TOPIC 3

BASIC COMMUNICATIONS THEORY

INTRODUCTION

A brief definition of *radio* is the transmission of signals through space by electromagnetic waves. We use this term to refer to the transmission of intelligence codes and sound signals, although television and radio also depend on electromagnetic waves.

Radio equipment has three broad categories:

1. *Transmitting* equipment provides generation, amplification, and modulation of the transmitted signal.

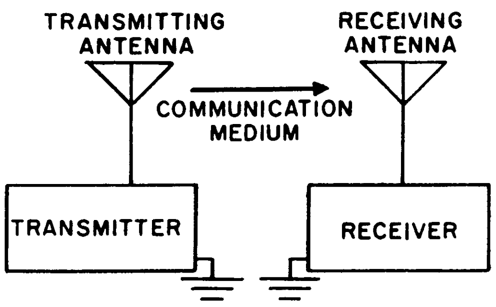

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Figure 3-1.—Basic radio communications system.

2. *Receiving* equipment receives radio waves and converts them into modulation signals.

3. *Terminal* equipment converts the modulation into the original intelligence. (Terminal equipment is used primarily where coded transmissions convert the modulated signal into the original intelligence.)

A basic radio communications system may consist of only a transmitter and a receiver, connected by the medium through which the electromagnetic waves travel. See figure 3-1. The transmitter consists of an oscillator, the necessary radio frequency amplifiers, and any stages required to place the audio intelligence on the Radio Frequency (RF) signal (modulator). The electromagnetic variations are propagated through the medium (space) from the transmitting antenna to the receiving antenna. The receiving antenna then converts that portion of the transmitted electromagnetic energy received by the antenna into a flow of alternating RF currents. The receiver either converts these current changes into the intelligence contained in the transmission during voice communications or provides these current changes to terminal equipment (such as teletype or facsimile) for conversion to the original intelligence.

Figure 3-2 shows the portion of the overall frequency spectrum used for communications.

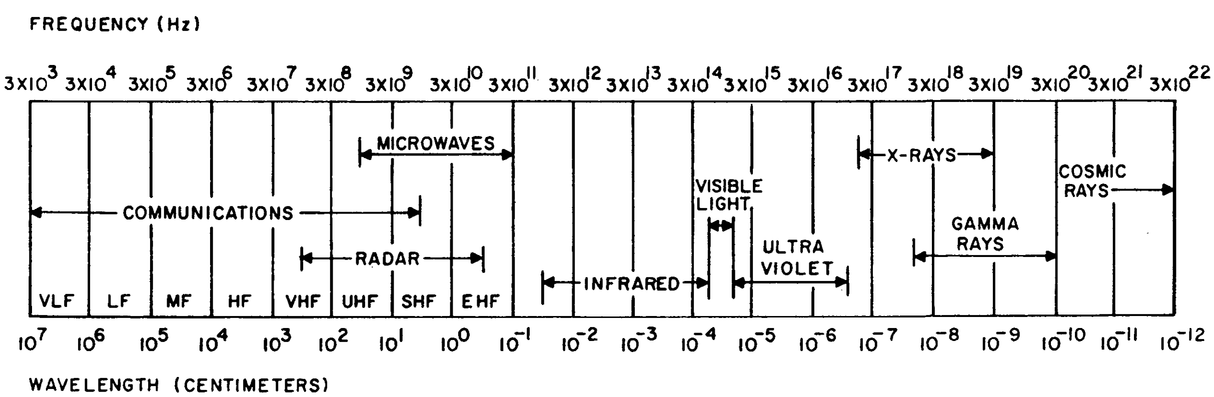

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Figure 3-2.—Frequency spectrum.

Table 3-1.—Frequency Bands

ABBREVIATIONS	FREQUENCY BAND	APPLICATION
VLF	VERY LOW FREQUENCY	SHORE BASED COMMUNICATIONS, EXPERIMENTAL
LF	LOW FREQUENCY	SHORE BASED COMMUNICATIONS, NAVIGATION
MF	MEDIUM FREQUENCY	COMMERCIAL BROADCAST BAND, 550 kHz TO 1700 kHz. COMMUNICATIONS ON EITHER SIDE OF BROADCAST BAND.
HF	HIGH FREQUENCY	SHIP AND SHORE LONG RANGE COMMUNICATIONS.
VHF	VERY HIGH FREQUENCY	COMMUNICATIONS, NAVIGATION
UHF	ULTRA HIGH FREQUENCY	LINE OF SIGHT COMMUNICATIONS TO 400 MHz. ABOVE THIS FREQUENCY, RADAR AND SPECIAL EQUIPMENTS.
SHF	SUPER HIGH FREQUENCY	RADAR AND SPECIAL EQUIPMENTS
EHF	EXTREMELY HIGH FREQUENCY	RADAR AND SPECIAL EQUIPMENTS.

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Table 3-1 lists the frequency bands in the lower end of the spectrum and their applications. At the present time, these are the frequencies we use for communications. A description of a few of the general characteristics of some of these frequency bands are described in the following paragraphs.

The Very-Low-Frequency (VLF) and Low-Frequency (LF) bands require great power and long antennas for efficient transmission. Therefore, the Navy normally uses these bands for transmissions emanating from shore. (The antenna length varies inversely with the frequency.)

Because the commercial broadcast band extends from 550 kHz to 1700 kHz, the Navy uses only the upper and lower ends of the Medium-Frequency (MF) band. We use the High-Frequency (HF) band for long-range shipboard radio communications. Therefore, the majority of shipboard transmitters and receivers operate in the HF band.

Commercial television uses a large portion of the lower end of the VHF band. The Navy also uses this part of the VHF band for amphibious operations and in special instances. The Navy makes extensive use of the upper portion of the VHF band (225 MHz to 300 MHz) and the lower portion of the Ultrahigh-Frequency (UHF) band (300 MHz to 400 MHz) for short-range and aircraft communications. We normally use the frequencies above 400 MHz in the UHF band through the Superhigh-Frequency (SHF) and Extremely High-Frequency (EHF) bands for radar and special equipment.

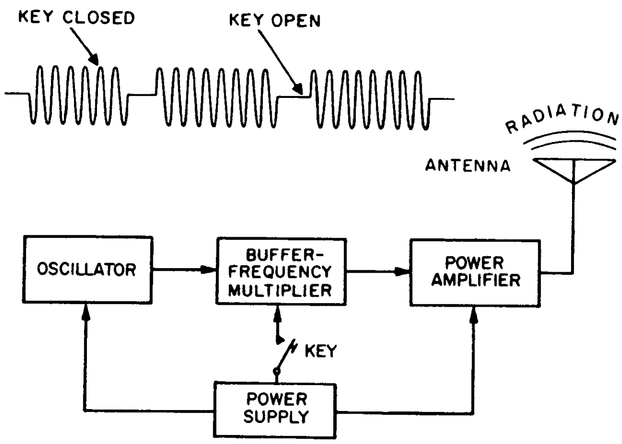

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Figure 3-3.—Continuous-wave transmitter.

CONTINUOUS WAVE TRANSMISSION

Continuous Wave (CW) is one of the oldest and least complicated forms of radio communications. The system consists of little more than a transmitter and a receiver, connected to facilitate their control from a central location.

The CW transmitter is keyed (turned on and off) to produce long or short pulses that correspond to the dots and dashes of the Morse code characters. The four essential components of the transmitter are

1. a generator of RF oscillations,
2. a means of amplifying and multiplying the frequencies of these oscillations,
3. a method of keying the RF output in accordance with the code to be transmitted, and
4. a power supply to provide the operating voltage to the various electron tubes and transistors.

Although not physically a part of the transmitter, an antenna radiates the keyed output radio wave of the transmitter. See figure 3-3.

MODULATION

Modulation is the process of varying some characteristics of a periodic wave with an external signal. The Audio Frequency (AF) spectrum (15 to 20,000 Hz) contains the voice frequencies (about 15 to 3000 Hz). In naval communications, the terms *audio*, *voice*, and *analog communications* are sometimes used interchangeably. The audio signal is impressed upon the RF carrier because it is impractical to transmit frequencies in the audio range. This is due, in part, to the excessive wavelength of the audio signal. As you can see in figure 3-2, the wavelength at 3000 Hz is 10,000,000 cm. The physical size of circuit components at these frequencies is too large to be practical.

There are three characteristics of the carrier wave that may be varied at an external signal rate: amplitude, frequency, and phase.

AMPLITUDE MODULATION (AM)

Amplitude modulation is the process of combining audio frequency and radio frequency signals to cause the amplitude of the radio frequency waves to vary at an audio frequency rate. We can accomplish amplitude modulation by removing the key and modifying the CW transmitter so that the audio output from a microphone (and any necessary amplifiers) is impressed on the carrier frequency.

FREQUENCY MODULATION (FM)

Frequency modulation is the process of combining audio and carrier signals to cause the frequency of the carrier waves to vary at an audio rate, while the amplitude of the carrier waves remains essentially constant. The carrier frequency can be varied a small amount on either side of its average or assigned frequency by means of the audio frequency-modulating signal.

PHASE MODULATION (PM)

Phase modulation and frequency modulation are essentially the same—the difference lies in the physical method of accomplishing the frequency shift in the transmitter. FM receivers can receive, and both FM and PM are commonly referred to as FM.

RECEIVERS

An AM receiver processes AM signals received by its antenna, and outputs a reproduction of the original signal that modulated the RF carrier at the transmitter. The signal can then be applied to a reproducing device (such as a loudspeaker) or a terminal device (such as a teletypewriter). Actual AM receivers vary widely in complexity. Some are very simple, while others contain a relatively large number of complex circuits.

Whatever its degree of sophistication, a receiver must perform certain basic functions to be useful. These functions, in order of their performance, are:

1. **Reception**: Reception occurs when a transmitted electromagnetic wave passes through the receiver antenna, inducing a voltage in the antenna.
2. **Selection**: Selection is the ability to select a particular station's frequency from all other station frequencies appearing at the receiver's antenna.
3. **Detection**: Detection is the action of a detector circuit separating the low (audio) frequency intelligence from the high (radio) frequency carrier.
4. **Reproduction**: Reproduction is the action of converting the electrical signals to sound waves that the ear can interpret as speech or music.

RECEIVER CHARACTERISTICS

Receiver characteristic measurements are useful in determining operational conditions and as an aid for comparison to other units. Important receiver characteristics are sensitivity, noise, selectivity, and fidelity. The following paragraphs discuss these characteristics.

Sensitivity

The ability of a receiver to reproduce very weak signals is a function of the receiver's sensitivity. The weaker a signal can be applied to a receiver and still produce a certain value of signal output, the better that receiver's sensitivity rating. Sensitivity of a receiver is measured under standardized conditions. It is expressed in terms of the signal voltage (usually in microvolts) that must be applied to the antenna input terminals to give an established level of the output. The output may be an alternate current (ac) or a direct current (dc) voltage measured at the detector output, or it may be a power measurement at the loudspeaker or headphone terminals.

Noise

All receivers generate a certain amount of noise that must be taken into account. Noise is a limiting factor on the minimum usable signal that the receiver can process and still deliver a usable output. Therefore, the measurement is made by determining the amplitude of the signal at the receiver input required to give a signal-plus-noise output at a predetermined ratio above the static noise output of the receiver.

Selectivity

Selectivity is the degree of distinction made by the receiver between the desired signal and unwanted signals. The better the receiver's ability to exclude unwanted signals, the better its selectivity. The degree of selection is determined by the sharpness of resonance to which the frequency-determining circuits have been engineered and tuned. Measurement of selectivity is usually by a series of sensitivity readings in which the input signal is stepped along a band of frequencies above and below resonance of the receiver's circuit (for example, 100 kHz below to 100 kHz above the tuned frequency). As the frequency of the tuned receiver approaches, the input level required to maintain a given output level will fall. As the tuned frequency is passed, the required input level will rise. Input voltage levels are then plotted against frequency. The steepness of the curve at the tuned frequency indicates the selectivity of the receiver.

Fidelity

The fidelity of a receiver is its ability to accurately reproduce, in its output, the signal that appears at its input. In general, the broader the band passed by frequency selection circuits, the greater the fidelity. Fidelity may be measured by modulating an input frequency with a series of audio frequencies, and then plotting the output measurements at each step against the audio-input frequencies. The resulting curve will show the limits of reproduction.

Remember that good selectivity requires that a receiver pass a narrow frequency band. Good fidelity, on the other hand, requires that the receiver pass a broader band to amplify the outermost frequencies of the sidebands. Therefore, receivers in general use are a compromise between good selectivity and high fidelity.

SUPERHETERODYNE AM RECEIVER

The superheterodyne AM receiver was developed to overcome the disadvantages of earlier types of receivers. The essential difference is in the amplifier stages preceding the detector stage. See figure 3-4. The Intermediate Frequency (IF) amplifier in the superheterodyne receiver is pretuned to one fixed frequency.

- The RF amplifier and the local oscillator are variable frequency, dependent upon the selected frequency.

- The frequency of the RF amplifier and the local oscillation varies with the selected input frequency.

The IF is obtained through the principle of frequency conversion by heterodyning a signal generated in a local oscillator of the receiver with the incoming signal in a mixer stage. Thus, an incoming signal is converted to the fixed intermediate frequency, and the IF amplifier operates with uniform selectivity and sensitivity over the entire tuning range of the receiver.

Figure 3-4 shows a block diagram of a representative superheterodyne AM receiver. A superheterodyne receiver may have more than one frequency-converting stage and as many amplifiers as needed to obtain the desired power output. (The additional amplifiers are not shown in the figure.)

Heterodyning

The IF is developed by a process called *heterodyning*. This action takes place in the mixer stage (sometimes called *converter* or *first detector*). Heterodyning combines the incoming signal with the local oscillator signal. This action alters the input signal from its RF carrier frequency to an intermediate frequency more suitable for extracting intelligence from the transmitted signal. The local oscillator signal is of a constant amplitude and does not alter that intelligence.

The local oscillator is set to track with the tuning of the incoming signal to produce a frequency higher or lower than the frequency of the incoming signal by the exact amount of the fixed IF frequency. By heterodyning the incoming signal and the locally produced signal in the mixer stage, four frequencies appear at the mixer output. They are

1. the incoming RF signal,

2. the local oscillator signal,

3. the sum of the incoming RF signal and the local oscillator signal, and

4. the difference in these frequencies.

Although the sum frequency is present, it is the difference frequency to which the IF amplifier is tuned. A typical intermediate frequency for AM communication receivers is 455 kHz.

Detection

Once the IF stages have amplified the intermediate frequency to a sufficient level, it is fed to the detector (or second detector, if referring to the mixer as the first detector) to extract the modulating audio signal. The detector stage consists of a rectifying device and filter. They respond only to the amplitude variations of the IF signal to develop an output voltage varying at an audio frequency rate. The output from the detector is further amplified in the audio amplifier and is used to drive a speaker or a set of earphones.

SUPERHETERODYNE FM RECEIVER

The function of a superheterodyne FM receiver is the same as the AM superheterodyne receiver. There are certain important differences in component construction and circuit design

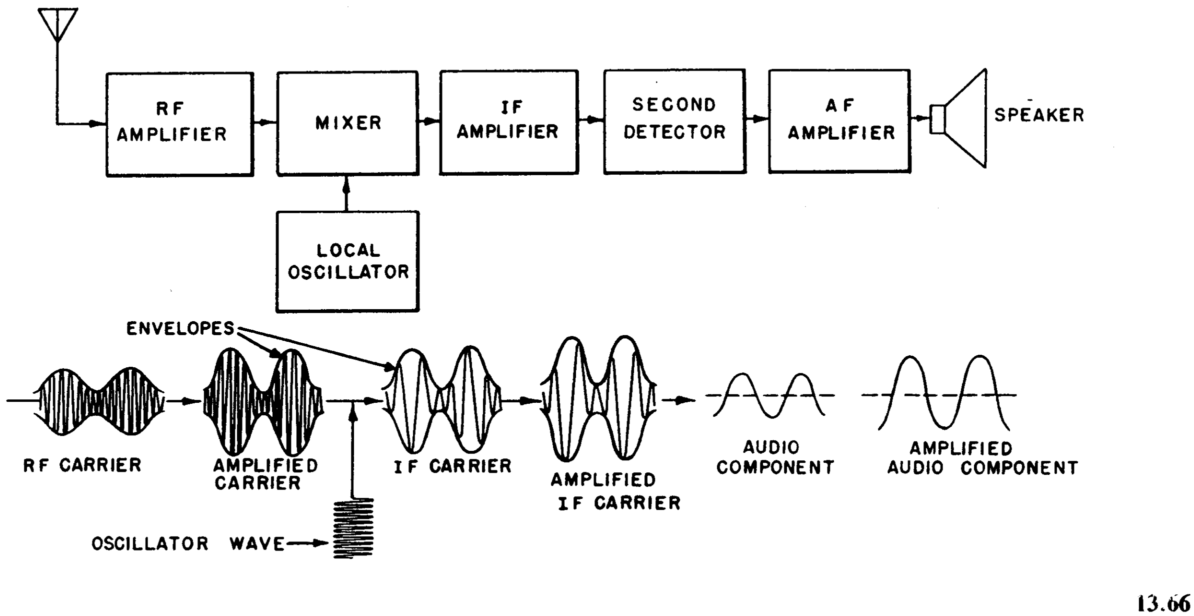

Figure 3-4.—Superheterodyne AM receiver and waveforms.

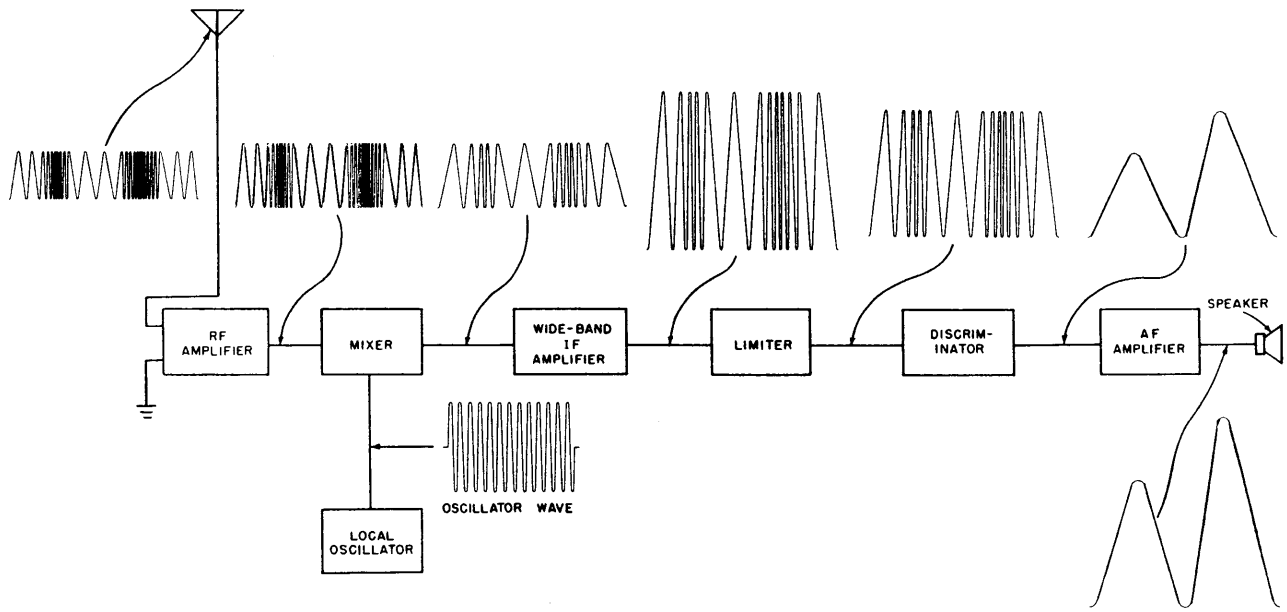

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Figure 3-5.—Block diagram of an FM receiver and waveforms.

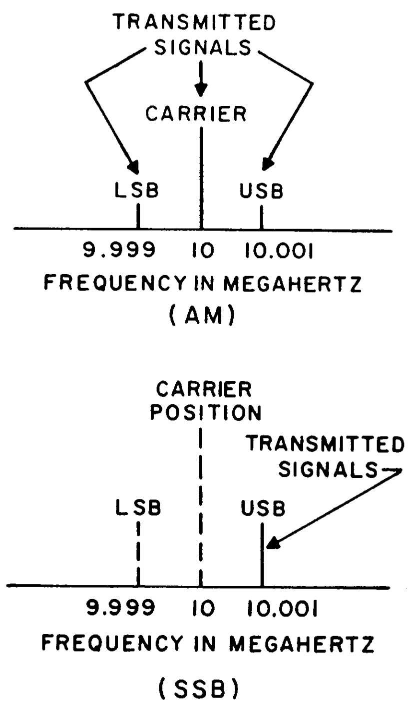

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Figure 3-6.—Comparison of AM- and SSB-transmitted signals.

because of differences in the modulating technique. The comparison of block diagrams in in figures 3-4 and 3-5 shows that in both AM and FM receivers, the amplitude of the incoming signal is increased in the RF stages. The mixer combines the incoming RF with the local oscillator RF signals to produce the intermediate frequency, which is then amplified by one or more IF amplifier stages. Note that the FM receiver has a wideband IF amplifier. Since the bandwidth for any type of modulation must be wide enough to receive and pass all the side-frequency components of the modulated signal without distortion, the IF amplifier in an FM receiver must have a broader passband than an AM receiver.

Sidebands created by FM and PM systems differ from the AM system. They occur at integral multiples of the modulating frequency on either side of the carrier wave. The AM system consists of a single set of side frequencies for each radio frequency signal that is modulated. An FM or a PM signal inherently occupies a wider band than AM, and the number of these extra sidebands that occur in FM transmission relates to the amplitude and frequency of the audio signal.

Beyond the IF stage, there is a marked difference between the two receivers. While AM demodulation involves the detection of variations in the amplitude of the signal, FM demodulation detects variations in the frequency of the signal. In FM receivers, a discriminator responds to frequency shift variations. A limiter precedes the discriminator. It limits all signals to the same amplitude level to minimize noise interference. The discriminator then extracts the audio frequency component, amplified in the AF amplifier, and used to drive the speaker. Electrically, there

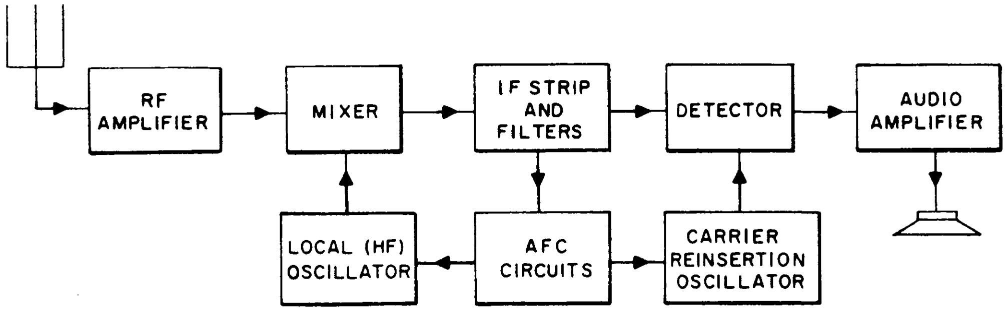

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Figure 3-7.—Basic SSB receiver.

are only two fundamental sections of the FM receiver that are different from the AM receiver: the discriminator (detector) and the accompanying limiter.

In normal reception, FM signals are totally absent of static, while AM signals are subject to cracking noises and whistles. FM followed AM in development and had the advantage of operating at the higher frequency where there is a greater spectrum. FM signals provide a much more realistic reproduction of sound because of an increased number of sidebands.

The major disadvantage of FM is the wide bandpass required to transmit FM signals. Each station must be assigned a wide band in the frequency spectrum.

SINGLE-SIDEBAND COMMUNICATIONS

The sidebands contain the intelligence of an AM signal, and for normal operation, the information in both sidebands is the same. It is possible to transmit only one sideband, eliminating the carrier and the other sideband, yet retain the information transmitted. This method of transmission is called *Single Sideband* (SSB).

Figure 3-6 illustrates the transmitted signals for both AM and SSB. There are several advantages to SSB communications. By eliminating the carrier and one sideband, all of the transmitted power can be concentrated into a single sideband. Also, an SSB signal occupies a small portion of the frequency spectrum in comparison to the AM signal. This results in two advantages:

1. a narrower receiver bandpass, and
2. the ability to place more signals in a small portion of the frequency spectrum.

SSB communications systems have some disadvantages. The process of producing an SSB signal is more complicated than simple amplitude modulation, and frequency stability is much more critical in SSB communication. While there is not the annoyance of heterodyning from adjacent signals, a weak SSB signal may be completely hidden from the receiving station by a stronger signal. Also, a carrier of proper frequency and amplitude must be reinserted at the receiver because of the direct relationship between the carrier and the sidebands.

Figure 3-7 illustrates the block diagram of a basic SSB receiver. It is not significantly different from a conventional superheterodyne AM receiver. However, a special type of detector and a carrier reinsertion oscillator is required. The carrier reinsertion oscillator must furnish a carrier to the detector circuit at a frequency that corresponds almost exactly to the position of the carrier in producing the original signal.

The filters used in the RF amplifier section of the SSB receivers serve several purposes. As previously stated, many SSB signals may exist in a small portion of the frequency spectrum. Therefore, filters supply the selectivity necessary to adequately receive only one of the many signals that may be present. They may also select Upper Sideband (USB) or Lower Sideband (LSB) operation when desired, as well as reject noise and other interference.

The oscillators in an SSB receiver must be extremely stable. In some types of SSB data transmission, a frequency stability of plus or minus 2 Hz is required. For simple voice communications, a deviation of plus or minus 50 Hz may be tolerable.

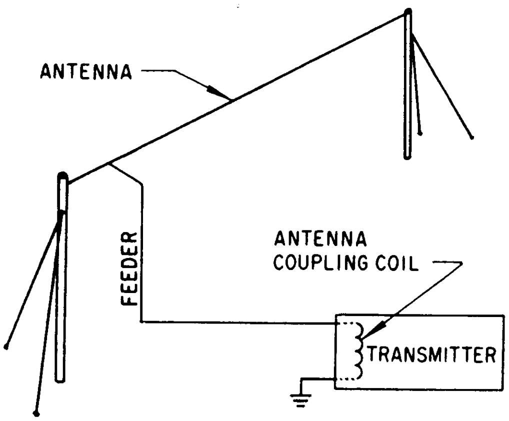

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Figure 3-8.—Typical antenna system.

SSB receivers may use additional circuits to enhance frequency stability, improve image rejection, or provide Automatic Gain Control (AGC). However, the circuits contained in the basic receiver will be found in all SSB receivers.

ANTENNA PRINCIPLES

The antenna is a basic component of any electronic system dependent upon free space as the propagating medium. It serves as the connecting link between free space and the transmitter or receiver. It is of primary importance in determining the performance of the system in which it is used.

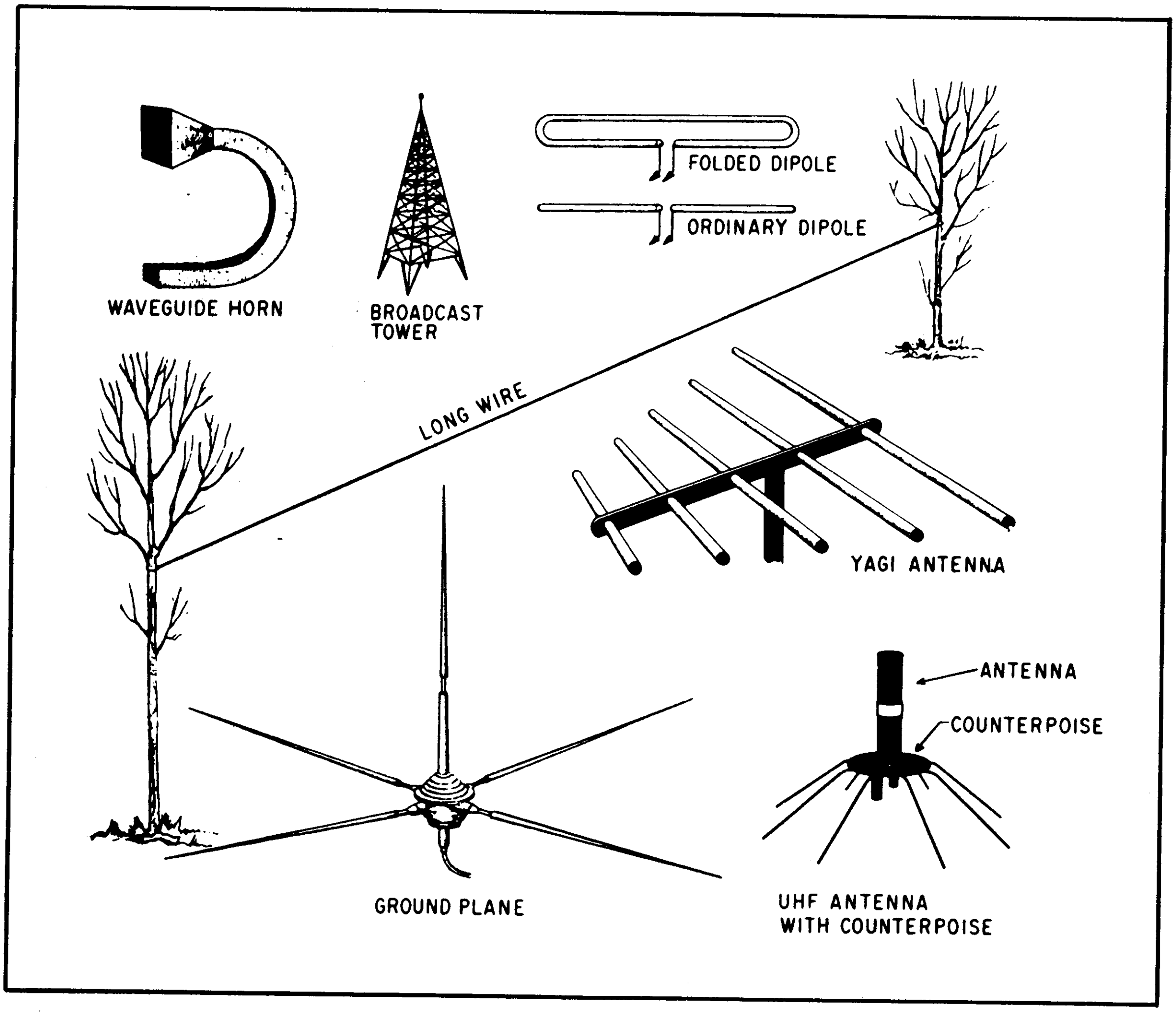

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Figure 3-9.—Typical antennas.

After generating an RF signal in a transmitter, there must be some means to radiate this signal through space to a receiver. The antenna performs this function. The transmitter signal is sent into space by a transmitting antenna; a receiving antenna then picks up the RF signal from space. The RF energy is transmitted into space in the form of an electromagnetic field. As the traveling electromagnetic field arrives at the receiving antenna, a voltage is induced into the conductor (antenna). The RF voltages induced into the receiver antenna are then used to recover the transmitted RF information.

At a transmitting station, the design of the antenna system is very important. The antenna must be able to radiate efficiently so that the power supplied by the transmitter is not wasted. An efficient transmitting antenna must have exact dimensions. The transmitting frequencies determine the dimensions. The dimensions of the receiving antenna are not critical for relatively low radio frequencies. However, as the frequency of the signal being received increases, the design and installation of the receiving antenna become more critical. An example of this is a television receiving antenna. If you give it a few more inches in height or a slight turn in direction, you can change a snowy blur into a clear picture.

The conventional antenna is a conductor, or a system of conductors, radiating or intercepting electromagnetic wave energy. An ideal antenna has definite length and uniform diameter, and is completely isolated in space. However, this ideal antenna is not realistic. Many factors make the design of an antenna for a communications system a more complex problem than you would expect. These factors include the height of the radiator above the earth, the conductivity of the earth below it, and the shape and dimensions of the antenna. All of these factors affect the radiated field pattern of the antenna in space. Another problem in antenna design is that the radiation pattern of the antenna must be directed between certain angles in the horizontal and vertical planes.

Most practical transmitting antennas are divided into two basic classifications:

1. Hertz (half-wave) antennas, and
2. Marconi (quarter-wave) antennas.

Hertz antennas are generally installed some distance above the ground and are positioned to radiate either vertically or horizontally. Marconi antennas operate with one end grounded and are mounted perpendicular to the earth or to a surface acting as a ground. Hertz antennas are generally used for frequencies above 2 MHz. Marconi antennas are used for frequencies below 2 MHz and may be used at higher frequencies in certain applications. A complete antenna system consists of three parts:

1. the coupling device,
2. the feeder, and
3. the antenna.

See figure 3-8. The coupling coil connects the transmitter to the feeder. The feeder is a transmission line carrying energy to the antenna. The antenna radiates this energy into space. The factors that determine the many types, sizes, and shapes of antennas are

- the frequency of operation of the transmitter,
- the amount of power to be radiated, and
- the general direction of the receiving set.

Figure 3-9 shows typical types of antennas. Figure 3-10 illustrates transmitting and receiving antennas, with their associated electromagnetic waves. The electromagnetic waves are divided into three components according to propagation characteristics:

1. groundwaves,
2. skywaves, and
3. spacewaves.

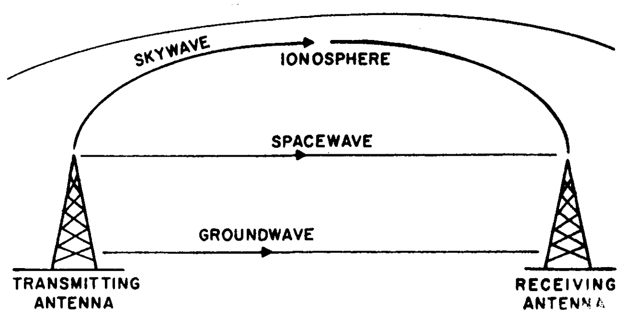

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Figure 3-10.—Divisions of the transmitted electromagnetic wave.

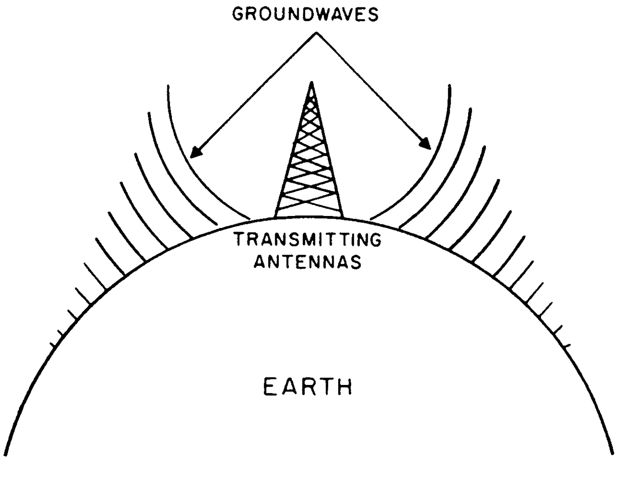

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Figure 3-11.—Groundwave component of the transmitted electromagnetic wave.

These types of propagation will be discussed in the following paragraphs.

GROUNDWAVE PROPAGATION

The groundwave is the portion of the radiated wave that moves along the surface of the earth. The field strength of the groundwave diminishes with distance much quicker than the waves that move through free space. There are many complex factors contributing to this; several are briefly described here.

Absorption by the earth increases with an increase in frequency. Therefore, long-distance communications by groundwaves are limited to low frequencies at very high power. Daytime reception of the standard broadcast band is an example. The type of soil near the antenna site will also be a factor in attenuation of the groundwave. Clay or loam will attenuate the signal less than sand or rock. However, salt water will propagate the signal better than either type of soil. The horizontally polarized wave is short-circuited by the earth and is attenuated much quicker than the vertically polarized wave. Thus, to gain maximum advantage, the groundwave must be transmitted and received using vertically polarized antennas. Despite these limiting factors, the groundwave remains the most reliable means of radio communications because most restricting factors are of a constant nature and do not vary with time of day or weather conditions. See figure 3-11.

SKYWAVE PROPAGATION

The skywave is the portion of the electromagnetic signal radiated upward that may or may not be refracted back to earth by the ionosphere (the upper atmosphere beginning 40 to 50 miles above the earth). See figure 3-12. Skywave propagation is not as reliable as groundwave propagation. However, greater distances may be covered by this means because the radiated electromagnetic field is directed toward the ionosphere and is refracted back at distances of hundreds or even thousands of miles. Refraction is a bending of the electromagnetic energy caused by the energy passing from a more dense medium to a less dense medium. Refraction of electromagnetic energy at radio frequencies is much like refraction at the high frequencies of light. Viewed from the surface, an underwater object appears to be foreshortened, or displaced, because of light refraction. This phenomenon is explained by differences in density of the media through which the light travels. Similarly, density differences in the ionosphere account for the refraction, or bending, of radio waves.

Electromagnetic energy is refracted, or bent, back toward earth when passing through the layers of the ionosphere. The angle at which the energy is refracted depends upon many variables; the major ones being frequency, radiation angle, height, and density of ionospheric layers. One angle where the electromagnetic waves enter the ionosphere will cause the energy to be refracted, but not enough to return to earth. This angle is referred to as the *critical angle*. Any transmitted energy entering the ionosphere beyond this angle continues into free space.

The ionosphere differs from other atmospheric layers because it contains a much higher number of positive and negative ions. The negative ions are believed to be ions in which energy levels have been raised to a high level by solar bombardment of ultraviolet and particle radiation. Extending from about 30 to 250 miles in the ionosphere, there are four layers of ionization: D, E, F1, and F2. See

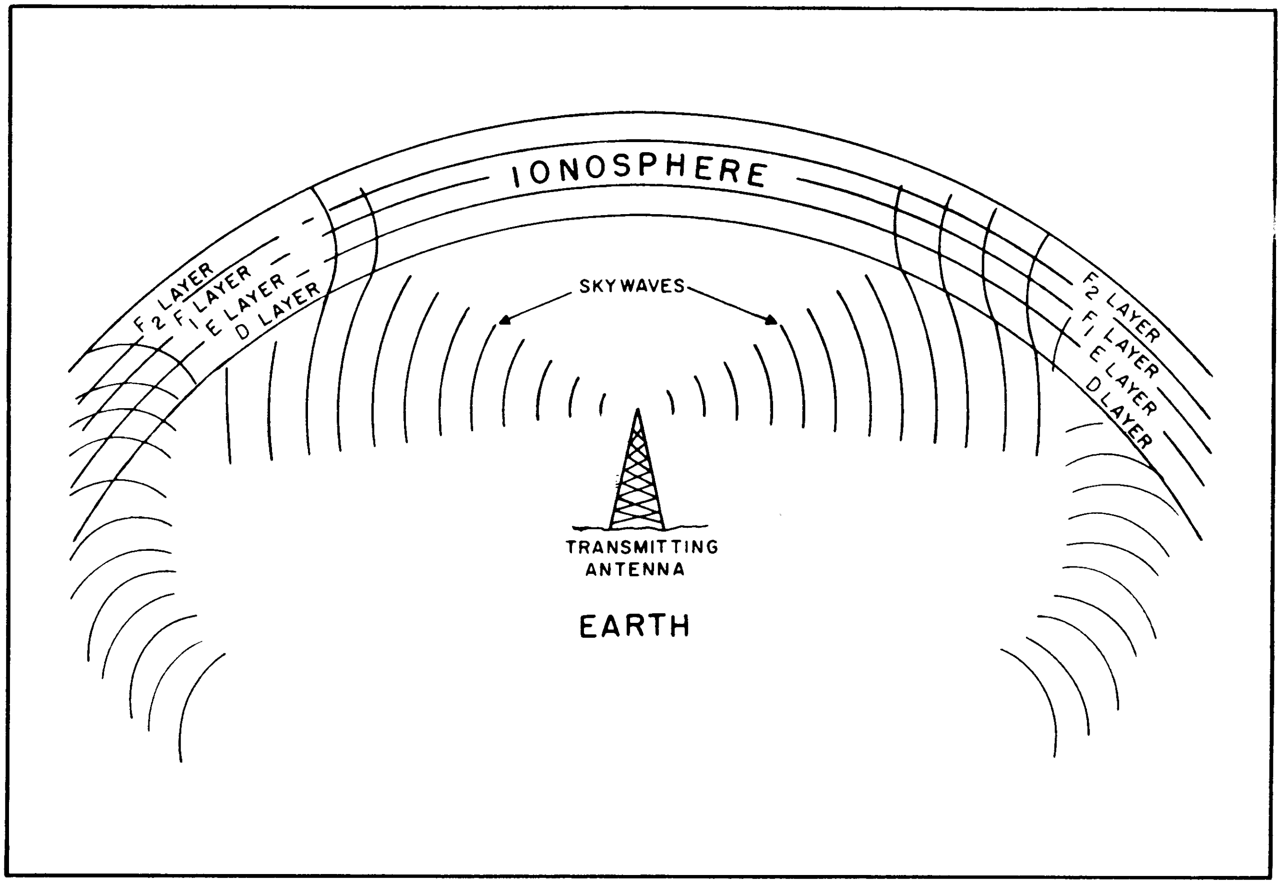

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Figure 3-12.—Refraction of the skywave component during daylight hours.

figure 3-12. Although ionization appears in distinguishable layers, the intensity and height of ionized layers in any given region depends on many factors, including season, sunspot cycle, and the time of day. The D layer is present only during daylight and has little effect on refraction. It is a factor in absorbing energy from the electromagnetic fields that pass through it. The E layer is much stronger during the day than at night and can refract frequencies up to approximately 20 MHz during the daylight hours. The F layers have the most effect on the refraction of electromagnetic energy. During the night the D layer fades, the E layer becomes much weaker, and the F1 and F2 layers combine into a single F layer. The reduction in absorption losses due to the fading of the D and E layers can cause the electromagnetic energy to cover greater distances at night, by permitting the combined F1 and F2 layers to reflect higher energy level signals at greater angles.

Multihop transmissions are used in long-distance radio communications. During these transmissions, a sequence of refractions in the ionosphere and reflections from the earth occur, causing the electromagnetic energy to bounce several times over the distance covered.

The complete effects of all the variables on skywave propagation are not fully understood. Researchers are continuously searching for means to improve the reliability of long-distance skywave communications.

SPACEWAVE PROPAGATION

The spacewave, sometimes referred to as the *direct ground wave*, is that part of the total wavefront that travels directly to the receiving antenna, or is reflected by the earth from the transmitting antenna to the receiving antenna.

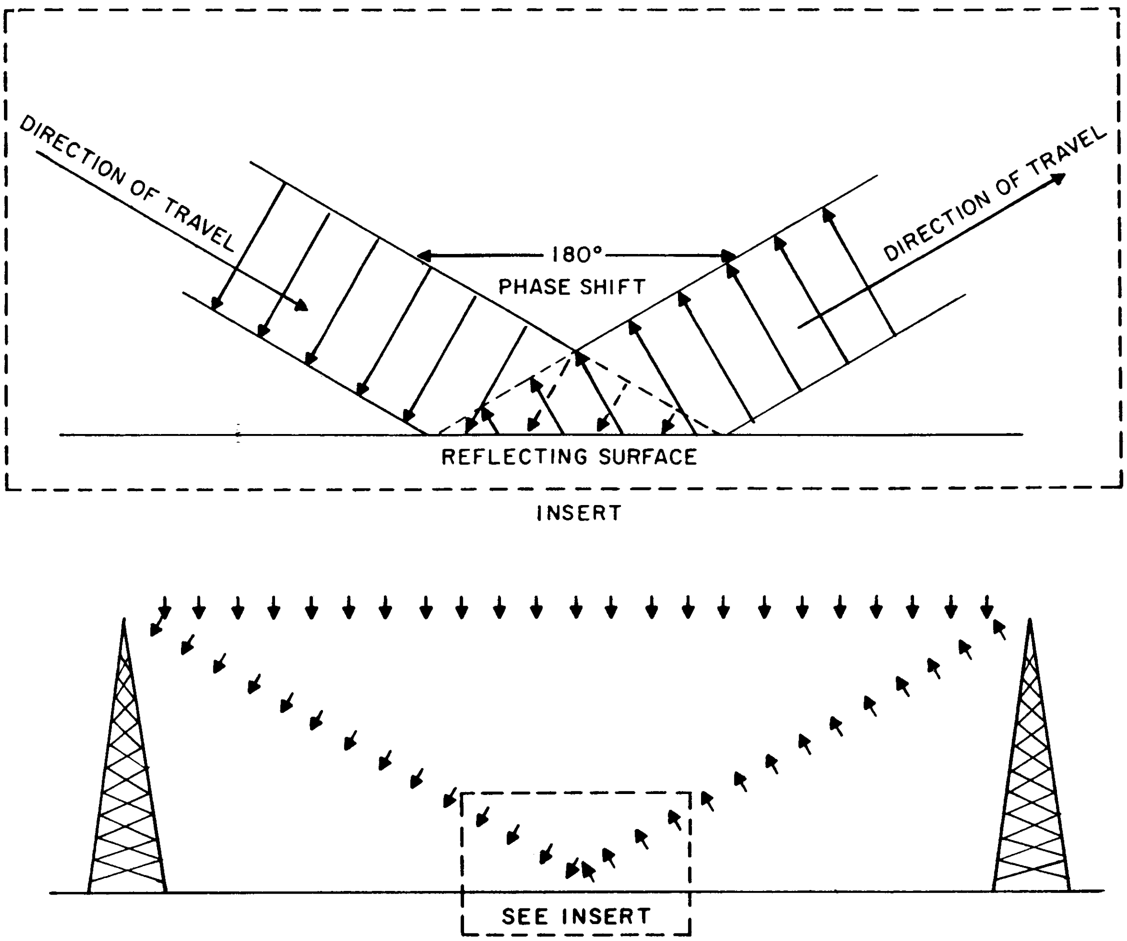

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Figure 3-13.—Spacewave component of the transmitted electromagnetic wave.

See figure 3-13. The spacewave is limited to line-of-sight distances, plus the additional small distance created by the bending of the wave (by atmospheric diffraction) a slight amount around the curvature of the earth. Line-of-sight distance can be increased by increasing either or both of the heights of the transmitting and receiving antennas.

The reflected spacewave is reflected by the earth at some distance between the transmitting and receiving antennas. When the transmitted wave strikes the surface of the earth, it loses a part of its energy in the form of heat dissipation, and the balance is reflected at the same angle at which it arrived. When the wave is reflected from the surface of the earth, it undergoes a phase reversal of 180 degrees. See insert, figure 3-13. In addition, the reflected spacewave, traveling a longer route, arrives at the point of reception later in time than the direct line-of-sight spacewave. These two factors are important considerations, since the 180-degree phase shift, plus its longer route, may cause the reflected spacewave to be out of phase with the direct wave at the point of reception. In other words, the two waves may have a tendency to cancel at the receiving antenna.

DATA COMMUNICATIONS

To understand the role of data communications, you must have an understanding of the basic components and techniques used to generate and process data on a communications link. Typically, data signals are transmitted over voice telephone channels, primarily because telephone facilities are available universally and, therefore, cost can be maintained at a reasonable level.

The most common situation in data communications is a computer interfacing with a remote terminal over voice telephone lines. This allows users at remote locations to use a central computer for data processing. The equipment used to perform this interaction has two general classifications:

1. **Data Communications Equipment (DCE)**: The equipment used to convey information between locations (for example, a modem).

2. **Data Terminal Equipment (DTE)**: The remote terminal, where information enters and exits from the data link for a user, and the computer, where information is processed and stored.

COMPUTER CODES

The first job in any data-processing operation is to convert the data into a code that the computer can understand. In all computer codes, each alphabetic and alphanumeric character is represented as a sequence of binary digits, 1s and 0s. The most common codes have 5-, 6-, 7-, or 8-bit characters. These include the

American Standard Code for Information Interchange,

Extended Binary Coded Decimal Interchange,

Six-Bit Transcode, and

Baudot Code.

AMERICAN STANDARD CODE FOR INFORMATION INTERCHANGE (ASCII)

Most terminal equipment and computer manufacturers use the ASCII code, allowing compatibility between different makes of devices. ASCII characters are coded in seven bits, with an eighth bit available for use as a parity bit.

EXTENDED BINARY CODED DECIMAL INTERCHANGE (EBCDIC)

IBM uses the EBCDIC. This code has 8-bit characters. Since the eighth bit is part of the code, there is no condition for character-by-character parity checking.

SIX-BIT TRANSCODE (SBT)

IBM has also developed this abbreviated 6-bit code. Its use increases the efficiency of high-speed data lines.

BAUDOT CODE

This 5-bit papertape code was the first serial asynchronous code developed for telegraph equipment. It is the most common code today, used by most of the world's printing telegraph communications networks.

LOCAL LINKS

Local or short-distance communications are generally accomplished by causing either voltage or current changes in a local wire circuit. This local wire circuit is commonly referred to as a *local link*. Typical applications rarely exceed 1 mile, although the actual distance limit depends on individual circuit characteristics. Two of the most commonly used local links are 20-milliampere (mA) loops (current) and Electrical Industry Association (EIA) lines (voltage).

TWENTY-mA LOOPS

The scheme to modify the current flowing through the wire dates back to the early days of telegraphy. With this method, binary code (data) is created by turning off and on a constant current (20 mA) in both the sender and the receiver.

Circuits with 20-mA currents are suitable for a wide range of applications. Such a circuit is often referred to as a 20-mA loop.

These 20-mA loops require two wires to complete the loop for transmitting data and

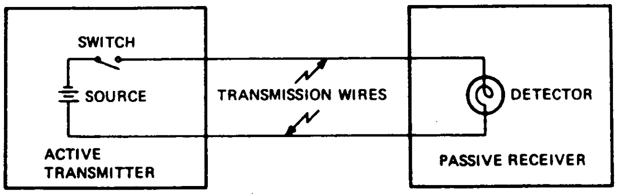

Figure 3-14.—Four basic elements of a 20-mA loop.

two wires to complete the loop for receiving data. A current loop requires four basic elements:

1. a current source,
2. the transmission wires,
3. a switch to interrupt the current flow, and
4. a detector to sense current in the circuit.

See figure 3-14.

ELECTRICAL INDUSTRY ASSOCIATION (EIA) LINES

In the early days of data communications in the United States, American Telephone and Telegraph (AT&T) was virtually the only provider of data communications services. Therefore, the Data Communications Equipments (DCE) developed by Bell Laboratories and manufactured by Western Electric were the standards of the industry.

Manufacturers of computers and Data Terminal Equipment (DTE) who wanted to use the services provided by the Bell System needed to know the electrical characteristics of Bell's equipment, so that they would be able to interface to them. To solve this problem, the EIA, in cooperation with the Bell System, the independent modem manufacturers, and the computer manufacturers, developed a standard for the interface between DTE and DCE.

Both 20-mA loops and EIA lines have problems that limit their uses. For example, 20-mA loops usually do not provide added wires for controlling the DCE. Therefore, they usually cannot be used for data communications that require some type of DCE. As you will see, most long-distance communications use some type of EIA DCE. EIA lines are more open to electronic noise than 20-mA loops and, therefore, do not adjust well to electrically noisy environments. Because of such problems, 20-mA loops and EIA lines can be used for short, local transmissions only. The maximum effective data rate drops rapidly as the length of the local link increases.

INTERFACE STANDARDS

Since most long-distance data transmissions take place over common carrier facilities, modems must conform to the standards set by the Federal Communications Commission (FCC).

In data communications, standards have been established for interfacing DTE to DCE. DTE to DCE is synonymous with source-to-sync or line-to-equip. The current EIA RS-232-C standard is the most popular standard being used. Because of its limitations, a new EIA standard, RS-449, is being introduced as a replacement for RS-232-C.

Another popular standard is the International Standards Organization (ISO) 2593 and the equivalent European Comite Consultatif Internationale de Telegraphic et Telephone (CCITT) V.35 standard.

EIA STANDARDS OVERVIEW (RS-232-C VERSUS RS-449)

The RS-232-C interface has some serious limitations for use in modern data communications systems, the most critical being that of speed and distance. EIA RS-232-C restricts speed to 20,000 bits per second (20 kbs), and distance between DCE and DTE to 50 feet. Because of these restrictions, the industry has developed the RS-449 standard that maintains compatibility with the RS-232-C to accommodate the complete transition to the new standard.

The most significant difference between RS-449 and RS-232-C is in the electrical characteristics of signals between DTE and DCE. RS-232-C uses only unbalanced circuits, while RS-449 uses both balanced and unbalanced circuits (balanced circuits improve both speed and distance).

ASYNCHRONOUS AND SYNCHRONOUS TRANSMISSION

Data can be transmitted over electric media asynchronously or synchronously. In an asynchronous transmission, each character generated by the transmitting end moves separately over the link. In asynchronous communications, there is no common timing between the transmitting and receiving ends.

Each asynchronous character sent over the link has two bits added to it. The preceding bit

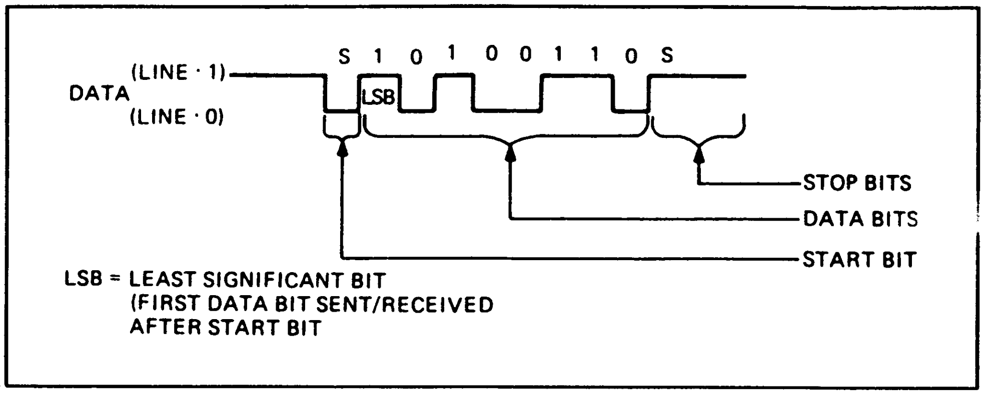

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Figure 3-15.—Typical asynchronous character frame.

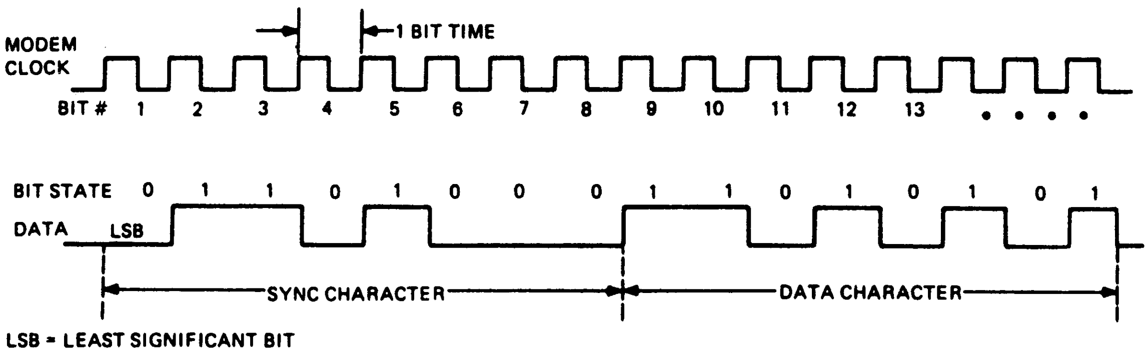

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Figure 3-16.—Typical synchronous transmission.

is called the *start bit* and the trailing bit is called the *stop bit*. See figure 3-15.

The start bit is used to inform the receiving end of the line that a character is coming. The stop bit (which may be the length of one, one and one-half, or two bits) is provided to give the receiving end enough time to reset itself before the next character arrives.

Asynchronous mode is commonly used for low-speed or irregular transmissions (i.e., operator chatter). The main characteristic of asynchronous communication is that the receiver cannot predict when the next character will arrive, and must constantly monitor the line for start and stop bits.

Synchronous communication is characterized by a common timing source between transmitter and receiver. This timing is usually provided by the DCE.

In synchronous transmission, the transmitter stores up a string of characters and transmits them together at a high, steady rate of speed. One or more special synchronizing characters, called *sync*, at the beginning of the block of characters (usually 256 characters to a block), synchronizes the receiving end with the transmitting end, since all the bits representing data must use a timing device to tell where one character ends and the next starts. See figure 3-16. If the receiving end is out of sync with the transmitting end by as much as a single bit, the message has no meaning.

Synchronous transmission is the usual method of computer-to-computer communication. It is also used with terminals designed to transmit characters in blocks.

Since start and stop bits are not required in synchronous transmissions, a standard ASCII 8-bit character can be transmitted without the 20% overhead (start and one stop bit) required in asynchronus transmissions.

MODEM

The term *modem* is an acronym for MOdulator-DEModulator. It is a device that modulates and demodulates signals. Modems are primarily used for converting digital signals into analog and audio signals for transmission and for reconverting the analog signals into digital signals.

While all modems perform the prime functions of modulation and demodulation, there are other functions that can add to their capability, or, in some situations, can restrict performance to better accomplish a specialized task.

TESTING

The three types of testing can be performed with modems are the

1. analog loopback test,
2. self test, and
3. digital loopback test.

The analog loopback test allows the operator to check out the local modem, using any data pattern that can be generated. This test provides for local looping of transmit and receive data.

The self test, by itself or in conjunction with other loopback tests, allows the operator to check out the local modem, using a built-in message generator and a message comparator. This does not require a data input and can be configured to check out both local and remote modems.

The digital loopback test allows the operator to verify the data link through the local DTE, local modem, telephone line, and remote modem. This test requires that someone be present at both the local and remote ends to configure the modems for the test.

A special feature of some modems is the remote digital loopback test. This test gives you the testing features of digital loopback, but does not require that someone be present at the remote end to prepare the modem for the test.

TRANSMIT ONLY/RECEIVE ONLY FEATURES

With the transmit only/receive only features, users can configure their modems to either transmit only or receive only. In some applications, it may not be necessary for a modem to be able to do both. Also, a modem with only one of these features will cost less than a modem with both.

ACOUSTIC COUPLING

Acoustic couplers are special modems that have a small speaker that changes digital signals directly into audible tones. These tones are picked up by a standard telephone handset.

COMMUNICATIONS INTERFACE

The communications interface is located between the modem and the computer. The functions of the interface include modem control, buffering, and multiplexing.

Modem Control

The RS-232-C standard allocates added wires for modem control. Modem control includes certain "handshaking" routines that must take place before and after a data transmission. These routines are made up of a sequence of questions and answers, such as: "Are you there?" "Yes." "Can you receive a message?" "Yes, I can. Go ahead."

Buffering

As entering data arrives from the modem, the interface assembles characters in an output buffer. When the terminal is ready to process a character, it takes it from the output buffer. If the terminal does not recover the character from the output buffer before the interface receives the next character, the first character is lost. This is called an *overrun*. To minimize the occurrence of overruns, some interfaces use a method called *double buffering*. In high-speed interfaces or multiplexers, the buffer may be expanded by using a memory device called a *silo* for the FIFO (First-In, First-Out) buffer. Asynchronous communication interfaces remove the start and stop bits from each entering character. Synchronous interfaces remove control elements from entering blocks of data and divide the bit stream into its component characters.

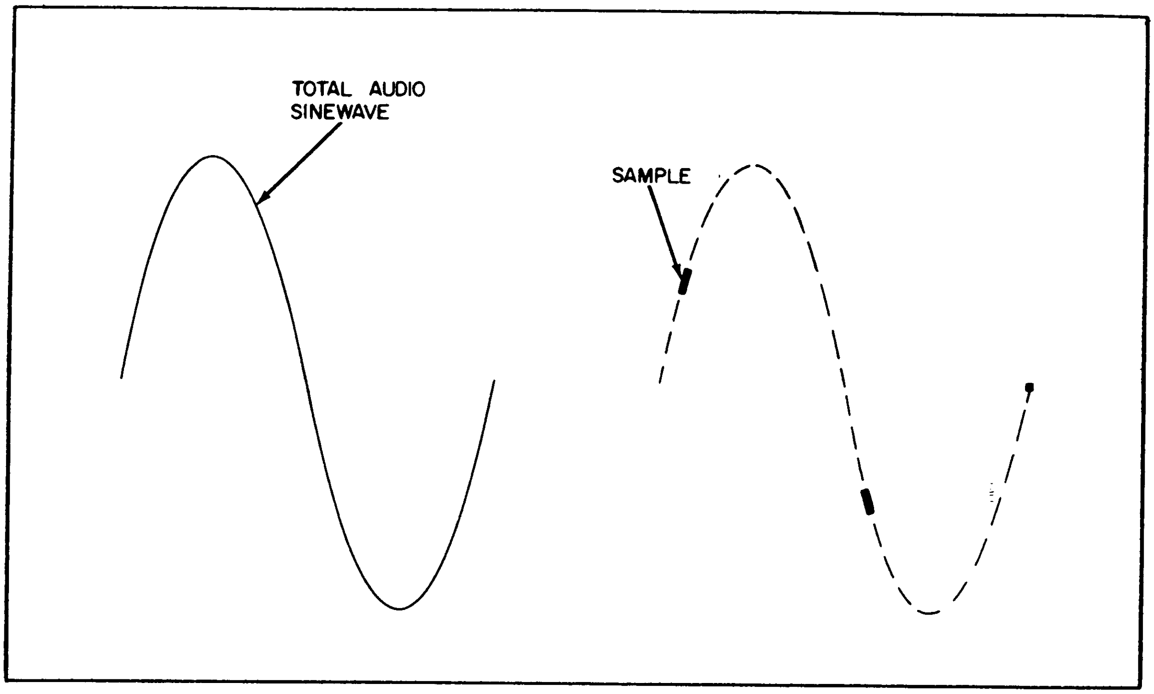

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Figure 3-17.—Components of a sine wave.

MULTIPLEXING

In multiplexing, the physical link is divided so that it can carry multiple messages at the same time. The multiplexer takes in data from several remote sources and sends it on over the same number of subchannels. Since fewer physical links are needed to carry multiple circuits, this method significantly increases the usage of each link. The two most commonly used methods of multiplexing are time-division multiplexing and frequency-division multiplexing.

TIME-DIVISION MULTIPLEXING (TDM)

An audio signal may be transmitted and received satisfactorily by periodically sampling the signal. The result of the sampling process yields a received signal such as that shown in figure 3-17. Figure 3-18 illustrates, in a highly simplified form, the basic principle of TDM. Assume that a 3-kHz tone is applied to each of the six channels in the transmitter. Assume also that the rotating switch turns fast enough to sample, in turn, each of the six channels 2.4 times during each cycle of the 3-kHz tone. The speed of rotation of the switch must then be 2.4 × 3,000 or 7,200 rotations per second for the optimum sampling of 2.4:1.

If the transmitter and receiver switches are synchronized, the signals will be fed in the proper sequence to the receiver channels. The transmitted samples from transmitter channel 1 will be fed to receiver channel 1. Thus, in the time-division method of multiplexing, many channels of audio are combined (with time spacing between components of the separate channels) to form a single output (multiplexed) chain. The chain is transmitted to distant demultiplexing receivers. Each receiving channel selects and reconstructs only the information included in the originally transmitted channel.

More than six channels may be used. However, as the number of channels is increased, the width of each sample segment must be proportionately reduced.

Time-Division Switching

The theory described in the previous paragraphs is basically time-division switching. User lines are switched to channels by allocating time

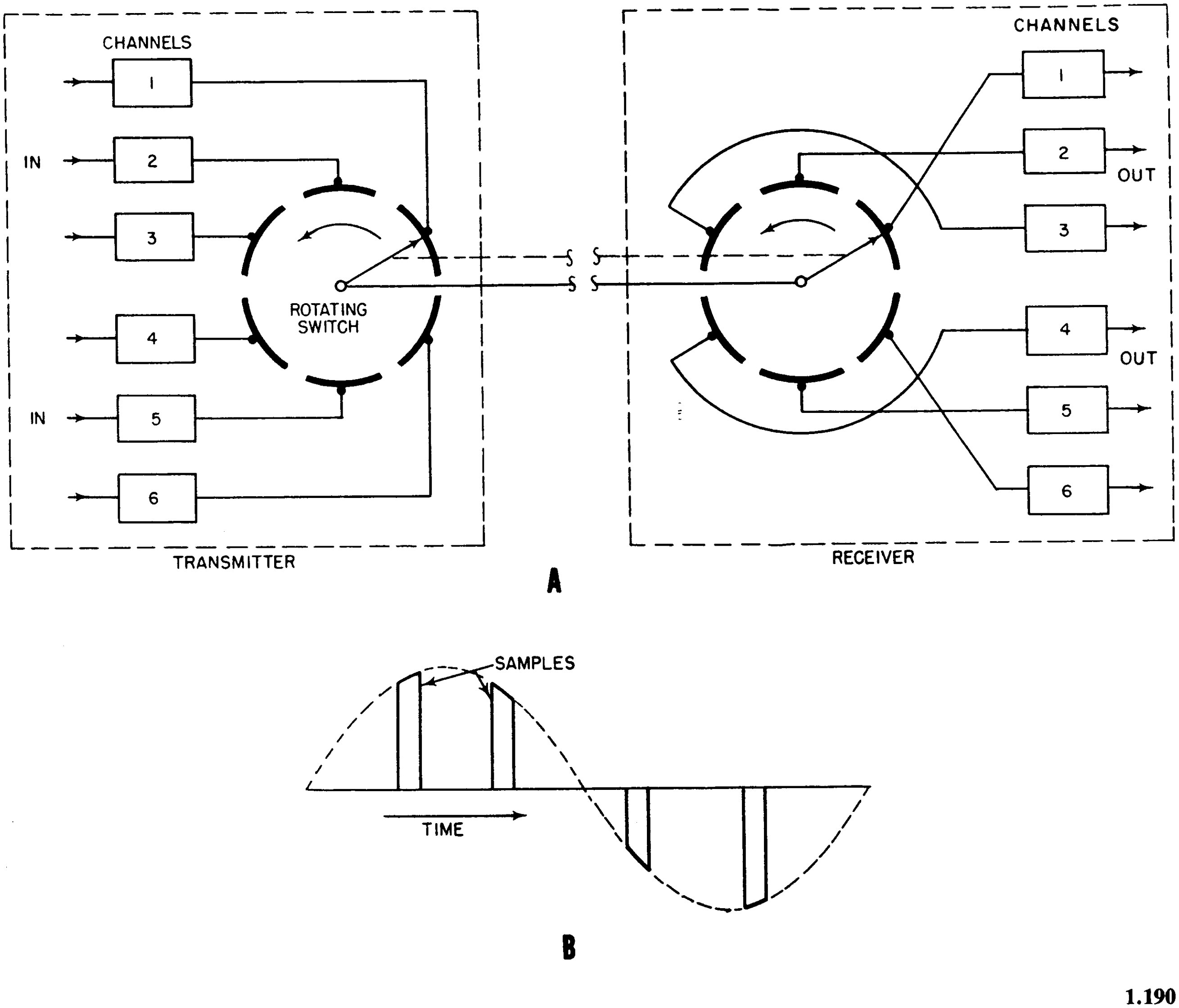

Figure 3-18.—Basic principle of time-division multiplexing.

slots in the channels to the lines and vice versa. Switching centers using time-division switching, or multiplexing, simply expand on this concept.

Types of Time-Division Switching

There are three basic types of time-division switching: circuit, message, and packet.

1. CIRCUIT SWITCHING: In circuit switching, each circuit is assigned a path through the switching equipment. In time-division switching, this is accomplished by assigning a different time slot to each line and channel appearing at the switching system.

2. MESSAGE SWITCHING: Message-switching systems route entire messages from incoming to outgoing lines. Incoming messages are stored on paper or magnetic tape or in electronic buffers (memories) until the required outgoing line is available. Message switching can be done manually by means of the now outdated torn-tape relay. Although outdated, it is still in widespread use for non-real-time communications. Electronic message switching is also used for non-real-time communications, but the delays are relatively insignificant.

3. PACKET SWITCHING: Packet switching is a method for handling data transmission through a communications system. The Packet-Switching Nodes (PSN), to which subscriber

computers are attached, divide information streams into small packets and route each packet as a separate message. The individual packets contain the data being transmitted, the source and destination address, and a check for error detection. The PSN then either forwards the packet to another PSN over a predetermined path or delivers the packet directly to the addressee. The more advanced packet-switching networks, such as the DDN, use adaptive routing. Predetermined routing data cannot be forwarded easily if any network component in the predetermined path fails. With adaptive routing, PSNs avoid portions of the network that are congested or logged down.

Packet switching differs from message switching in that in message switching, no data is transmitted over the line until the entire message has been received in the message switch. This means that the first message to be completely received is the first one to go out. Messages are not equal in length or size. The first message that may be completely received may not be the first one that was starting to be received by the switch. A short message may start coming in after a long message has started and be completely received before the long message is fully received. In this case, the short message would be transmitted out ahead of the long message, even though much of the long message is received in storage at the switch awaiting the "end-of-message" receive signal before it may be transmitted from the switch.

Packet switching partially alleviates the delay problem of message switching, because long messages are partitioned into packet segments and queued for transmission. This way, transmission begins before the total message has been received and stored at the first packet-switching node.

The term *packet switching* normally implies the movement of fixed length blocks of data from point to point via a switching system matrix. Packet switching could be considered a distributed processing medium shared by unrelated users. The system is normally designed for the movement of large volumes of data. The user provides the data-processing system at each user location and is a subscriber of the packet-switching, interlinking computer network. The user's data is sent to a packet-switching computer over a local access line. At the switching computer, the data is organized into packets of from one to a given number of characters, depending on the user's application. Packets received at the switching computer from many users are merged into a high-speed data stream and passed to the network line, where each line is connected to two or more switching computers.

Depending on the address of the destination's switching computer in the packet header, a packet will be routed to or through one or more network-switching computers until it arrives at the final destination. The destination's switching center will then pass the packet to the addressee by another local access line.

During transmission, within the packet-switching system, the data is divided into blocks, framed by control and header characters and error control bits. In this manner, the transferred data is protected against distortion and possible loss. Each data block being transferred is retained at each switching computer until a positive acknowledgment is received from the next two switching computers receiving the data block. In addition, each switching computer has an alternate route to each of the other switching computers, allowing continuous service when equipment or a connecting line fails.

The advantage of the packet-switching matrix concept is that it provides flexibility of operation where distributed data processes large volumes of data transfers, data storage, and data protection.

FREQUENCY-DIVISION MULTIPLEXING (FDM)

FDM, unlike TDM, transmits and receives the full 360 degrees of a sine wave. FDM used by the Navy may be divided into two categories, one for voice communications and the other for teletype (TTY) communications.

The normal voice speaking range is from .1 to 3.5 kHz. During single-channel AM voice communications, the audio frequency amplitude modulates a single radio frequency (carrier frequency). However in voice FDM, each voice frequency modulates a separate frequency lower than the carrier frequency (subcarrier frequency).

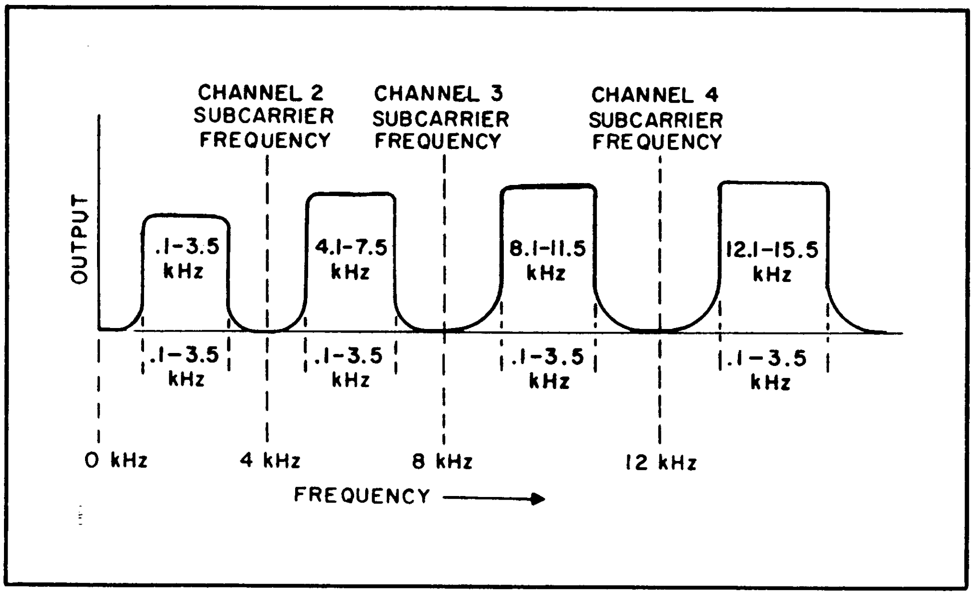

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Figure 3-19.—Block diagram of a frequency-division multiplexing system.

If these subcarrier frequencies are separated by 3.5 kHz or more, they may be combined on a composite signal to modulate the carrier frequency without causing excessive interference with each other.

For example, in figure 3-19, the output of channel 1 is the voice frequency range (.1 to 3.5 kHz). The output of channel 2 is the combination of a different voice frequency with a subcarrier frequency of 4 kHz, giving an output frequency range of 4.1 to 7.5 kHz. Similarly, the output of channel 3 is another voice frequency combined with a subcarrier frequency of 8 kHz, giving an output frequency range of 8.1 to 11.5 kHz. The overall bandwidth for the composite modulation package shown in figure 3-19 is, therefore, .1 to 15.5 or 15.4 kHz, with each separate channel occupying its own band of frequencies. This composite signal is then used to modulate the carrier frequency of the transmitter.

Multichannel broadcast and ship and shore terminations use teletype FDM, whereby each channel of the composite tone package of the broadcast or termination is assigned an audio frequency. An advantage in multiplexing teletypewriter circuits is that many individual teletypewriter circuits may be carried in any one of the 3-kHz audio channels described above. The two types of multiplexing should not be confused. In the first case, 3-kHz audio channels have been combined. In the second case, a number of dc teletypewriter circuits are converted to tone keying and are combined into a single 3-kHz audio channel. Figure 3-20 illustrates a 16-channel, teletypewriter multiplexing system in which the output of the dc-pulsed circuits is converted to audio keying. Each channel has a separate audio center frequency. Channel frequencies range from 425 Hz, for the lowest channel, to 2975 Hz, for the highest channel. A mark in an individual teletypewriter loop will key an audio tone 42.5 Hz above the center frequency. Referring to figure 3-20, the mark and space frequencies for channel 1, for example, may be calculated as 382.5 Hz and 467.5 Hz, respectively (425 +/− 42.5). Combining these keyed tones into a composite signal results in a tone package within a standard 3-kHz bandwidth. By occupying no more than 3 kHz of the audio spectrum, the output signal is suitable for transmission via radio or landline.

DESIGNATION OF MODULATION CLASSES

As communication systems grew more complex, an international designation system was needed to describe the characteristics of the type of modulation being transmitted. This

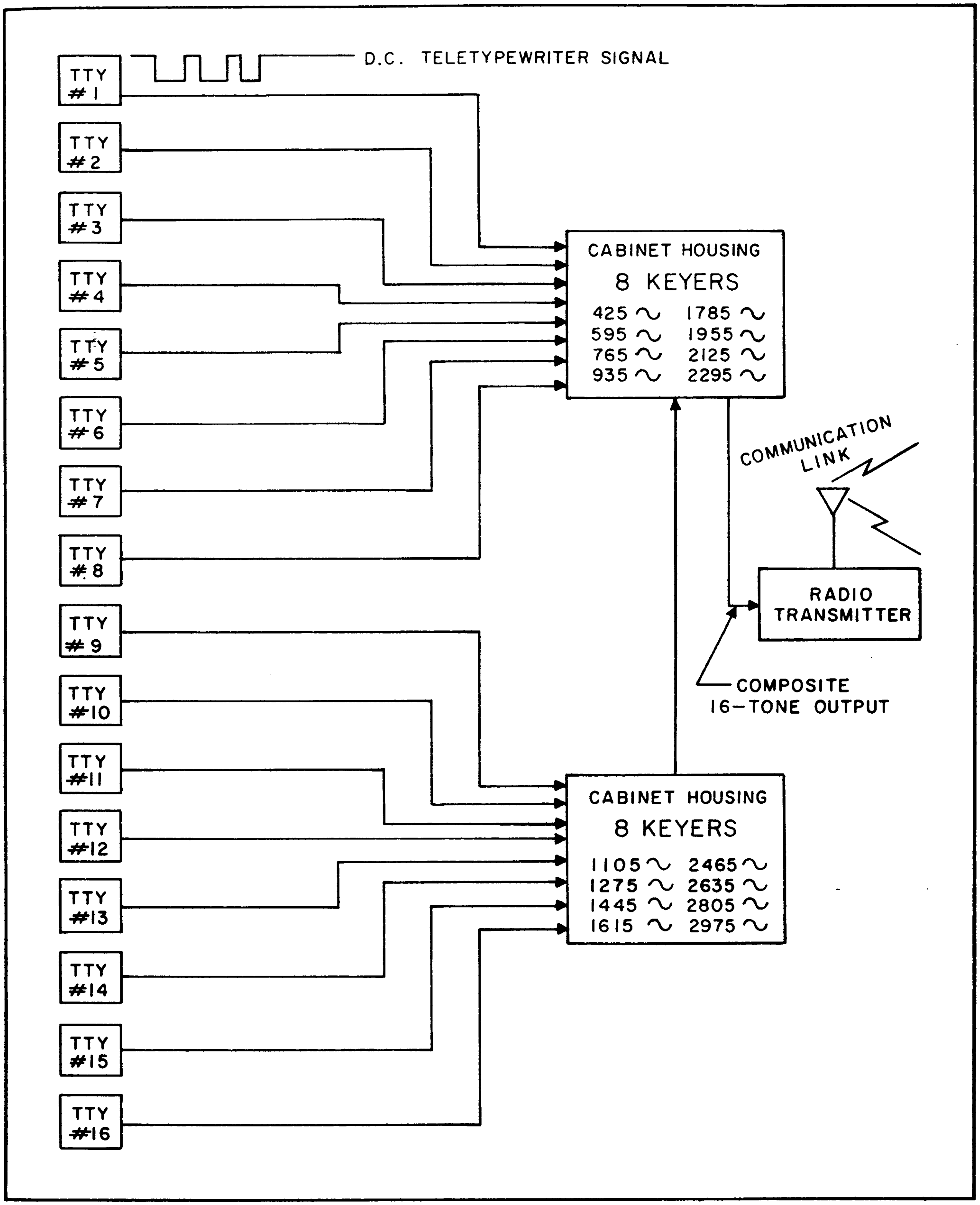

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Figure 3-20.—Block diagram of modulator units.

international system is shown in table 3-2. It designates the RF emissions by type, mode, and supplementary characteristics. For example, A3B denotes telephony, two independent sidebands, with suppressed carriers. A number preceding this designation, such as 6A3B, indicates the bandwidth in kHz. Table 3-2 is the international system used throughout the Navy.

TRANSMISSION

Transmission of wideband signals requires media capable of transmitting many channels simultaneously over a link. Present wideband equipment in the Defense Communications System (DCS) usually provides from 12 to 600 Variable-Frequency (VF) channels. Unlike HF radio, a distinct separation of transmitter and receiver facilities is not required because of simultaneous transmission and reception using a single antenna.

Transmission media using VHF, UHF, microwave, tropospheric scatter, and satellite radio may include repeaters. These facilities are called *relay stations* (for radio) and *repeater stations* (for cable). However, the functions performed give the same results where the transmitted signals are concerned. The repeaters provide equalization and amplification for incoming signals to permit the onward transmission of a signal that, as nearly as possible, duplicates the original transmitted signal.

TYPES OF TRANSMISSION MEDIA

Different types of transmission media facilitate communications between stations. The type of equipment used for any link is determined by the distance and terrain between stations, strategic and economic factors, and the technical parameters required to provide appropriate service to users. The different media include

- high-frequency radio;
- line-of-sight radio (such as microwave, VHF, and UHF);
- tropospheric scatter radio;
- satellite;
- cable; and
- open wire.

Table 3-2.—Emission Designations

Emission	Type
	Modulation Types
Amplitude	A
Frequency	F
Pulse	P
Modulation (Transmission Mode)	
None	0
Telegraphy (Keyed RF carrier)	1
Telegraphy (Tone)	2
Telephony	3
Facsimile	4
Television	5
Four Channel Duplex Telegraphy	6
Multichannel Voice Frequency Telegraphy	7
Cases not covered above	9
Supplemental Characteristics	
Double Sideband	NONE
Single Sideband	
—reduced carrier	A
—full carrier	H
—suppressed carrier	J
Two independent Sidebands	
—suppressed carriers	B
Vestigial Sideband	
—amplitude modulated	D
—width modulated	E
—phase modulated	F
—code modulated	G

Link Application

Transmission links are communications paths of the same transmission media between two points. They vary in location and length due to limitations of the equipment and propagation characteristics of the operating frequencies. The following are the normal limitations for each type of link:

VHF, UHF, and microwave radio: Line of sight between transmitting and receiving stations.

Tropospheric scatter radio: 100 to 500 miles between transmitting and receiving stations.

Satellite radio: Usually from 2,000 to 6,000 miles, depending mainly on the height of the satellite above the earth.

High-frequency radio: Up to approximately 6,000 miles, dependent upon total system design and use.

Signaling Speed and Line Capacity

The rate at which data can be transmitted is measured in bits per second (bps). The bit rate refers to the number of bits that can be transmitted in a second. Signaling speeds vary from about 50 bps to over a million bps.

The capacity of a line to carry information is determined by its bandwidth. Bandwidth refers to the frequency range of a channel. It is the difference, in Hertz (frequency speed in cycles per second), between the maximum frequency and the minimum frequency that can be transmitted over the line. Therefore, a frequency range from 5 to 10 Hz and another from 15 to 20 Hz have the same bandwidth—5 Hz.

Data communications lines are classified according to bandwidth:

Subvoice (narrow) band: 0 to 300 Hz

Voice band: 300 to 3000 Hz

Wideband grade: Over 3300 Hz (3000 to 3300 Hz is not used)

These frequency ranges are general classifications. Many times, the actual bandwidths may overlap. The speed of transmission increases as bandwidth and frequency increase. Therefore, a narrowband is considered low speed, a voice band is considered medium speed, and a wideband is considered high speed.

Subvoice Band

A subvoice band channel has the narrowest bandwidth and the lowest speed of the three grades. Typical transmission speeds range from 45 to 150 bps. This is also called *telegraph grade*, since it is often used with telegraph and similar equipment. This band is normally used in asynchronous operations. The bits of each character are synchronized separately, but the characters themselves are generated at random intervals.

Voice Band

Voice band channels allow the complete 3-kHz bandwidth used for voice transmission to be used for data. Most voice-grade services operate from 60 to 4800 bps. Speeds of up to 10,800 bps are possible with some modems. Voice grade provides a good, economical means of data transmission up to 4800 bps.

Wideband Grade

Because of the bandwidth available with wideband grade (up to 240 kHz with leased lines), data can be passed at speeds from 19,200 to 500,000 bps. Even high speeds are possible for special applications. If needed, wideband grade is used to transmit large volumes of data at high speed.

Line Conditioning

Voice-grade lines are low-fidelity data paths. To improve the quality of the line, the telephone company must condition it to provide the quality desired. Private, leased lines are the only types of lines that can be conditioned. Switched lines cannot be conditioned, because the same data paths are not used every time. With private, leased lines, the same physical path is used each time.

DEDICATED FACILITIES

Dedicated facilities provide communications for common-interest, high-volume, or priority traffic users. These facilities are connected by DCS circuits allocated to full-time support of their traffic. However, the DCS retains full authority over, and responsibility for, transmission links providing the connecting circuits.

GENERAL CIRCUITS

A circuit is an electronic path between two or more users, between a user terminal and a switching terminal, or between two switches. Individual circuits may be engineered in any of the following configurations:

- **Full-duplex circuit**: A method of operation whereby all telecommunications between stations take place in both directions simultaneously.

- **Half-duplex circuit**: A circuit that permits unidirectional electrical communications between stations. This term is qualified by one of the following suffixes: S/O (send only) or R/O (receive only).

- **Simplex circuit**: A circuit capable of transmissions in both directions, but not simultaneously. This term is qualified by the suffix S/R (send or receive).

DATA CIRCUIT CONTROL (DCC)

As discussed in topic 1, monitoring and controlling of traffic within the CRITICOMM system is maintained by CRITICOMM System Management (CSM). The DCC facility in CSM serves as the control point for all matters regarding multiplexed and non-multiplexed, high-speed circuit operation. To this end, MUX/DEMUX nodal points of the Pacific, Atlantic, and Continental U.S. (CONUS) areas have separate broadcasts connecting them with CSM and the Fort Meade TCF. Each broadcast is tied together by secure, full-period, 75-baud circuits from each facility into a switching device located at selected nodal points and extended to CSM. The DCC is a broadcast-type orderwire that provides for the rapid secure coordination and prompt restoration of CRITICOMM multiplexed circuits. It establishes re-routes in case of primary circuit path outage.

DATA-SWITCHED NETWORKS

The DCS switched networks are comprised of the operational Defense Data Network (DDN), the Automatic Digital Network (AUTODIN), and the Inter-Service/Agency Automated Message Processing Exchange (I-S/A AMPE). The DDN is a worldwide, common-user, packet-switching network designed to meet the long-haul and area data communications requirements of all DOD Automated Data-Processing (ADP) systems and data networks.

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