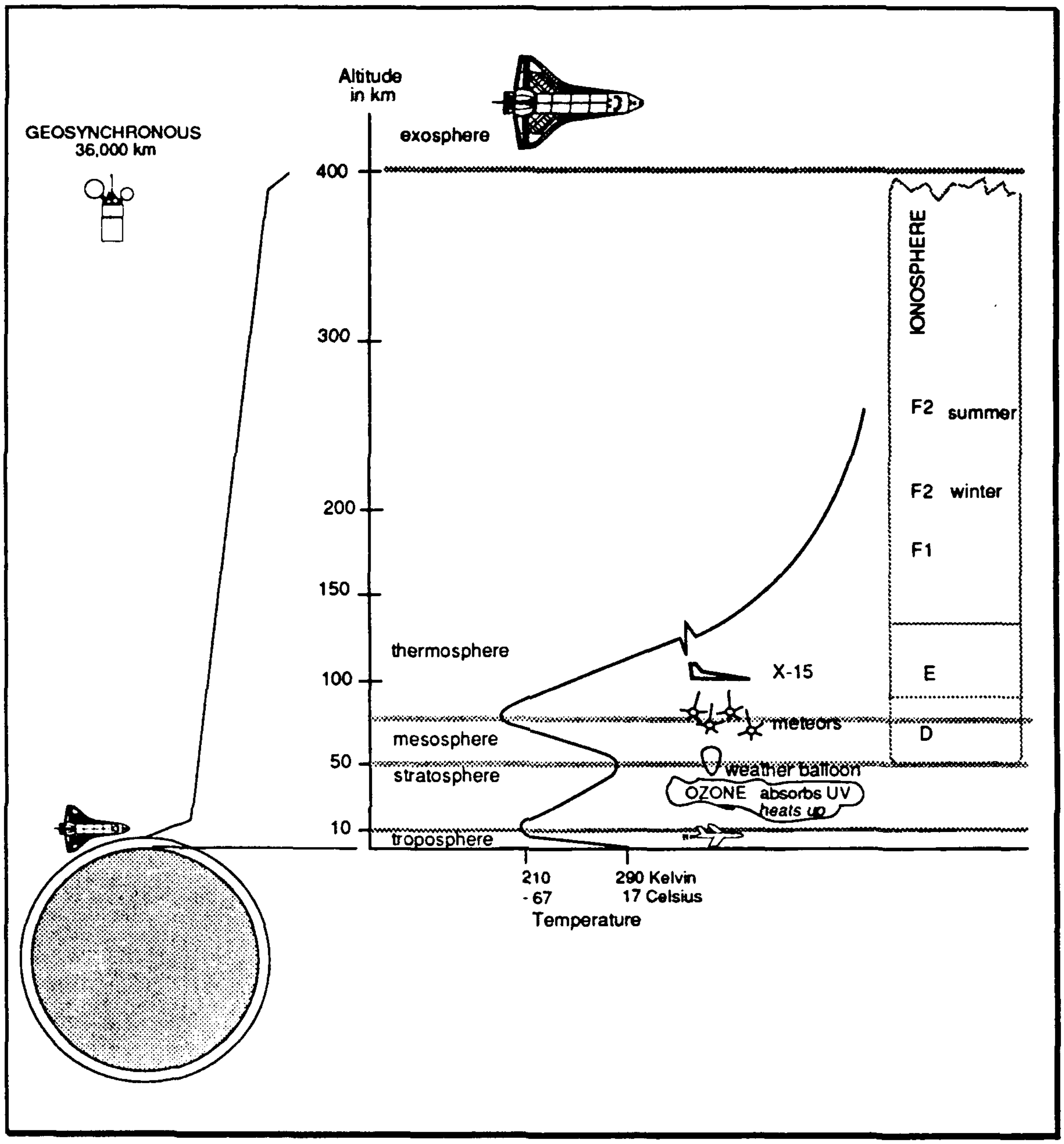

Figure 7-2. Proportional Altitudes of Space and Atmosphere

The four radio wave propagation categories mentioned above support different frequencies. These frequencies are commonly divided into radio bands with frequencies ranging from hertz to gigahertz, and the associated wavelengths (wavelength = speed of light /frequency) are megameters to millimeters. Table I is given below for reference purposes. (Gleditsch, 87; p. 16)

TABLE I

Band Designator		Frequency	Wavelength
ELF	- extremely low frequency	30 - 300 Hz	10 - 1 Mm
VF	- voice frequency	300 - 3000 Hz	1 - 1 Mm
VLF	- very low frequency	3 - 30 kHz	100 - 10 km
LF	- low frequency	30 - 300 kHz	10 - 1 km
MF	- medium frequency	300 - 3000 kHz	1 -.1 km
HF	- high frequency	3 - 30 MHz	100 - 10 m
VHF	- very high frequency	30 - 300 MHz	10 - 1 m
UHF	- ultra high frequency	300 - 3000 MHz	1 -.1 m
SHF	- super high frequency	3 - 30 GHz	100 - 10 mm
EHF	- extremely high frequency	30 - 300 GHz	10 - 1 mm

Figure 7-3 is a composite of the four propagation categories with the radio bands forming the horizontal axis and the atmospheric altitudes forming the vertical axis. Various frequencies are affected differently by the physical structure of the atmosphere. The atmospheric property that primarily determines how the electromagnetic wave behaves is listed below the radio bands. The shaded areas on the diagram indicate at what altitude and frequency the propagation occurs. Each propagation category is discussed immediately following the composite chart.

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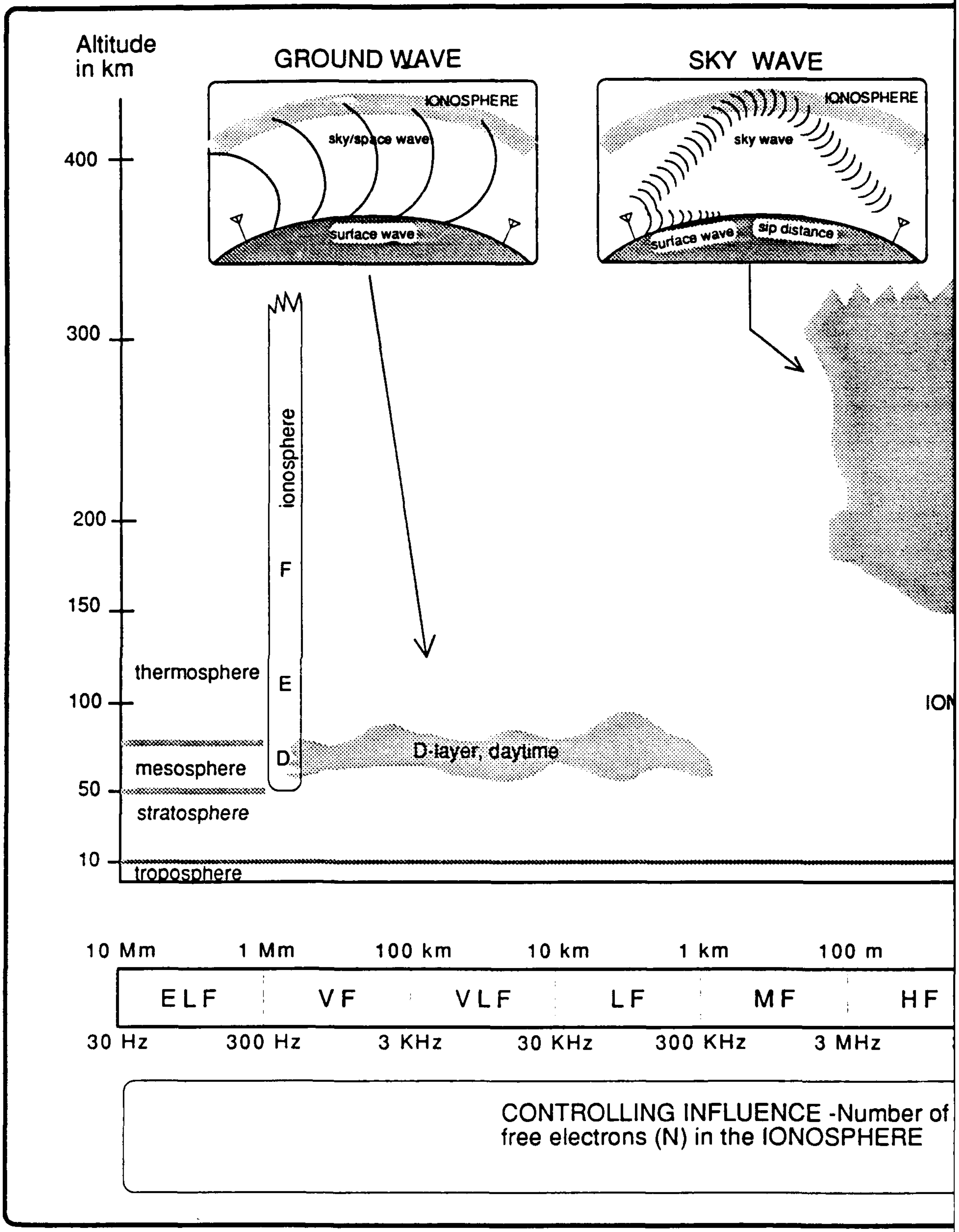
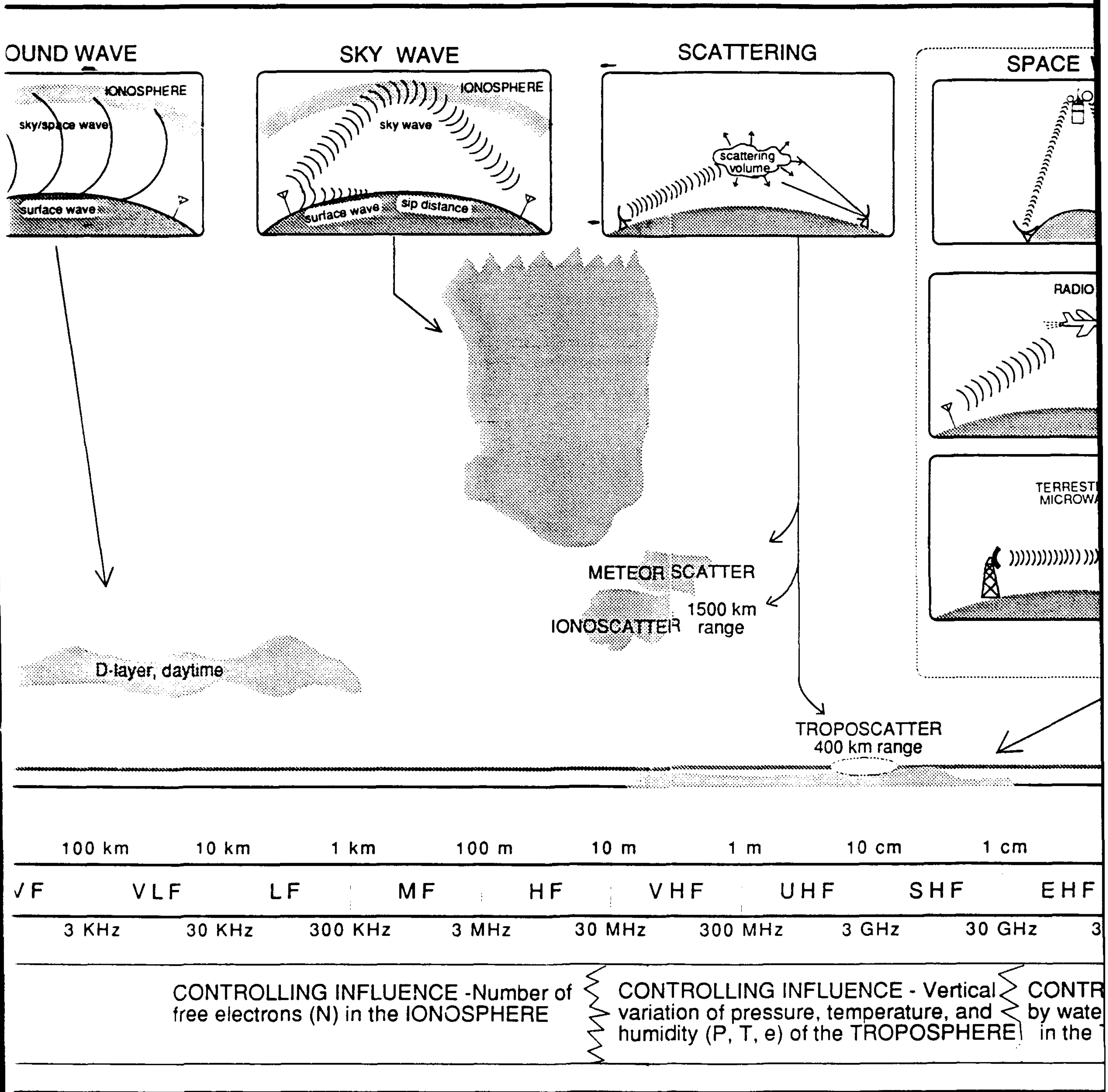

Altitude in km
GROUND WAVE
IONOSPHERE
sky/space wave
surface wave
SKY WAVE
IONOSPHERE
sky wave
surface wave
sip distance
400
300
200
150
100
50
10
ionosphere
F
E
D
thermosphere
mesosphere
stratosphere
troposphere
D-layer, daytime
ION
10 Mm
1 Mm
100 km
10 km
1 km
100 m
ELF
VF
VLF
LF
MF
HF
30 Hz
300 Hz
3 KHz
30 KHz
300 KHz
3 MHz
CONTROLLING INFLUENCE -Number of
free electrons (N) in the IONOSPHERE

1 of 3



OUND WAVE
IONOSPHERE
sky/space wave
surface wave
SKY WAVE
IONOSPHERE
sky wave
surface wave
sip distance
SCATTERING
scattering volume
SPACE
RADIO
TERREST
MICROW
METEOR SCATTER
IONOSCATTER
1500 km range
D-layer, daytime
TROPOSCATTER
400 km range
100 km
10 km
1 km
100 m
10 m
1 m
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VF
VLF
LF
MF
HF
VHF
UHF
SHF
EHF
3 KHz
30 KHz
300 KHz
3 MHz
30 MHz
300 MHz
3 GHz
30 GHz
CONTROLLING INFLUENCE -Number of free electrons (N) in the IONOSPHERE
CONTROLLING INFLUENCE - Vertical variation of pressure, temperature, and humidity (P, T, e) of the TROPOSPHERE
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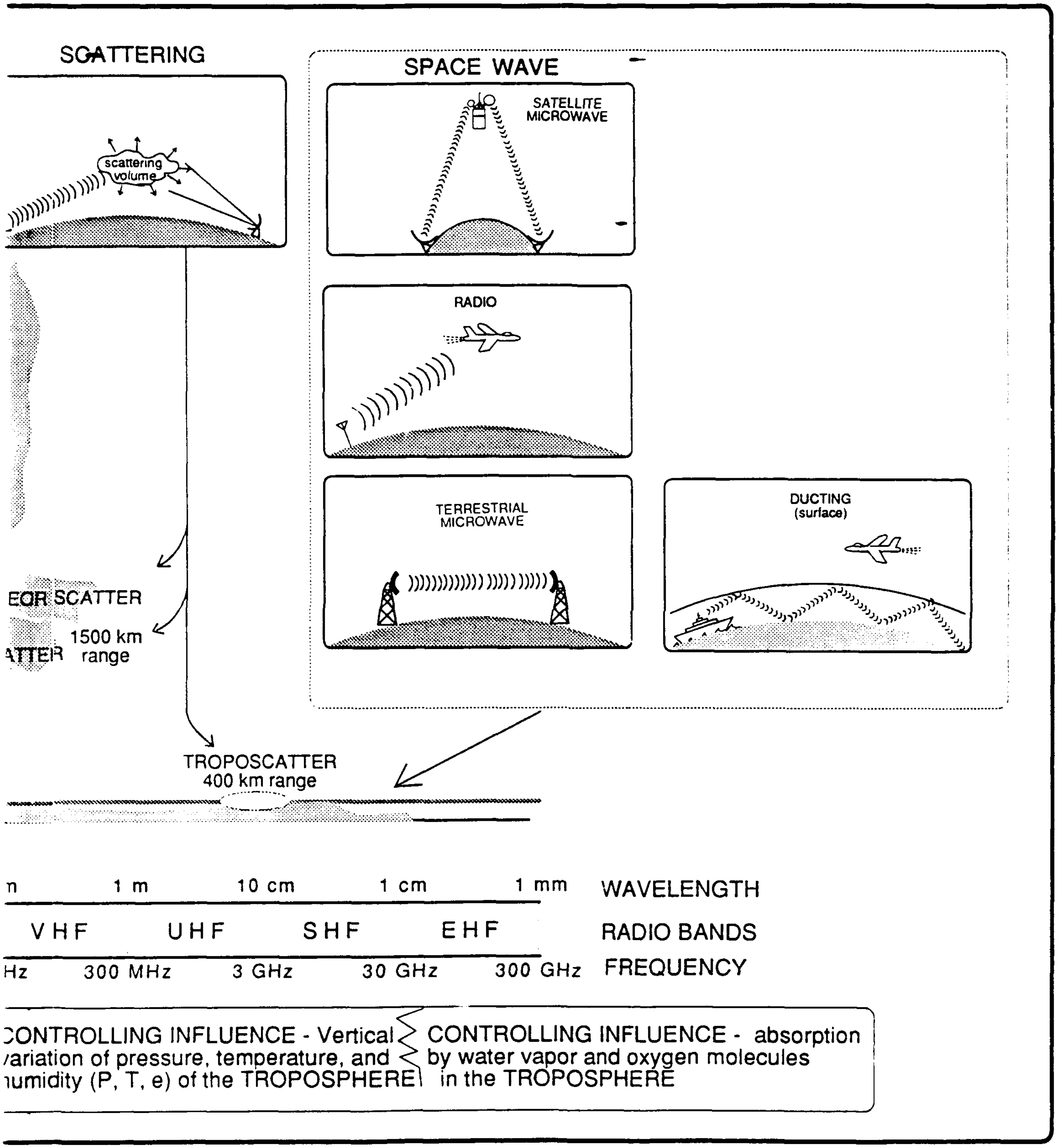

Figure 7-3.
Radio Wave Propagation

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1. Ground Wave

Ground wave propagation occurs in the ELF to mid-MF frequency ranges. These frequencies have very long wavelengths and can be thought of as "going everywhere." There are different theories for the actual propagating mechanism. The ground wave can be thought of as a surface wave which follows the contour of the earth, with a wavefront propagating via a sky wave (or space wave) to the ionosphere where it is finally refracted. Ground waves can also be thought of as one wave propagating in a waveguide formed by the earth and the ionosphere. (Miller 88, p. 467; Griffiths 87, p. 27, Burrows 68, p. 62)

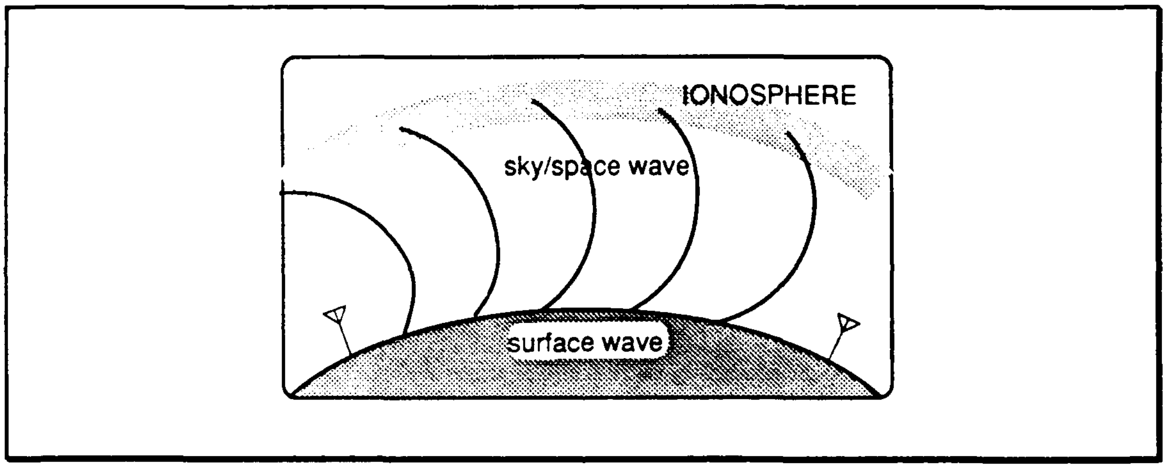

Figure 7-4. Ground Wave Propagation

Precise mechanisms aside, ground wave propagation is very stable (not affected by daily or seasonal ionospheric changes) and supports virtually world-wide propagation (VLF frequencies). Unfortunately, the low frequencies provide very limited bandwidth and require very long antennas. For example, a 30 kHz carrier has a wavelength of 10 kilometers. With one-quarter or one-tenth the wavelength usually cited as the minimum antenna length for efficient transmission, the ratio implies an antenna that is 1,000 to 2,500 meters long. For most applications this situation would not be practical; shorter antennas are substituted which severely limits the efficiency. (Smith 89)

The ground wave is used for long range navigation, such as LORAN (LF) and Omega (VLF). It is also used to communicate with submerged submarines, as shown in Figure 7-5 (copied from C31 Handbook 88, p. 55). ELF penetrates seawater to 100 meters, VLF to10 meters, and LF only to 1 meter (Gleditsch 87, p. 16).

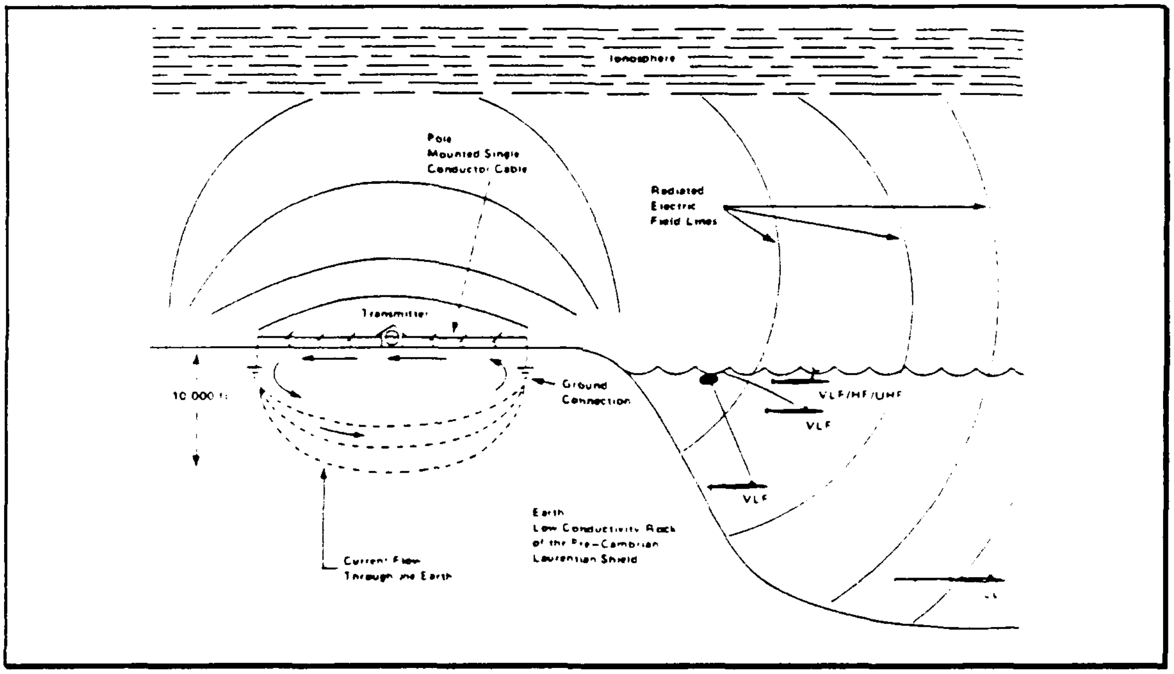

Figure 7-5. Ground Wave Communication With Submarines

2. Sky Wave

Sky waves occur in the HF range where the ionosphere refracts the wavefront back to earth. The surface wave is absorbed by the ground and is not propagated very far.

The ionosphere gets its name from the ionization of the thermosphere, producing free electrons. There are three layers: D, E, and F. The D layer is practically present only in the daytime, the E layer is sporadic and unpredictable, and the F layer is always present due to a slow recombination of free electrons and positive ions. These layers are not uniform in thickness and are subject to solar disturbances. Although HF is not stable, it is very popular because it allows for long haul communication of reasonable data rates without satellite links or laying of cables and repeaters. (Griffiths 87, p. 63)

There are a host of anomalous propagation phenomena associated with HF that are not discussed here. You should refer to a standard text book, such as *Radio Wave Propagation and Antennas* (Griffiths 87) for details.

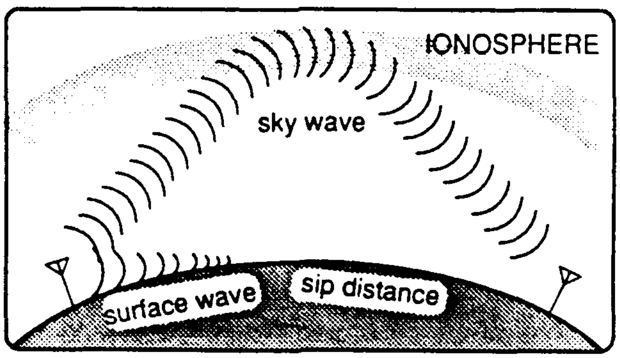

Figure 7-6. Sky Wave Propagation

3. Space Wave

This propagation occurs in the VHF, UHF, and SHF ranges. Space waves travel in straight paths for the most part and are also called *direct* or *line-of-sight* (LOS) waves. The signal may be reflected off objects such as buildings and then are called a *reflected* wave. Space waves become severely attenuated in the EHF and higher range by the absorbing effects of water vapor and oxygen molecules in the troposphere. The loss due to absorption varies logarithmically and it is usually expressed as a loss in decibels per kilometer.

Below absorption frequencies the primary loss of signal strength is due to the spreading of the wavefront. This is analogous to throwing a rock in a calm pond and watching the ripples spread. The spreading loss varies as the inverse of the square of the distance ($1/r^2$) from the source. Spreading loss occurs for ground as well as for sky waves.

VHF, UHF, and SHF frequencies are associated with FM radio, TV, aircraft communications, microwaves, radars, and satellite communications. Unlike ground and sky waves, space waves are not severely affected by the ionosphere. Instead, the pressure, temperature, and humidity of the troposphere affect space wave propagation. The rate of change of these variables with respect to altitude determines how much the space wave is refracted. The most severe refraction results in the formation of a *trapping layer*. The trapping layer bends the waves back towards the earth (the trapping layer is frequency sensitive, so not all the frequencies will be trapped). The trapping layer forms a duct in which the electromagnetic wave propagates. (Patterson 88, pp. 25-31)

A duct can have a lower boundary at the surface (surface duct) or above the surface (elevated duct). The transmitter must lie within the duct for trapping to occur; if conditions are right a signal can propagate hundreds, and sometimes thousands, of miles. (Picquenard 74; pp. 25-27; Burrows 68, p. 60)

The trapping phenomenon is most prevalent around large bodies of water where there is cold moist air at low levels and a rapid increase in temperature and decrease in humidity with increasing altitude. Ducting can be a good or bad phenomena depending on the application. A plane outside the duct, as shown in Figure 7-7, is at the best attack altitude since it would not be picked up by radar, yet at the worst altitude if its mission is jamming. (Patterson 88, p. 93)

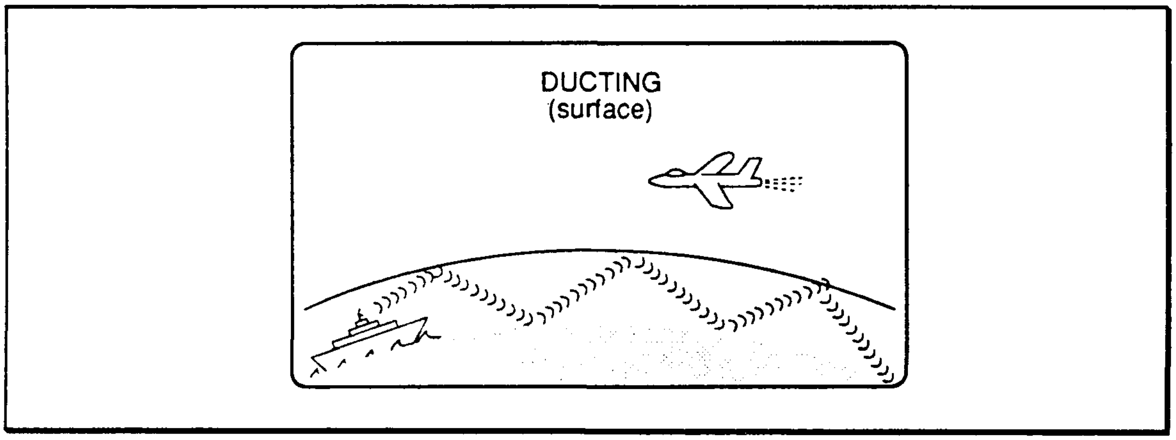

Figure 7-7. Surface Duct Caused by Trapping

A plane inside the duct (Figure 7-8) is at the best altitude for a jamming mission. Conversely, it is at the worst altitude for an attack mission since the plane could be picked up on radar even hundreds of miles beyond the horizon. (Patterson 88, p. 93)

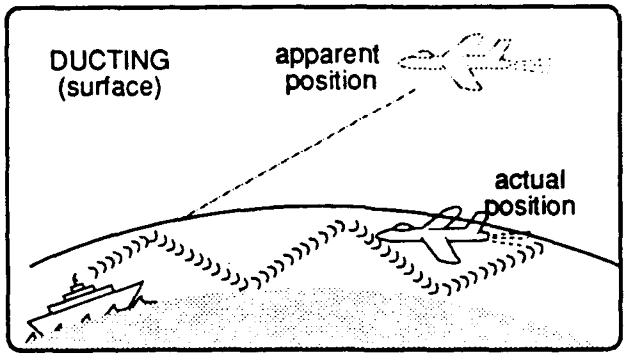

Figure 7-8. Detection of Low Flying Aircraft

The following scenario highlights the need for understanding the underlying physical principles in command and control systems in order to obviate potential catastrophic events. Most radar systems assume a standard atmosphere for refraction corrections (4/3 earth radius model). However, the radar system does not know if ducting is occurring. Assume the ship in Figure 7-9 (adapted from Wickerts 89, p. 36) is friendly and the aircraft is of unknown origin. The aircraft has an apparent position to the ship's radar as shown. A few minutes later the aircraft climbs out of the duct and the radar angle is such that the radar is no longer trapped. The aircraft is actually climbing, but the system says it is diving. In a hostile environment this perception could be viewed as a threat. The commander in this situation has only minutes or seconds to sense, process, assess, decide and act whether missiles are to be fired. This scenario was suggested as a possible explanation for why the Iranian Airbus A300 was shot down on 3 July, 1988 by the Aegis system onboard the USS Vincennes. (Wickerts 89, p. 36)

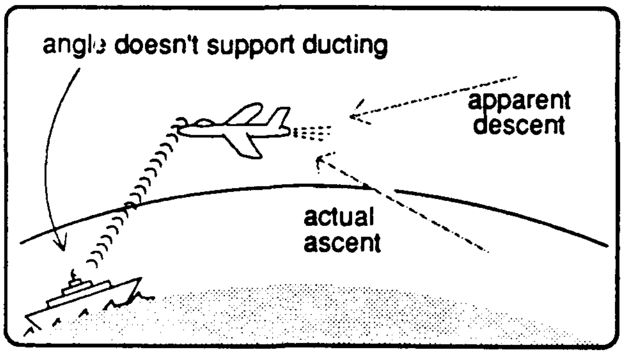

Figure 7-9. Apparent Aircraft Descent When Actually Climbing

a. Terrestrial microwave

Terrestrial microwave is line-of-sight communications commonly used for long-haul voice and television communications. Repeater spacings of 10 to 100 km are typical. Microwaves are also being used at high SHF (22 GHz, absorption high) frequencies for short point-to-point links between buildings. The high frequencies support high data rates and allow for smaller and cheaper antennas. (Stallings 88, p. 56)

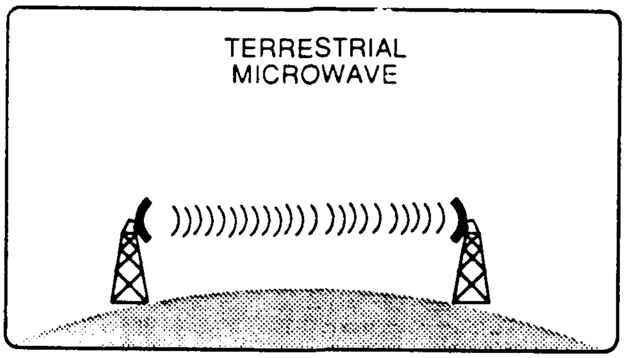

Figure 7-10. Terrestrial Microwave

b. Satellite microwave

A satellite is the optimum medium for high usage international trunks, and it is competitive with terrestrial microwave and coax for many long-distance international links. The optimum frequency range is 1 to 10 GHz. Below 1 GHz noise is a factor, both from natural noise (galactic, solar, atmospheric) and man-made noise (such as electronic devices). Above 10 GHz, there is severe attenuation by atmospheric absorption. (Stallings 88, p. 60)

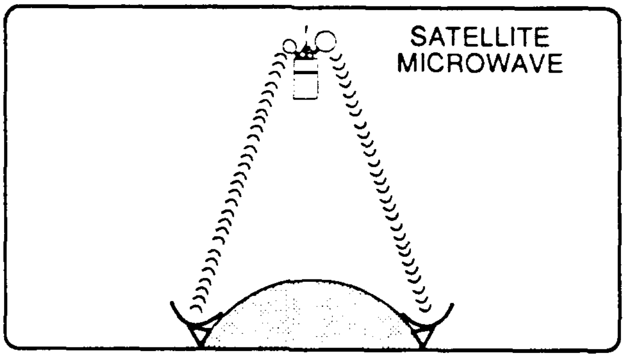

Figure 7-11. Satellite Microwave

c. Radio

Radio is usually associated with broadcast communications. Typical frequencies are from 30 MHz to 1 GHz and support kilobit rather than megabit (microwave) rates for digital transmission. The primary impairment is multipath interference, where reflection from natural or man-made objects creates multiple paths. An example of multipath interference is ghosting on TV pictures. (Stallings 88, p. 60)

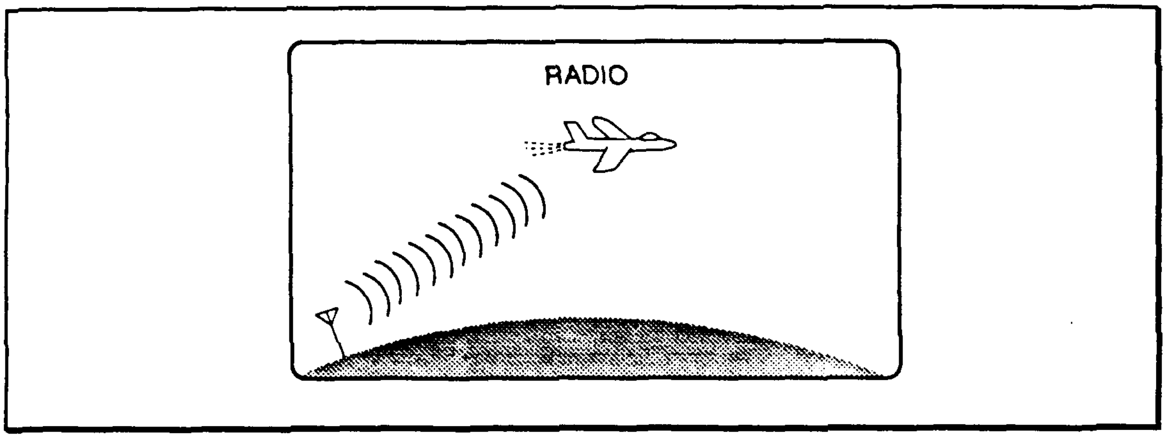

Figure 7-12. Radio

4. Scattered Wave

Scattering occurs when the electromagnetic waves are reflected by a very large number of small particles in the atmosphere or ionosphere. There are several theories as to what actually causes scattering, but the effect is real and ionospheric scatter, meteor burst, and tropospheric scatter links can be established. *Tropospheric scattering* is the most common form and is discussed next. *Ionospheric scattering, meteor burst* are not addressed in this thesis, but the reader is referred to Griffiths 87, pp. 216-234.

LOS ranges can be extended over the horizon through tropospheric scattering or troposcatter. It works by the transmitting and receiving antenna pointing to a common location in the troposphere. The transmitted signal is scattered by a scattering volume and intercepted by the receiving antenna as shown in Figure 7-13. Ranges can extend from 70 to 700 km, and operating frequencies are from 400 MHz to 7 GHz. Transmitter elevation angles are small; typically 4 degrees or less. (Griffiths 87, p. 216)

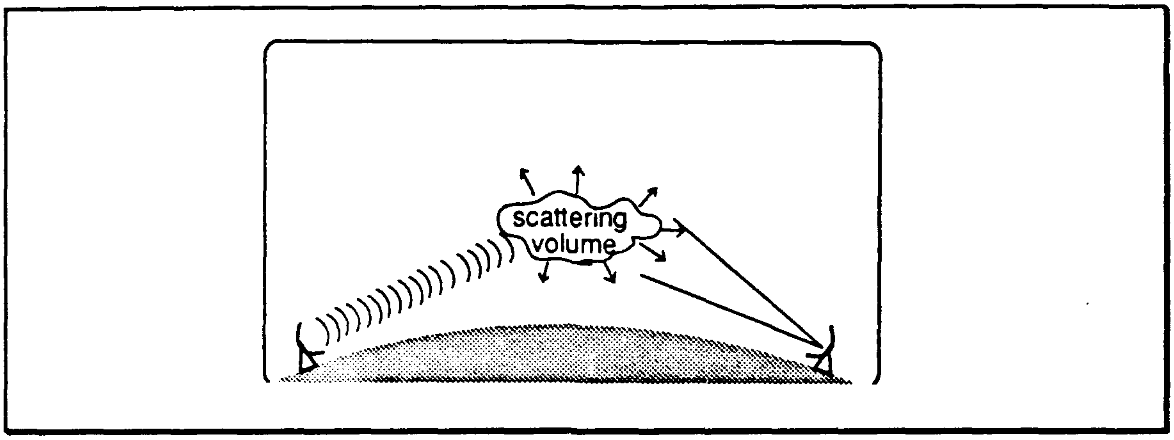

Figure 7-13. Troposcatter

Table II identifies typical modulations, bandwidths, data rates, and applications for the various radio bands (copied from Stallings 88, p. 48).

TABLE II

BAND	ANALOG DATA		DIGITAL DATA		APPLICATION
	Modulation	Bandwidth	Modulation	Data Rate	
LF	Generally not	practical	ASK, FSK, MSK	0.1 - 100 bps	Navigation
MF	AM	To 4 kHz	ASK,FSK, MSK	10 - 1000 bps	Commercial AM radio
HF	AM, SSB; FM	To 4 kHz	FSK, PSK	To 100 kbps	Shortwave radio, CB radio
VHF	AM, SSB; FM	5 kHz-5 MHz	FSK PSK	To 100 kbps	VHF television, FM radio
UHF	FM, SSB	To 20 MHz	PSK	To 10 Mbps	UHF Television, Terrestrial Microwave
SHF	FM	To 500 MHz	PSK	To 100 Mbps	Terrestrial Microwave Satellite Microwave
EHF	FM	To 1 GHz	PSK	To 750 Mbps	Experimental short point-to-point

B. GUIDED MEDIA

Table III includes data rates and bandwidths for guided media (copied from Stallings 88, p. 47).

TABLE III

Medium	Total Data Rate	Bandwidth	Repeater Spacing
Twisted pair	4 Mbps	250 kHz	2 - 10 km
Coaxial cable	500 Mbps	350 MHz	1 - 10 km
Optical fiber	2 Gbps	2 GHz	10 - 100 km

1. Twisted Pair

Twisted pair is the term given to two insulated copper wires twisted together in a regular spiral fashion. They are twisted to reduce the electromagnetic interaction between neighboring pairs. It is the backbone of the telephone system and intrabuilding communications, thus making it the most common transmission medium for both analog and digital signals. (Stallings 88, p. 47)

Twisted pair may be used for long-distance trunking and data rates up to 4 Mbps, provided the lines are conditioned. Analog signals require amplifiers every 5 to 6 kilometer; digital signals require repeaters every 2 to 3 kilometers. Twisted pair is susceptible to interference and noise because of its easy coupling with electromagnetic fields. (Stallings 88, p. 49)

2. Coax

Coax cable consists of an inner conductor surrounded by an outer conductor. The cable used for cable TV service is an example of coax. Coax is very versatile and it is used in applications requiring higher data rates than can be supported by twisted pair. Coax supports long-distance telephone. When using FDM, it can support 10,000 voice channels. Coax is also used in computer local-area networks and short-range system links. Coax's principle transmission impairments are attenuation, thermal noise, and intermodulation noise (associated with FDM). (Stallings 88, pp. 50-51)

3. Fiber Optics

Fiber optics is a transmission medium that has many appealing characteristics. Fiber allows for greater bandwidths (higher data rates and more channels), smaller size and weight, electromagnetic isolation, electromagnetic interference immunity, greater repeater spacing (lower transmission losses), and security against tapping. (Keck 85, p. 17)

Light travels in the waveguide by means of total internal reflection. Light emitting diodes (LEDs) or injection laser devices (ILDs) are used as light sources, and photodiodes or avalanche photodiodes (ADP) are used as detectors. Light propagates best in a fiber optic in three distinct wavelength[1] : 0.85, 1.30 and 1.55 microns (10^{-6} meters). This is the infrared region of the spectrum; the visible light range is 0.4 to 0.7 microns. The 0.85 micron wavelength with a LED source is used mostly for local applications and limited to data rates under 100 Mbps. The 1.30 micron wavelength is produced by a LED or ILD source and allows for higher data rates and longer distance transmission. 1.55 micron wavelengths are only produced by ILDs and they support the highest data rates and longest distances. (Keck 85, p. 17; Senior 85; p. 68; Stallings 88, pp. 54-56)

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[1] Wavelength is the more convenient unit when dealing in the optical range. Wavelength is simply the speed of light divided by the frequency.

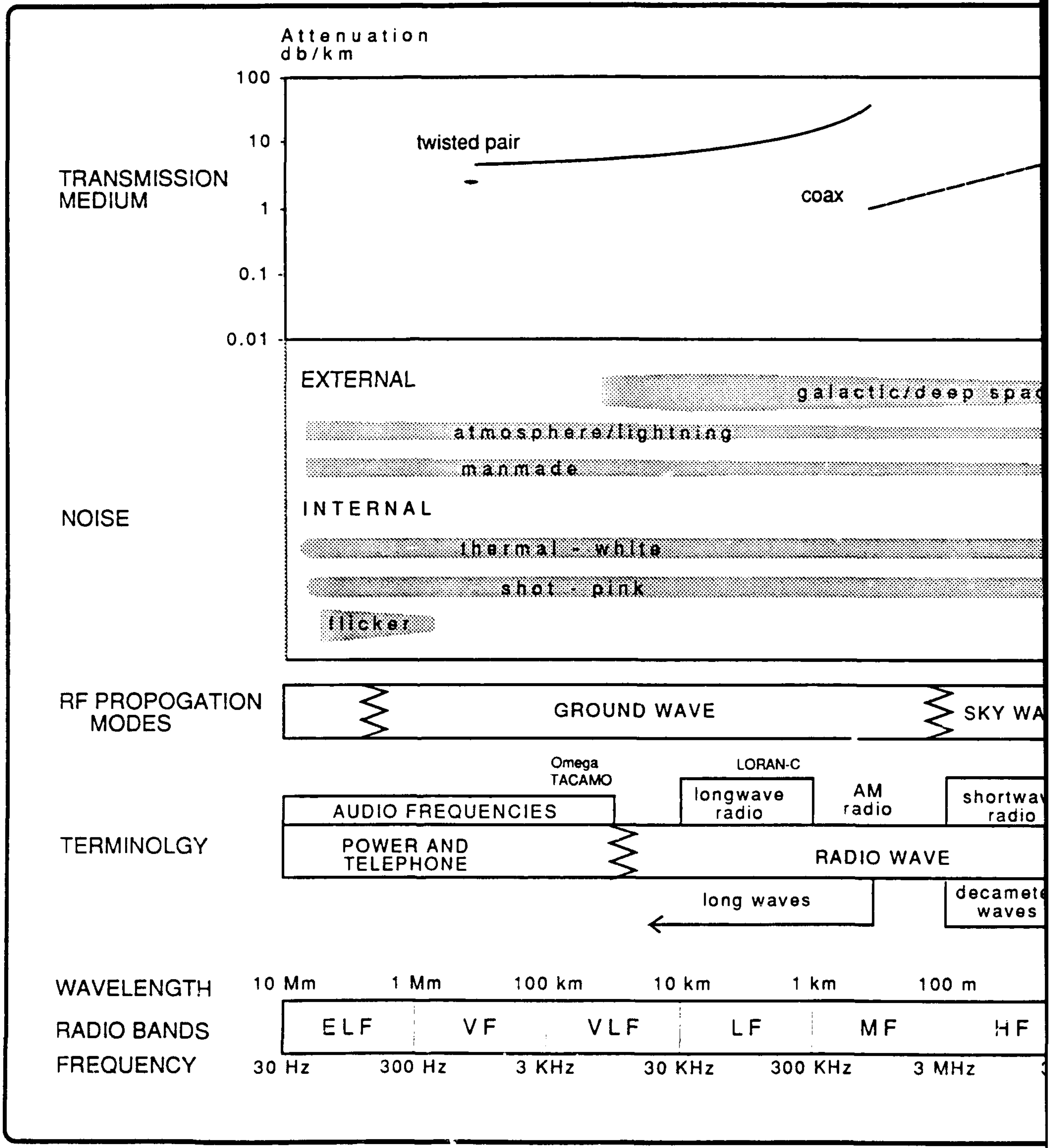
Fiber optics are extremly thin, with the thinnest fiber core diameter on same order of magnitude as light wavelengths. Core diameters range from 2 to 8 microns for single mode fibers and 50 to 125 microns for multimode fibers (Stallings 88, p. 56). For comparison, the average hair is about 70 microns in diameter and the thinnest blond hair is only 17 microns. Early developers of fiber optics had a difficult time trying to splice or connect the fibers because of its small size. New techniques have made the process easier, but is is still no trivial task.

Signal losses occur from absorption and scattering and they are strongly dependent on wavelength. The primary limitation on bandwidth is dispersion. *Dispersion* is the broadening of the light pulse as it travels along the fiber and it causes intersymbol interference when the pulses start to overlap one another (individual pulse are indistinguishable). The reader desiring more details on fiber optics should read Appendix D. (Keck 85, p. 17, Senior 85; pp. 76-78)

C. TRANSMISSION IMPAIRMENTS

Sections A and B mentioned transmission impairments as part of the discussion on unguided and guided media. However, transmission impairments are often cited in their own categories (such as attenuation, distortion, multipath, and noise) because they adversely affect the receiver's ability to reconstruct the source message. These categories are defined in Appendix D.

Figure 7-14 is a composite chart relating the earlier discussions on guided and unguided media to the frequency ranges for which noise and attenuation are significant. Care must be taken in reading the chart since several subjects are presented together. Attenuation (in decibels per kilometer) is shown for various transmission mediums. The frequencies for which noise is a factor are shown below the attenuation chart. Comparing the top two charts you can see why satellite communications within 1 to 10 GHz are so popular (they are above most noise sources and below high atmospheric absorption). Finally, different radio wave propagation modes and terminology are documented in the figure.


Attenuation
db/km
100
10
1
0.1
0.01
twisted pair
coax
TRANSMISSION
MEDIUM
EXTERNAL
galactic/deep spa
atmosphere/lightning
manmade
INTERNAL
thermal - white
shot - pink
flicker
NOISE
RF PROPOGATION
MODES
GROUND WAVE
SKY WA
Omega
TACAMO
LORAN-C
longwave
radio
AM
radio
shortwa
radio
AUDIO FREQUENCIES
TERMINOLGY
POWER AND
TELEPHONE
RADIO WAVE
long waves
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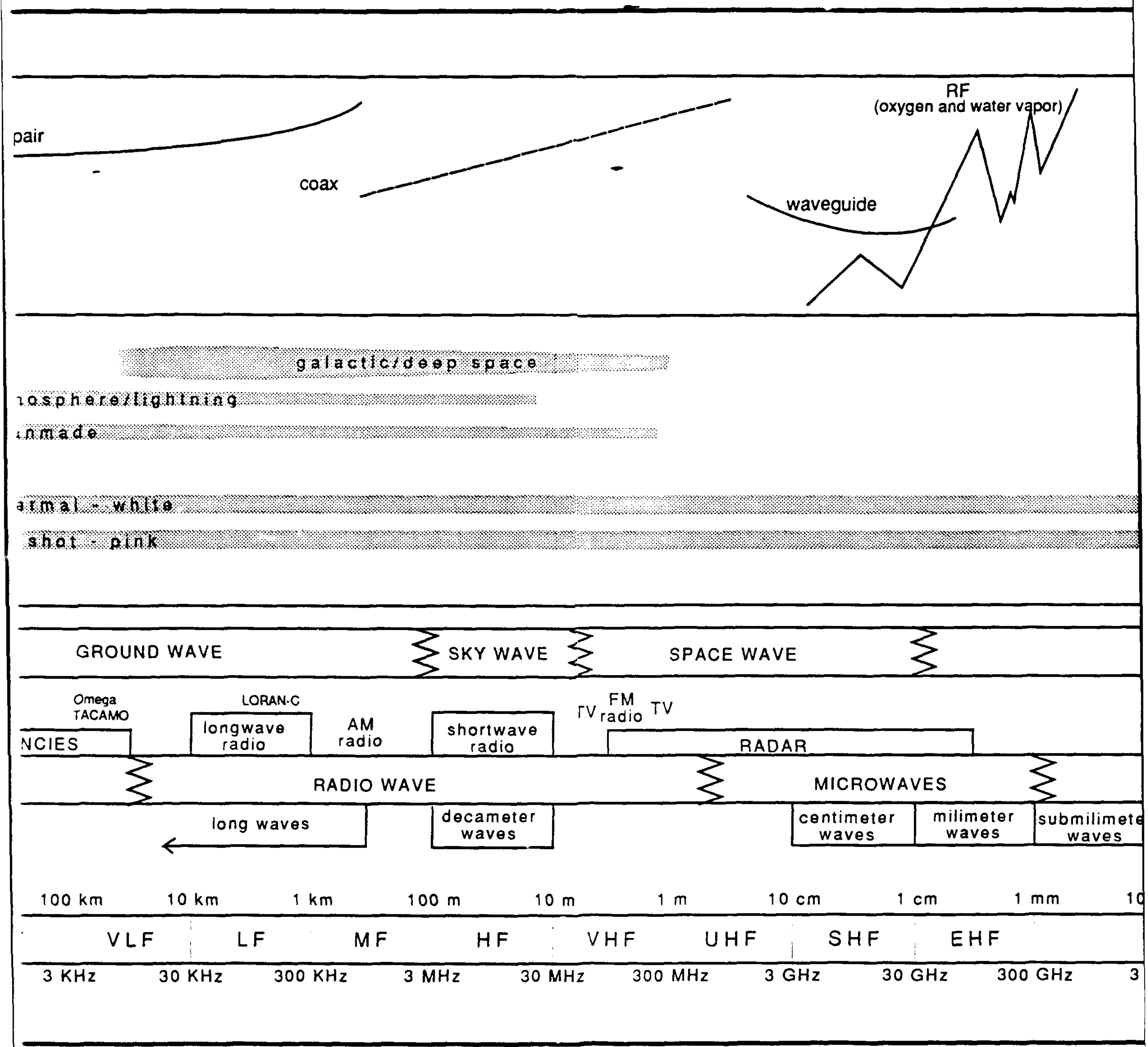
waves
WAVELENGTH
10 Mm
1 Mm
100 km
10 km
1 km
100 m
RADIO BANDS
ELF
VF
VLF
LF
MF
HF
FREQUENCY
30 Hz
300 Hz
3 KHz
30 KHz
300 KHz
3 MHz



pair
coax
waveguide
RF
(oxygen and water vapor)
galactic/deep space
osphere/lightning
nmade
rmal - white
shot - pink
GROUND WAVE
SKY WAVE
SPACE WAVE
Omega
TACAMO
LORAN-C
NCIES
longwave radio
AM radio
shortwave radio
TV
FM radio
TV
RADAR
RADIO WAVE
MICROWAVES
long waves
decameter waves
centimeter waves
milimeter waves
submilimete waves
100 km
10 km
1 km
100 m
10 m
1 m
10 cm
1 cm
1 mm
10
VLF
LF
MF
HF
VHF
UHF
SHF
EHF
3 KHz
30 KHz
300 KHz
3 MHz
30 MHz
300 MHz
3 GHz
30 GHz
300 GHz
3

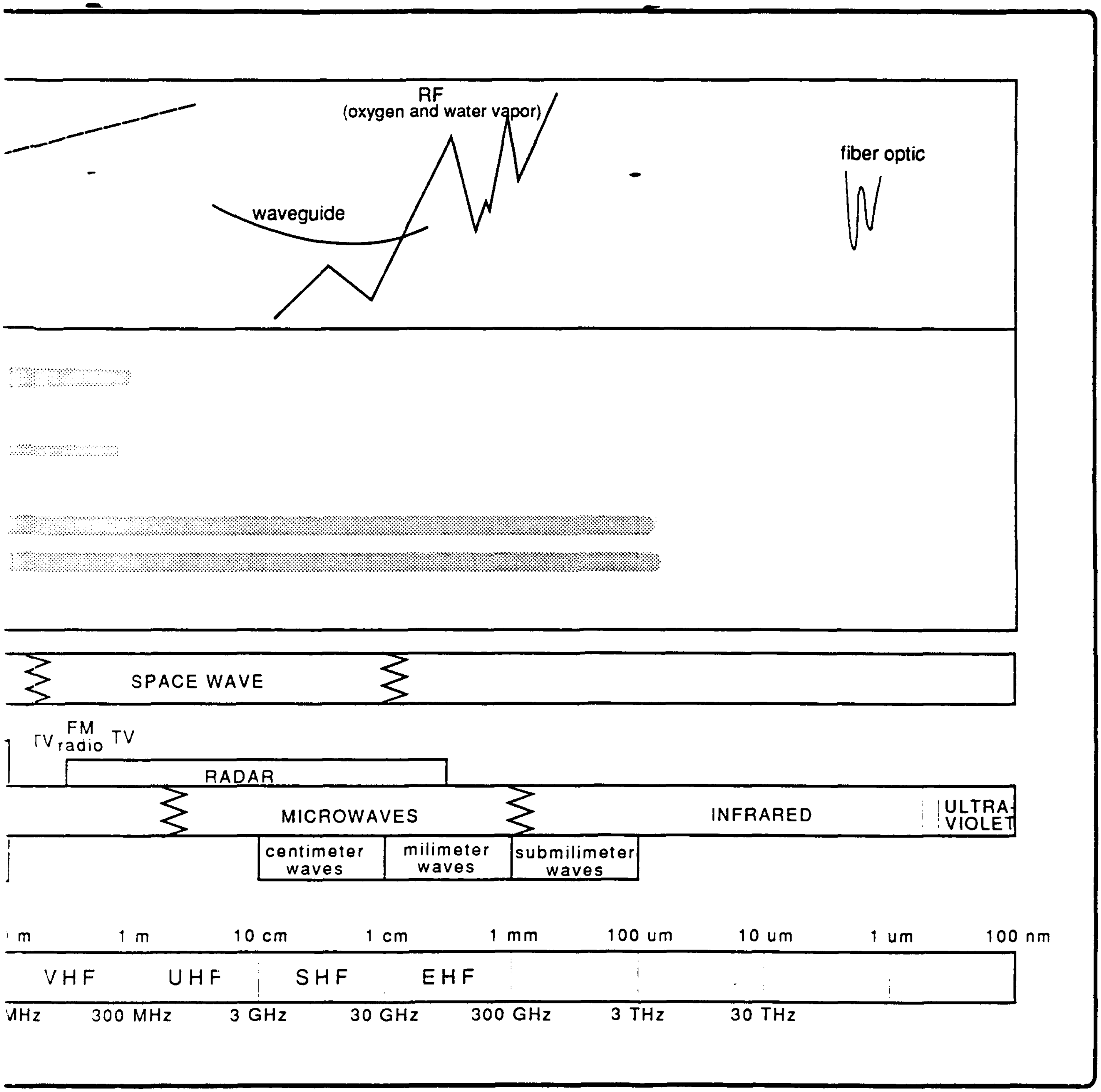


Figure 7-14.
Impairment and Terminology Composite

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VIII. SUMMARY

An electronic communications system and its associated frequency domain is very dynamic and complex. They can be intimidating to the average person. However, this thesis has shown there are few fundamentals upon which all communications systems are built. Also, focusing on the important factors mitigates the tendency to get lost in the details.

Any communication system, regardless of how well it meets the technical specifications, is worthless if it does not meet the needs of the commander. Therefore, one must understand certain basic requirements on the human level, relating to the command and control process itself, before effective technical specifications can be established for an appropriate (acceptable) communication system.

Central to any communications system are the limitations imposed by bandwidth and noise. Mathematics plays a key role in providing insight to the processes involved. The concept of bandwidth is developed through Fourier analysis. An integrated set of graphics best shows the relationship between the time and frequency domains and illustrates how the bandwidth increases as the pulse width decreases.

Transmitting information often requires higher data rates which, in turn, require higher frequencies. Different categories of radio wave propagation are based on natural phenomena affecting only some frequencies at different atmospheric layers. Low frequencies support worldwide communications, but only at low data rates. The higher frequencies provide higher data rates, but require expensive relay systems to provide beyond line-of-sight communications.

Channel coding and information theory is a major communication component not discussed in this thesis. This is a critical topic since it is central to modern day communications. Thus, it should be documented in a follow-up thesis at the same academic level in a similar format.

APPENDIX A - COMPANDING AND LINE ENCODING

Linear quantization, as discussed in Chapter V, produces larger signal-to-noise S/N ratios for larger input signals and lower S/N for low voltages. Achieving the same S/N ratio for small as well as large amplitude signals is accomplished through nonlinear step sizes called *nonlinear encoding* (see Figure A-1, copied from Stallings 88, p. 85).

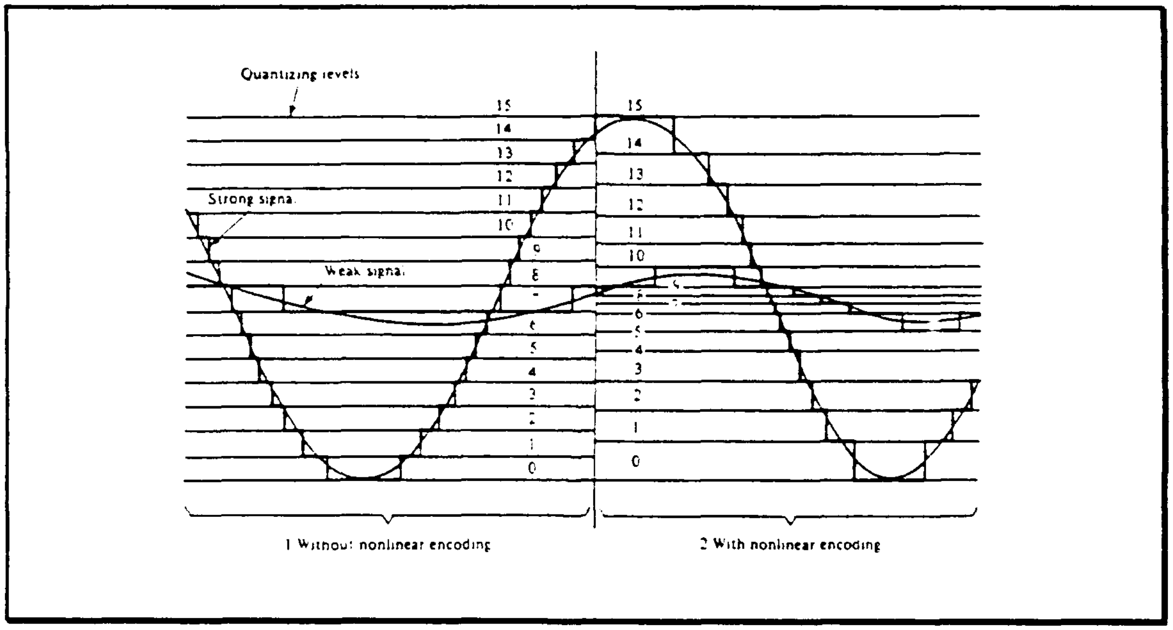

Figure A-1 Linear and Nonlinear Encoding

The same effect can be achieved by keeping the step sizes uniform but companding (compressing-expanding) the input analog signal (see Figure A-2, copied from Roden 88, p. 117). "Companding is a process that compresses the intensity range of a signal by imparting more gain to weak signals than to strong signals on input." (Stallings 88, p. 84)

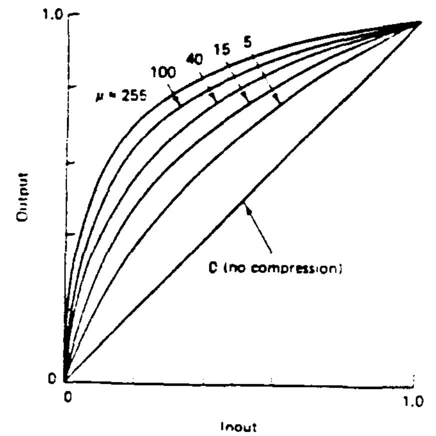

Figure A-2. Companding

Line Encoding assigns fixed voltage levels to represent 1's and 0's. Nonreturn to Zero (NRZ) is the technique shown throughout this report. Other methodologies are shown in Figure A-3 (copied from Couch 90, pp. 145, 148).

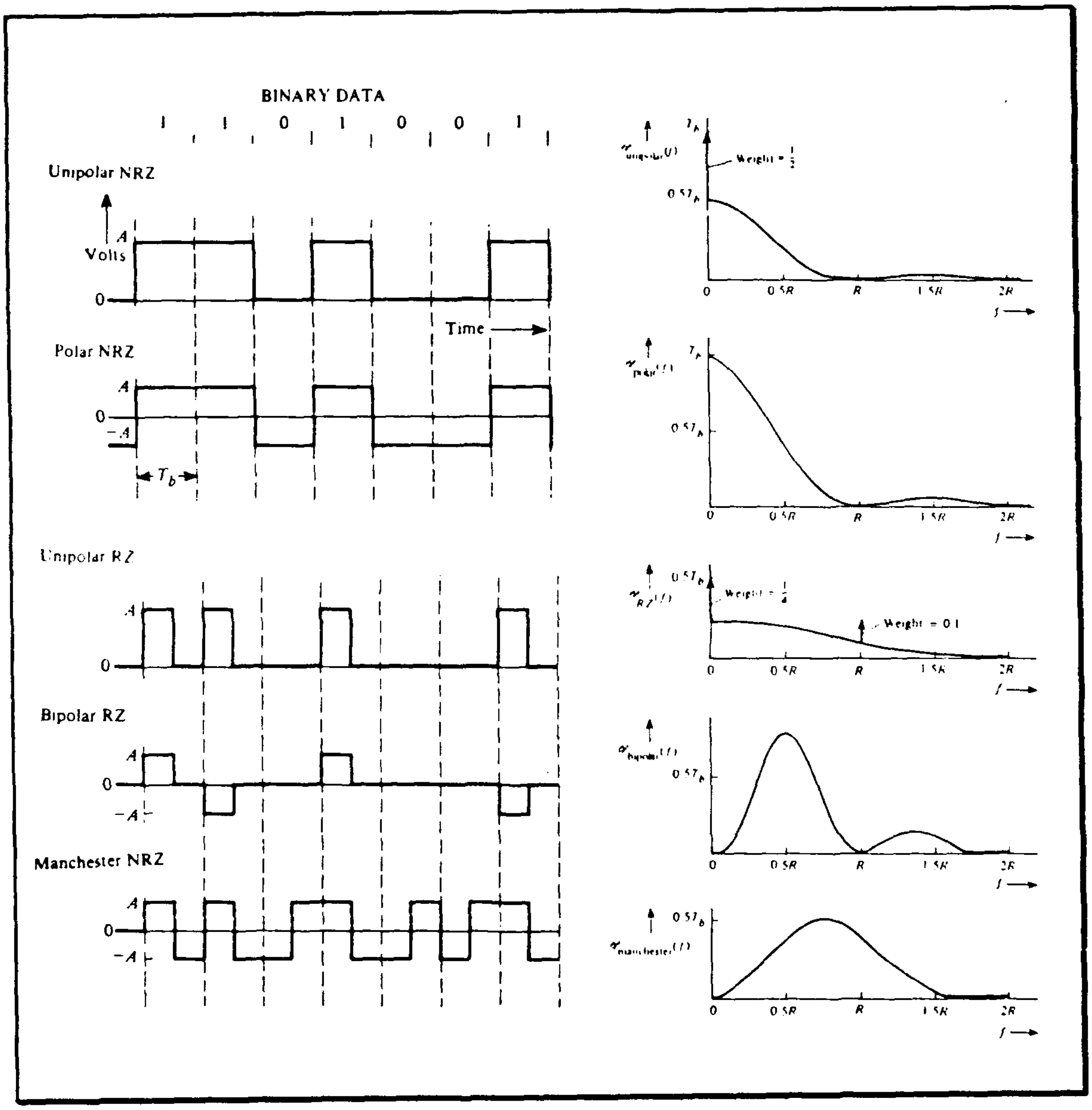

Figure A-3. Line Encoding Formats

APPENDIX B - CODE DIVISION MULTIPLEXING

Code division multiplexing (CDM) is used with a special class of signals called *direct sequence spread spectrum* (DS-SS). Spread spectrum signals are characterized by the transmitted signal occupying a much wider frequency band than is necessary for transmission. For example, a spread spectrum system takes a baseband voice signal that is only a few kilohertz wide and distributes it across a band that may be a megahertz wide. There are three general techniques used to spread a signal (Dixon 84, p. 4):

1. Modulation of a carrier by a digital code sequence whose bit rate is much higher than the information signal bandwidth. Such systems are called *direct sequence* modulated systems.

2. Carrier frequency shifting in discrete increments in a pattern dictated by a code sequence. These are called *frequency hoppers*. The transmitter jumps from frequency to frequency within some predetermined set. The order of frequency usage is determined by a code sequence.

3. Pulsed-FM or *chirp modulation* in which a carrier is swept over a wide band during a given pulse interval.

Spreading a signal's bandwidth more than necessary seems like a wasteful thing to do since the frequency spectrum is already crowded. Nevertheless, some of the reasons for spreading a signal are (Dixon 84, p. 7):

- Selective addressing capability
- Code division multiplexing, possible for multiple access
- Low-density power spectra for signal hiding
- Message screening from eavesdroppers
- High resolution ranging
- Interference rejection

Only the direct sequence spread spectrum method is shown in Figure B-1. The baseband signal of Figure B-1 is added to a spreading code modulo 2 before it is sent to the modulator. The spreading code[1] shown is only 10 times faster than the information signal. In reality it may be a 1000 times faster than the information signal. The chip rate is equivalent to the code generator clock rate and corresponds to the smallest bit duration of the spreading code. (Dixon 84, p. 13)

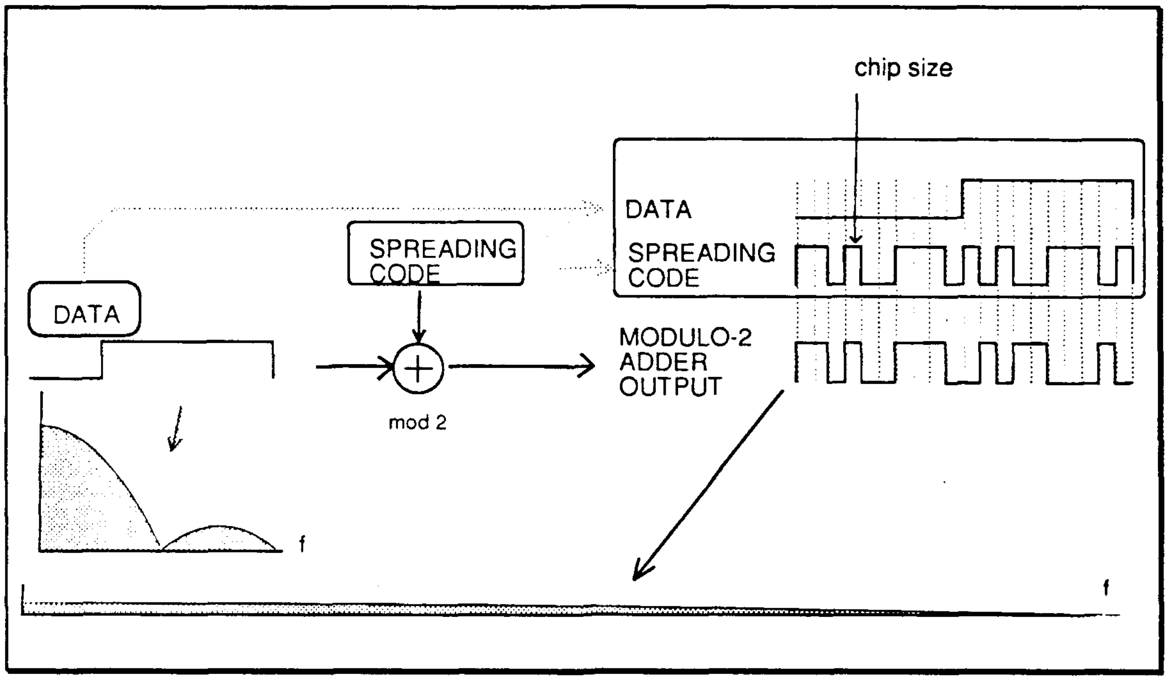

Figure B-1. Direct Sequence Spread Spectrum

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[1] The spreading code itself is generated separately from the information and may use a process called linear recursive sequence which also uses modulo-2 addition.. Therefore, don't confuse the linear recursive sequence modulo 2 with the modulo 2 in the data and information addition when discussed in textbooks.

An alternative method for spreading a signal is shown in Figure B-2. In this method the spreading code modulates the data modulated carrier. (Stremler 90, p. 629)

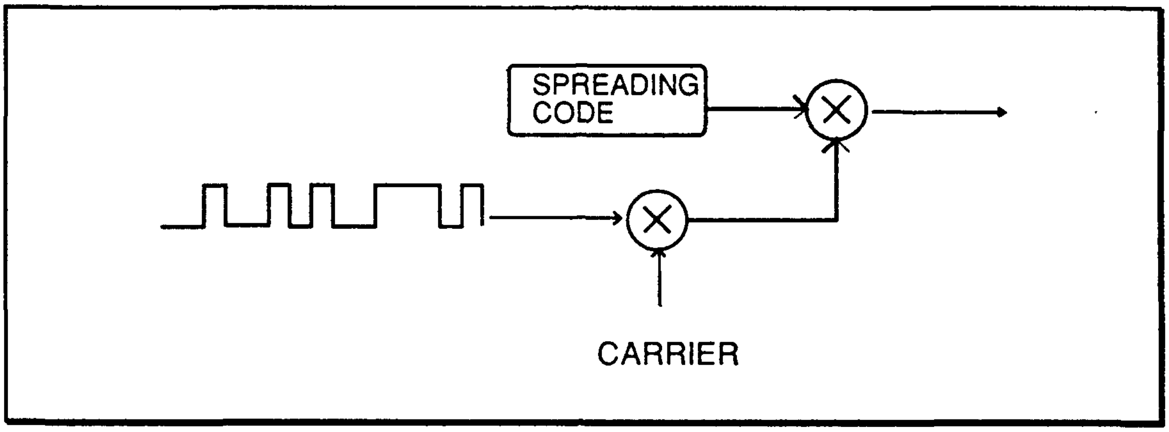

Figure B-2. Direct Sequence Spread Spectrum; Alternative Method

Since each code is unique, only those receivers that have the spreading code are able to despread the signal. Signals that do not contain the code are in turn spread by the receiver. Therefore, multiple DS-SS signals (see Figure B-3) can be transmitted at the same frequency with minimal interference. (Dixon 84, p. 11)

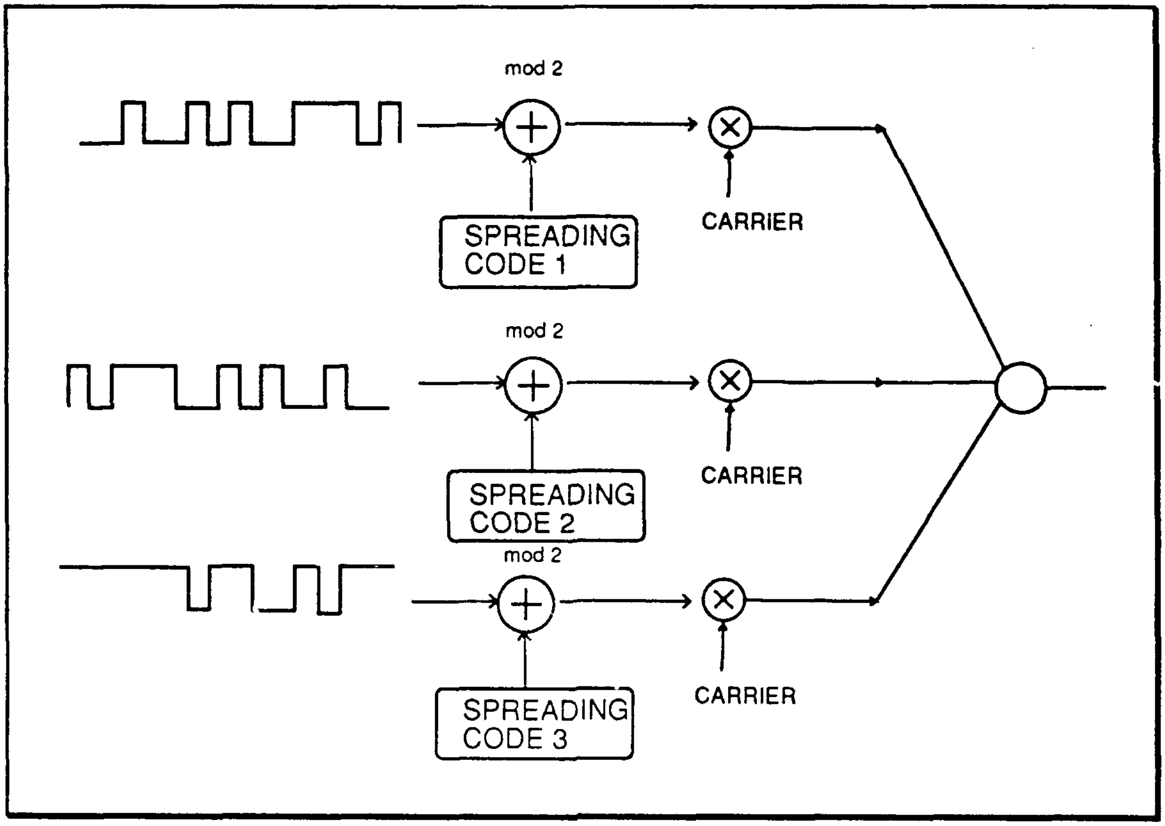

Figure B-3. Code Division Mutiplexing

Figure B-4 is included for comparing CDM to the FDM and TDM multiplexing methods discussed in Chapter V. The digital data input to the CDM could be the output of a Pulse Code Modulated signal, as shown in Chapter V. Note the increase in bandwidth from FDM to TDM to CDM.

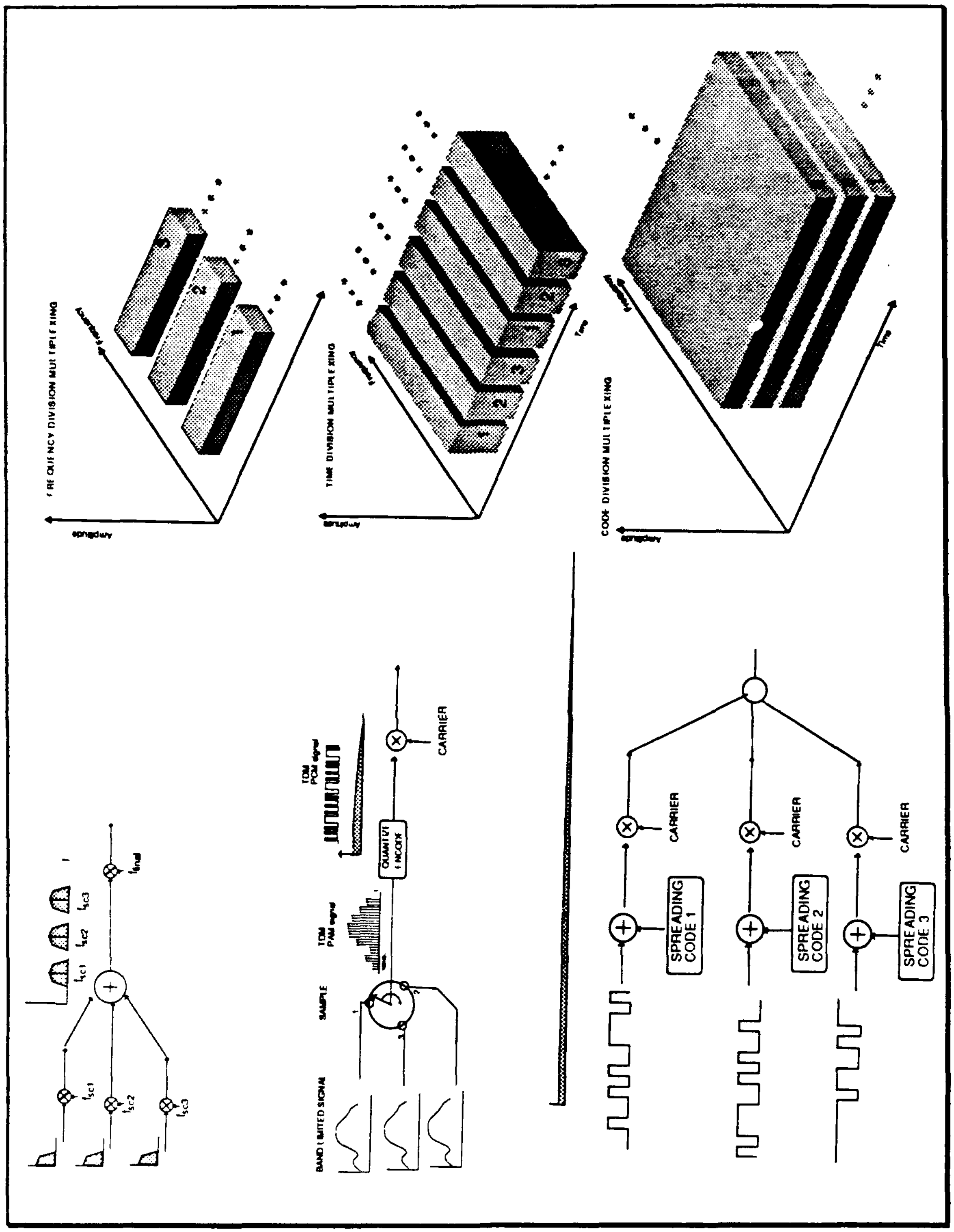

Figure B-4. Multiplexing Compared

APPENDIX C - MODULATION/DEMODULATION

Modifications are made to the binary (0,1) techniques allowing more than one bit to be transmitted for the same signaling state. This results in less bandwidth required for the same data rate. Alternatively, one could increase the data rate while maintaining the same bandwidth. The efficiency increase is accomplished by having more than one bit represented by a signaling state (amplitude, phase, or frequency). This type of transmission is known as M-ary, in contrast to binary[1].

For example, M-ary PSK, where M=4, is better known as QPSK. It generates four discrete phases, and two data bits can be represented for each phase (00,01,10,11). With eight phase (8-PSK), three data bits are sent for each signal state (000,001,010,011,100,101,110,111).

There are a myriad of modulation variations and it is very easy for you to get lost in the "alphabet soup" terminology (BPSK, PRK, QPSK, OQPSK). Figure C-1 shows the relationships between the various types. Each block contains the modulation type and the basic operation. The various techniques were developed to increase data rates for a given bandwidth, reduce the effects of abrupt phase shifts, or to improve performance in nonlinear channels. Nonlinear channels distort the signal. Therefore, it is beneficial to have a waveform with constant amplitude in order to reduce these distorting effects.

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[1] Binary can be thought as M-ary transmission where M = 2.

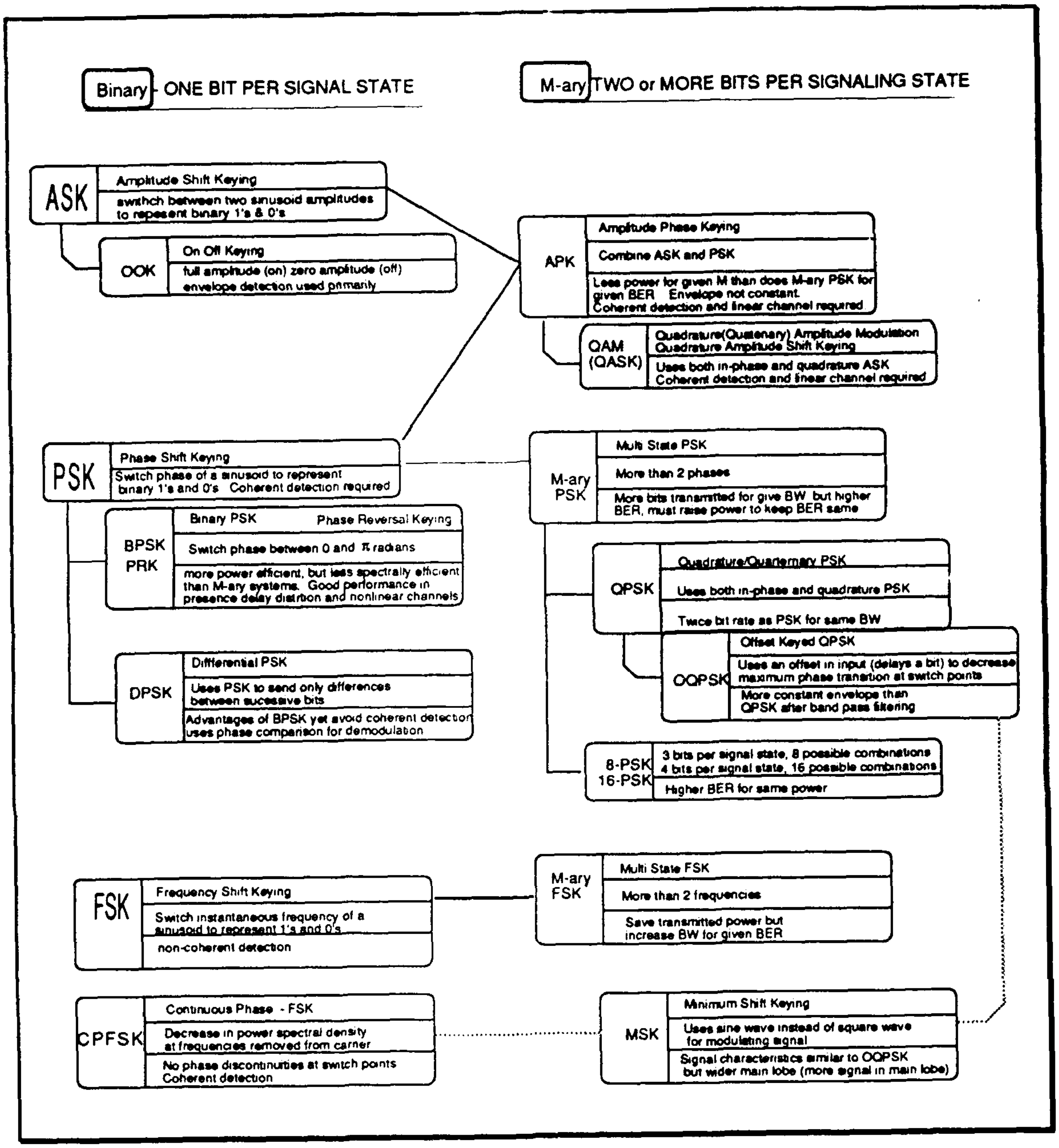

Figure C-1. Modulation Relationship

The following discussion should shed some light on the need for more spectrally efficient techniques with constant envelopes. Transmitted power and channel bandwidth are the primary resources for any communication system. In many cases, one resource may be more precious than the other. For example, satellites are power limited and error-correcting codes may be used to reduce BER in order to save power but at the expense of bandwidth. Telephones, on the other hand, are limited to the 3.4 kHz bandwidth. (Pasupathy 79, p. 14)

Although many communication systems have these constraints, the author focuses briefly on communication satellites for the following reason. Our nation is heavily reliant on satellites for modern communication needs. Most new satellites use digital modulation and they provide an encapsulated view of the need for power efficient[2] and spectrally efficient[3] modulation techniques.

> During the late 1970s and early 1980 most operational satellite and terrestrial line-of-sight microwave systems used analog FM techniques. However, the trend in new development is such that the overwhelming majority of new satellite systems employ digital methods. (Feher 83, p. xviii)

Satellites continue to be popular methods of communications because they offer the most cost-effective method of signal transmission in many applications. Almost all commercially operational communications satellites operate in the 6 GHz (uplink) and 4 GHz (downlink) frequency band.[4] Since this frequency range is very crowded, various methods are used to impro satellite bandwidth use. These

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2 Power efficiency is expressed in terms of the carrier-to-noise C/N ratio required to have an acceptable probability of error P(e) (or bit error rate, BER). It is also defined in terms of average received bit energy-to-noise density ratio for a given BER.

3 Spectrally efficiency (bandwidth efficiency) may be expressed in terms of bits/second/hertz (b/s/Hz). For example, if 10 Mb/s are transmitted in a 6 MHz wide channel, the spectral efficiency is 10 Mb/s per 6 MHz, or 1.67 b/s/Hz. (Feher 83, p. 78)

4 This frequency range is used since it falls in a "window" where external noise is a minimum and atmospheric absorption isn't a factor yet. See Figure 7-14 for clarification.

include, narrow antenna beamwidth, orthogonal polarization[5], digital modulation techniques (higher spectral efficiency), multiplexing, and digital speech interpolation (DSI)[6]. (Feher 83, p. 17) The student desiring more information should read the texts by Feher 83, Sklar 88, or Couch 90.

Changing the subject now to demodulation (the receiver end). The received signal is corrupted by noise and a decision needs to be made on whether a 1 or a 0 was transmitted. Optimum results are obtained through use of correlation receivers or matched filters.[7] Correlation receivers and matched filters correlate the received signal with both 0 and 1 decision bit levels to determine whether a 0 or 1 was transmitted. The bit that provides the best match (largest correlation) is chosen. (Sklar 88, p. 134) Explaining matched filters and correlation can be lengthy and the explanation involves advanced mathematics. Nevertheless, the end results are dramatic and are shown in Figure C-2 (adapted from Stremler 90, p. 182).

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[5] Orthogonal means perpendicular, independent of one anther. Horizontal and vertical polarizations are examples of orthogonal polarization .

[6] DSI takes advantage of periods of inactivity between calls; pauses, hesitations, and intervals of silence (Feher 83, p. 65).

[7] Matched filters are beyond the scope of this thesis. However, they are mentioned so one knows where in the communications process matched filters are used.

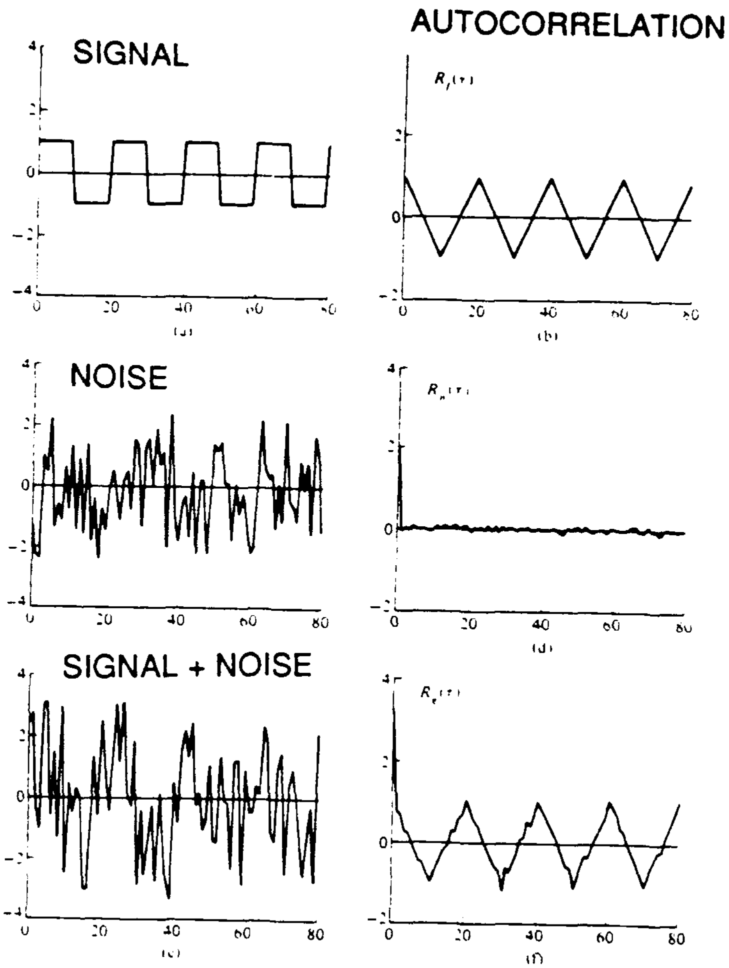

Figure C-2. Information Recovered via Correlation

APPENDIX D - TRANSMISSION IMPAIRMENTS

A. FIBER OPTICS

Three major performance criteria affect choice of fibers: signal losses, ease of connection, and bandwidth (Keck 85, p. 17). Signal losses and bandwidth are discussed below.

Absorption and scattering losses are strongly dependent on wavelength. Scattering can be decreased by increasing the wavelength. However, this decrease is offset by an increase in material absorption. Material absorption is caused by *intrinsic* and *extrinsic* properties. Intrinsic losses are due to the glass properties themselves and it is a source of ultraviolet and infrared absorption. Extrinsic loss relates to impurities in the glass, specifically the hydroxyl (OH) group in silica. Figure D-1 (copied from Senior 85, p. 68) shows a combined attenuation spectrum due to absorption and scattering[1].

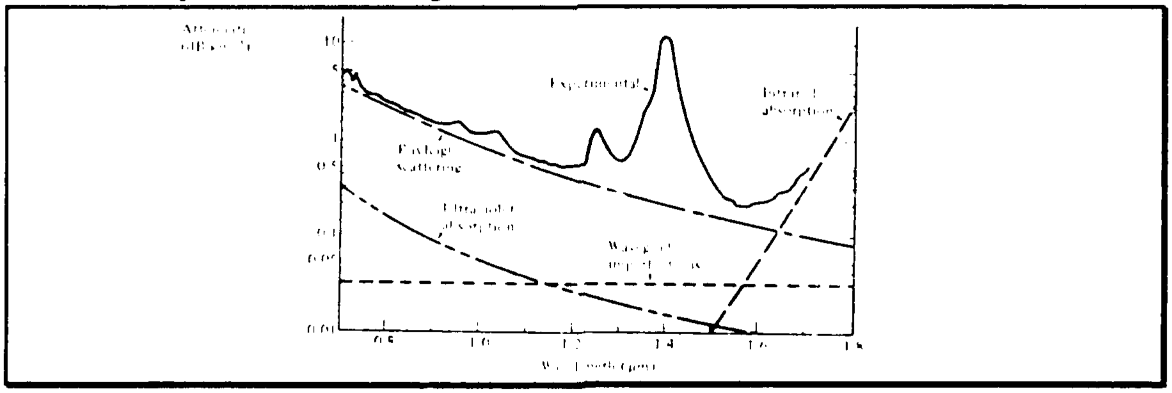

Figure D-1. Fiber Optic Attenuation

[1] Note that the wavelength scale is increasing as you read from left to right. The corresponding frequencies (not shown) would decrease. You would have to flip the fiber optic chart around in order to make it consistent with the frequency charts given in the main body of the thesis.

The primary limitation on bandwidth is *dispersion* (pulse broadening), as shown in Figure D-2 (copied from Senior 85, p. 77). It is caused by *intermodal* and *intramodal* phenomenon. Intermodal dispersion is dependent on the type of fiber used. Fibers are classified as multimode or *single mode*. Multimode fibers are relatively thick (50-125 microns). This wide diameter allows the light to enter at different angles and propagate along different paths, setting up different modes of propagation; hence, the name *multimode*. Intermodal dispersion is also called *mode* or *modal* dispersion.

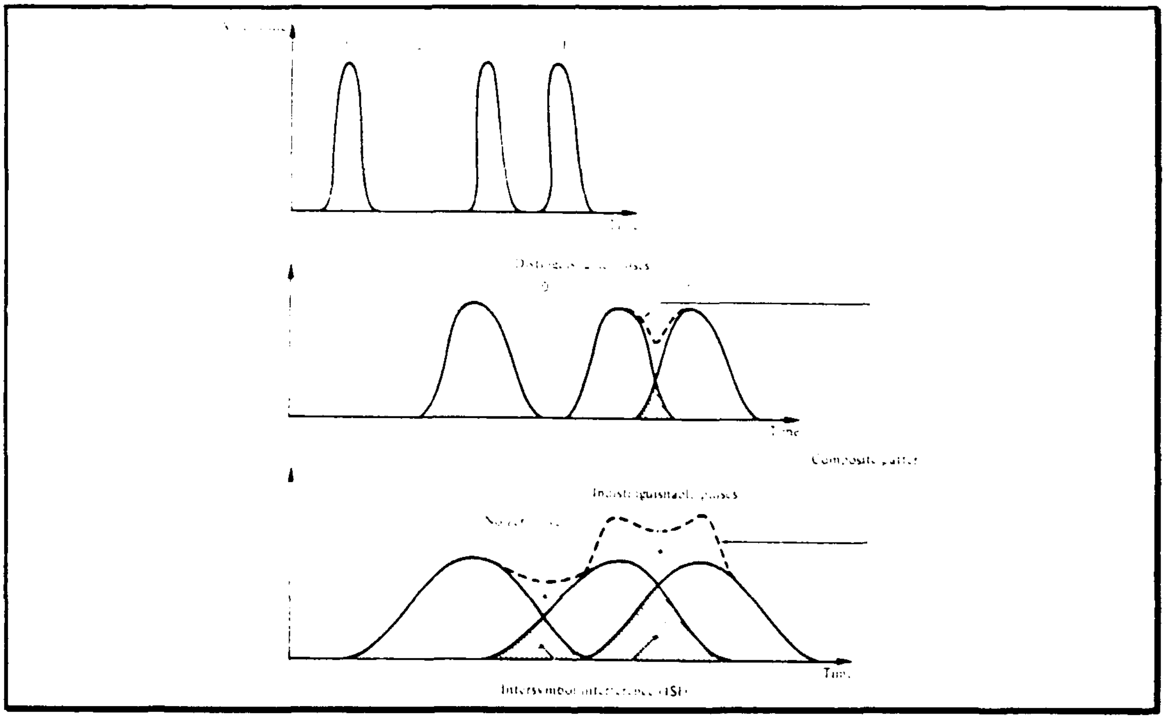

Figure D-2. Dispersion Causing Intersymbol Interference

The performance of a multimode fiber can be improved by changing the refractive index of the core so the light rays that stray from the core's center propagate faster and turn back toward the center. This is called *multimode graded* *index* fiber. Single-mode fiber is so thin it only allows propagation along the core's center; therefore, intermodal dispersion is negligible. Single-mode fibers support the highest bandwidth, but they are more difficult to manufacture and connect due to the small size involved. Figure D-3 (copied from Senior 85, p. 79) shows the difference in size and dispersion qualities for the three fiber types.

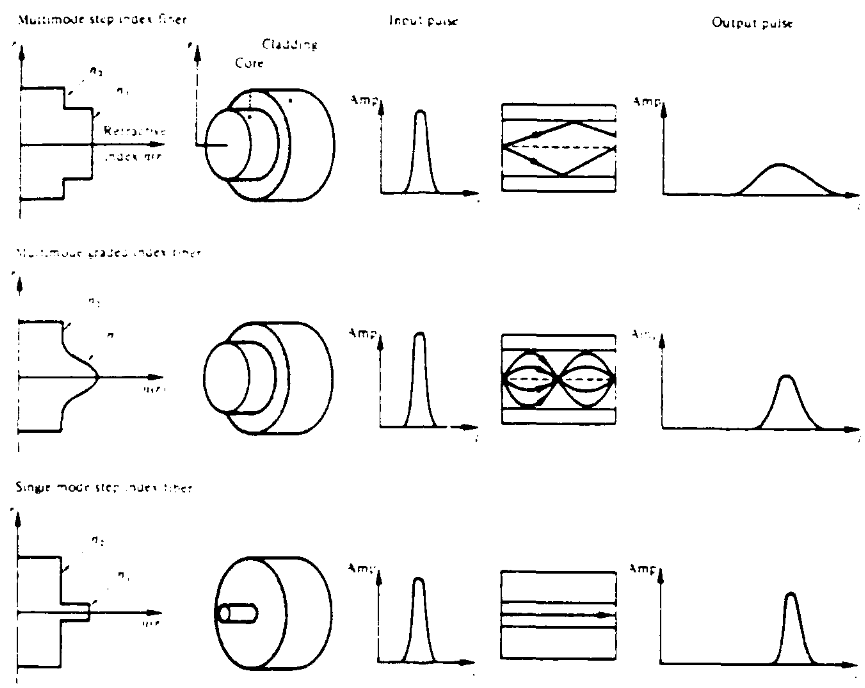

Figure D-3. Three Types of Fiber Optics

There is another type of dispersion that affects all three fiber types called *intramodal* dispersion. Intramodal dispersion is caused by two factors. The first is the variation of the refractive index of glass with wavelength and is called *material* or *chromatic* dispersion. The second factor results from some of the light straying from the core, this factor is called *waveguide* dispersion. Figure D-4 (copied from Keck 85, p. 20) shows the dispersion caused by material and wave guide for single-mode step index. Note that the minimum occurs around 1.3 microns.

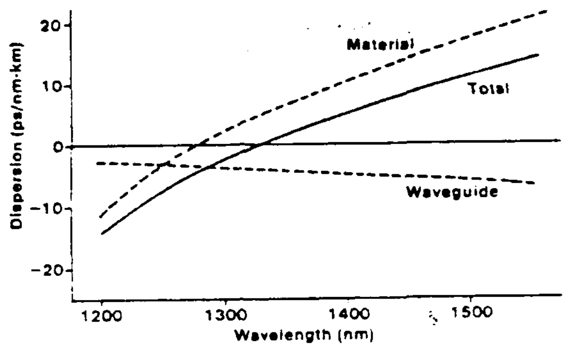

Figure D-4. Material and Waveguide Dispersion

Figure D-5 outlines the relationships between the various fiber optic impairments.

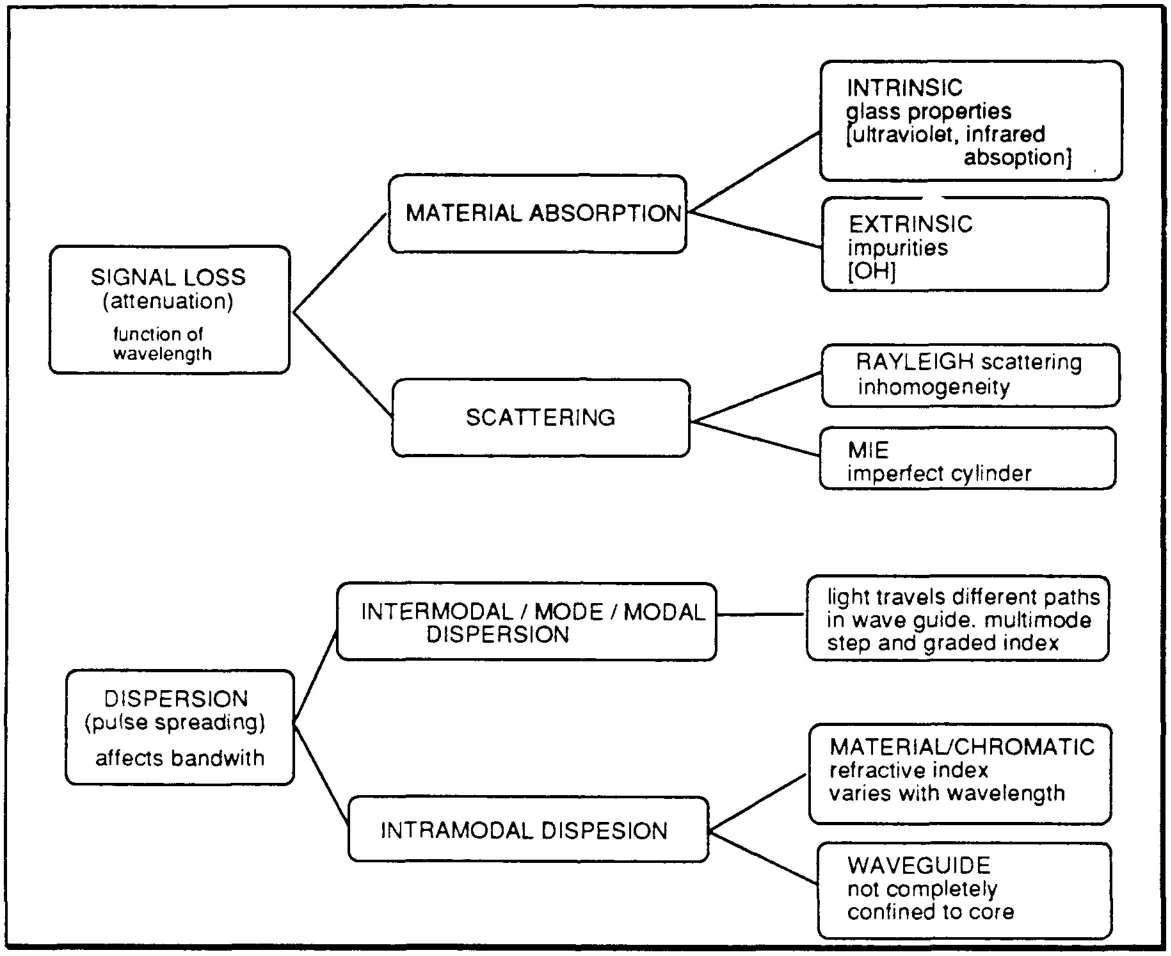

Figure D-5. Fiber Optic Impairments

As with any system, there is no easy "cookbook" answer. There are always tradeoffs between bandwidth, distance, number of repeaters, fibers, and cost. Single mode offers the longest transmission at a higher data rate with fewer repeaters. However, single mode fiber itself is more expensive and requires an ILD. On the other hand, multimode fibers are cheaper and support inexpensive LED sources.

B. IMPAIRMENT CATEGORIES

1. Attenuation

Attenuation is a reduction in signal strength caused by free space loss or absorption.

a. Free Space Loss

Free space loss occurs with unguided mediums (RF transmissions) and is caused by the spreading of the electromagnetic energy as it propagates through the atmosphere.

b. Absorption

Absorption occurs in both unguided and guided medias. The attenuation varies logarithmically with distance and is usually expressed in db/km. The atmosphere's water vapor and oxygen molecules significantly absorb RF energy above 10 GHz and limits the applications for long haul transmissions. Below 10 GHz the atmospheric affects are negligible.

2. Distortion

The *distortion* form of impairment distorts the shape of the waveform through attenuation or delay. It is associated with guided media.

a. Attenuation distortion

Copper paths *attenuate* higher frequencies at a logarithmic rate. Equalization is used to equalize the attenuation across a band of frequencies. Loading coils are used in voice-grade telephone lines to change the electrical properties of the line and smooth out attenuation effects. Amplifiers can also be used to amplify the higher frequencies. (Stallings 88, p. 42)

b. Delay distortion

Delay distortion is caused by a signal's velocity of propagation varying as a function of frequency in a guided media. The process results in different frequency components arriving at the destination at different times. This activity causes a smearing of one bit into the next, and the phenomenon is called *intersymbol interference*-(ISI); it is a major factor in determining the maximum bit rate. (Stallings 88, p. 42)

Delay distortion is similar to intramodal and intermodal dispersion (which cause a spreading of light pulses in fiber optics). In intramodal dispersion the refractive index of light varies with wavelength. In intermodal dispersion the light travels along different paths within the fiber.

One should not confuse ISI with crosstalk. *Crosstalk* is caused by overlapping frequencies from different sources and occurs when frequency division multiplexed signals do not have a wide enough guard band.

3. Multipath

Multipath is associated with RF transmission and occurs when part of the wave front is reflected off the ground, buildings, mountains, the moon, or other physical objects, and arrives later than the direct path. A familiar type of multipath interference causes "ghosting" on television.

If there are multiple multipaths and the received signal has noise-like characteristics, then it is termed *diffuse*; if there are only one or two strong reflected signals, then it is termed *specular* and results in *fading*. Fading is caused by the constructive and destructive interference of multiple received signals. (Zeimer 76, p. 5)

4. Noise

Noise is one of the fundamental limitations of a communications system. It consists of random/unpredictable electrical signals from sources internal and external to the system. Noise can be reduced through filtering, receiving antenna directivity, and use of low noise front ends, but cannot be totally eliminated. (Zeimer 76, p. 5; Miller 88, p. 5)

a. Noise External to receiver

Noise external to the system comes from both natural and man-made sources The atmosphere is a natural source and the noise is caused mainly by lightning. The effects are usually experienced below 100 MHz. The intensity of the noise is inversely proportional to frequency. (Miller 88, p. 5; Zeimer 76, p. 5)

Another natural source comes from extraterrestrial objects (hot bodies). This noise is sometimes called *galactic* noise or *space* noise. Space noise is evenly divided into *solar* and *cosmic* noise. Solar noise originates from our sun, and cosmic noise results from the background radiation along the galactic plane, or other stars and celestial objects. Space noise occurs from 8 MHz to 1.5 GHz. (Miller 88, p. 5; Stanley 82, p. 40)

Man-made sources are grouped into two categories. One is radio frequency interference (RFI) which is caused by other communication transmitters operating at the same frequency. The other is electromagnetic interference (EMI) which is often impulsive in nature and an irritation on voice grade channels. It is a serious source of error in digital signals. EMI results from many sources including the following: commutators in electrical motors, heavy inductive loads as in elevator motors, auto ignition systems, voltage surges, florescent lights, and high-voltage power line corona discharge. The signals can be induced onto copper wires or they enter the system through the antenna. Shielding the wire, as in coax, can help mitigate the effects. (Miller 88, p. 5; Zeimer 76, p. 5) Electromagnetic pulse (EMP) is associated with nuclear explosions. Fiber optic cable is immune to the electromagnetic effects of noise.

b. Internal

Internal noise is caused by the receiving equipment and it is generated by the equipment components (resistors, amplifiers, active solid state devices). The major noise source occurs at the first stage of amplification. Noise introduced at follow-on stages is negligible compared to the first stage. Therefore this first stage needs to be built with very low noise characteristics; hence the name, *low noise amplifier* (LNA). (Miller 88, p. 5)

There are two primary noise sources in electronic circuits: thermal and transistor noise. Both have several names attached to them. *Thermal noise* (also called *Johnson noise*) is caused by thermal interactions between free electrons and vibrating ions in a conductor. The frequency content is spread equally throughout the useable spectrum and results in *white noise*. The noise's amplitude has a normal distribution (in the statistical sense) and is added to the signal; hence additive white Gaussian noise. The other primary noise source is transistor noise or shot noise. Transistor noise is caused by the discrete particle nature of the current carriers of all the semiconductors. (Miller 88, p. 5)

Low frequency effects are called *excess noise* and occur at frequencies below 1 kHz. Excess noise is also called flicker noise, pink noise or 1/f noise. It is present in bipolar junction transistors and field effect transistors (FET). (Miller 88, p. 5)

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