CHAPTER 11

MICROWAVE MULTIPLEX EQUIPMENT

Multiplex operation is the simultaneous transmission of two or more messages in either or both directions over the same transmission path. There are two general methods of accomplishing this: Frequency Division Multiplex (FDM) is the division of the available frequency spectrum into discrete bands, each of which carries one of the functions; Time Division Multiplex (TDM) is the division of the time available into discrete intervals that are assigned successively to the several functions.

11.1 FREQUENCY DIVISION MULTIPLEX

11.1.1 Functional Description

Telephone multiplex subsystems provide the overall communication system with the capability to transmit and receive a number of voice frequency channels over a single transmission subsystem. In FDM subsystems, all channels to be multiplexed onto a single broadband channel are "divided" in the frequency domain to keep the signal channels separated. The information contained in each channel within the composite FDM signal is transmitted during the same instant of time. That is, all FDM signal channels overlap in the time domain and, in effect, are transmitted in parallel. This is accomplished through the use of various frequency translations. Basically, frequency translation is the shifting of a band of frequencies from one part of the frequency spectrum to another. During the process of frequency translation, the information contained in the original band of frequencies is not changed. Frequency translation is obtained through the use of modulation techniques, and the translation may be to a higher or a lower band of frequencies.

Figure 11-1 illustrates the basic process of FDM for a 12 channel voice frequency system. With reference to the figure, it should be noted that each channel originally occupies a 4 kHz band of frequencies. During transmission, each channel is translated up in frequency to occupy a unique 4 kHz band in a continuous frequency region. For example, channel 12 is translated to occupy the 60 to 64 kHz band, channel 11 is translated to occupy the 64 to 68 kHz band, et cetera. The result of this translation process is the stacking of all 12 channels within a continuous frequency band of 48 kHz. The composite signal contained within the 48 kHz band may then be transmitted over a single transmission subsystem. One or more steps of frequency translation may be required depending on the multiplex and transmission subsystems that will be used.

During reception, each 4 kHz band of frequencies in the composite signal is translated down in frequency to obtain the 12 individual 4 kHz channels.

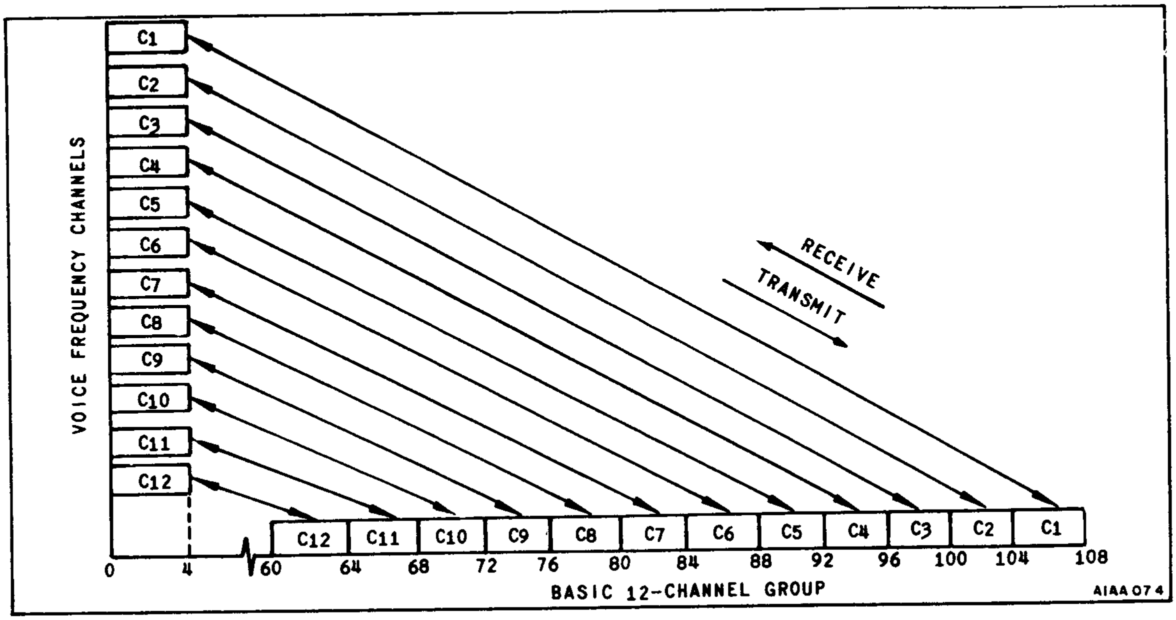


Figure 11-1. FDM Process for a Basic 12-Channel Group

All multiplex subsystems use some type of modulation scheme to translate the voice frequency signals to a composite signal of some suitable frequency range. The modulation scheme includes frequency allocations for the composite signal and the type of modulation used for frequency translation. The frequency allocation for the system shown in figure 11-1 represents the basic standard building block for long haul multiplex subsystems within the DCS. This standard, 60 to 108 kHz, 12 channel system is used to form systems with hundreds of channels by using additional steps of frequency translation.

The 12 channels contained within the 60 to 108 kHz band are commonly referred to as a 12 channel group, or simply a group. A 60 channel system, commonly called a supergroup, is obtained by translating five groups to another part of the frequency spectrum. The basic translation process is similar to that used to develop a group, the only difference being the frequency allocation and bandwidth of the composite supergroup signal. Figure 11-2 illustrates the structure of a 60 channel supergroup.

A 600 channel system is obtained by translating 10 supergroups using the same basic FDM process. The DCS standard frequency allocation for the 600 channel system is shown in figure 11-3. It should be noted that in the 600 channel system, supergroup 1 is translated down in frequency and supergroup 2 is transmitted without additional translation. All other supergroups are translated up in frequency by methods similar to the translation process for channels to group, and groups to supergroup. The use of guard bands between each supergroup when translated to the master group frequency spectrum should also be noted. Between supergroups 1 and 2, and 2 and 3, a 12 kHz guard band is used. An 8 kHz guard band is used between all other supergroups. In the case of translating from channels to group, and groups to supergroup, guard bands are not included in the composite frequency allocation.

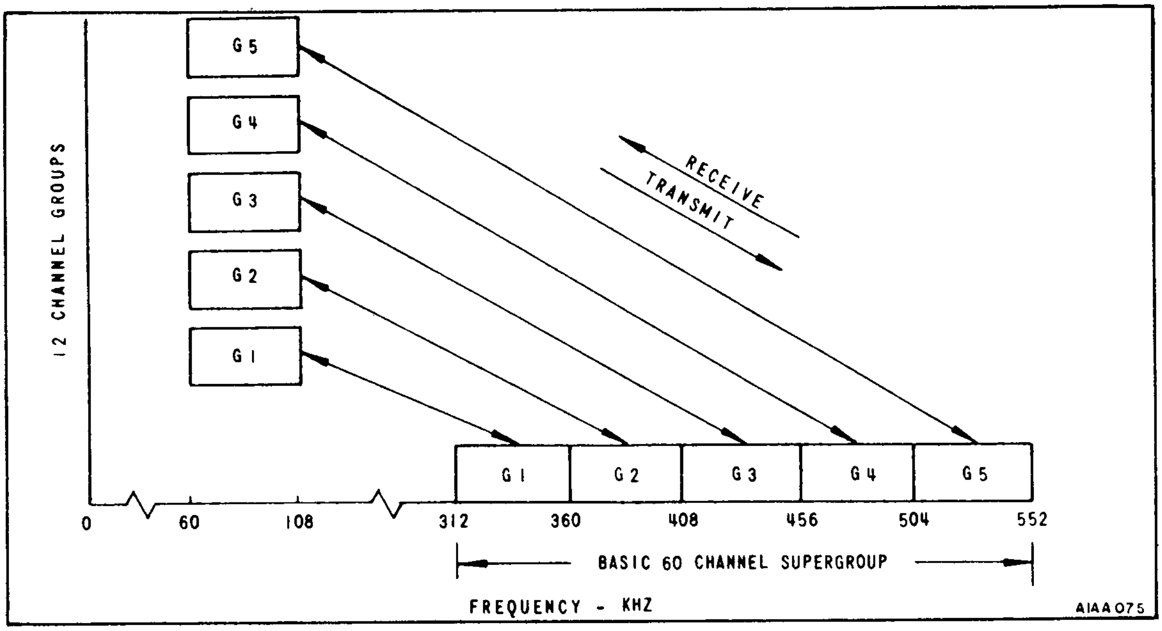


Figure 11-2. FDM Process for a 60-Channel Supergroup

The preceding discussion was intended to provide a broad functional description of the basic FDM process as applied to telephone channels. It should be noted that the actual frequency allocations for a particular telephone FDM multiplex system will be dependent upon:

- o The type of transmission subsystem
- o The number of channels required for the overall system.

Chapter 6, System Design, presents equipment performance criteria, including frequency allocation and interface parameters, for the various types of transmission subsystems used in the DCS.

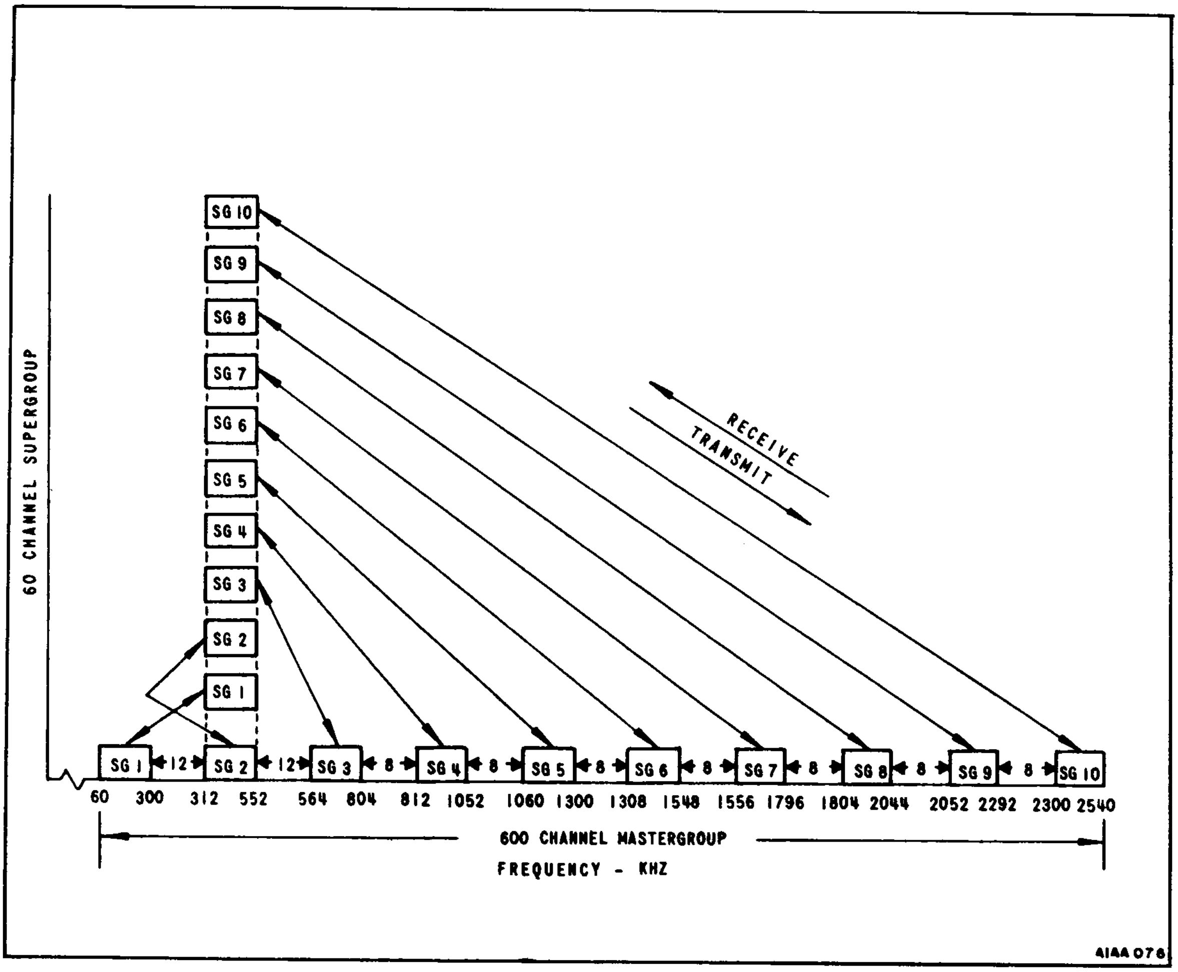


Figure 11-3. FDM Process for a 600-Channel Master Group

Telephone multiplex subsystems are designed primarily to handle voice frequency (VF) signals. DC telegraph signals, as found in teletypewriter operation, cannot be handled directly by telephone FDM equipment. By using the techniques of modulation and multiplexing, DC signals can be converted to AC tones in the VF range. The VF tones can be handled by the telephone FDM equipment. Voice frequency carrier telegraph (VFCT) equipment is used to provide this capability.

In FDM/VFCT operation within the DCS, a number of carriers within the VF range are modulated by DC signals from telegraph loops. Each telegraph loop is associated with a particular carrier. The nominal center frequencies of the carriers are generally spaced 170 Hz apart. With this spacing, it is possible to obtain 16 to 18 carriers within the VF range. Some VFCT equipments, such as the AN/FGC-75(V) and -76(V), use a phase modulation technique to transmit up to 32 telegraph channels over a single VF channel. This technique, however, is not specified in DCAC 330-175-1. In addition, such equipment is not compatible with conventional FDM VFCT equipment.

11.1.2 Theory of Operation

The basic technique used in FDM is frequency translation. Signals that occupy a particular band of frequencies are translated (shifted) to another part of the available frequency spectrum. The shifting of signals in frequency is accomplished through the process of modulation. Since AM is so important to the overall FDM process, the various forms of AM will be briefly described.

Basically, modulation is the process by which some property of a signal is varied in accordance with the intelligence to be transmitted. The signal that is varied by the intelligence is generally called the carrier signal, or simply the carrier. In AM systems, it is the amplitude of the carrier that is varied.

When a carrier is amplitude modulated, a complex signal is generated. In addition to the carrier frequency itself, an AM signal contains other frequencies commonly known as the upper and lower sideband frequencies. The upper sideband contains the sum of the carrier frequency and the frequencies in the modulating signal. The lower sideband contains the difference of the carrier and signal frequencies. Other sidebands centered at multiples of the carrier frequency are also generated, but are generally eliminated by filters.

Figure 11-4 illustrates the basic AM process. In this example, the intelligence to be transmitted contains frequencies up to 4 kHz, and the carrier frequency is 64 kHz. Both intelligence and carrier are mixed in the amplitude modulator in order to generate the AM signal. The resulting AM signal is centered around the carrier frequency of 64 kHz with the intelligence contained in the two 4 kHz wide sidebands. Thus, the intelligence is translated in frequency to another part of the frequency spectrum.

The AM process can be repeated to shift the signal frequencies to still another band, either higher or lower in frequency. It is this ability to translate signal frequency components to any part of the frequency spectrum that makes the process of modulation so important to FDM subsystems. The manner in which the sidebands are shown in figure 11-4 is the standard method when dealing with FDM subsystems. Upper sidebands are shown with the long side of the right triangle increasing in height from left to right. The reverse is true for lower sidebands. Also, it is common practice to call the upper sideband the erect sideband, and the lower sideband the inverted sideband. Furthermore, this AM process is also referred to as double sideband emitted carrier (DSBEC) in order to differentiate between certain variations of AM that are described below.

In AM systems, the carrier does not contain any intelligence. The carrier is used only to carry the intelligence to another band of frequencies. Furthermore, both the upper and lower sidebands contain identical information. Thus, the carrier and one of the sidebands are completely superfluous with regard to the intelligence contained in the AM signal. Because of this, certain FDM subsystems do not use AM directly but instead use variations of AM. The generally used variations of AM are:

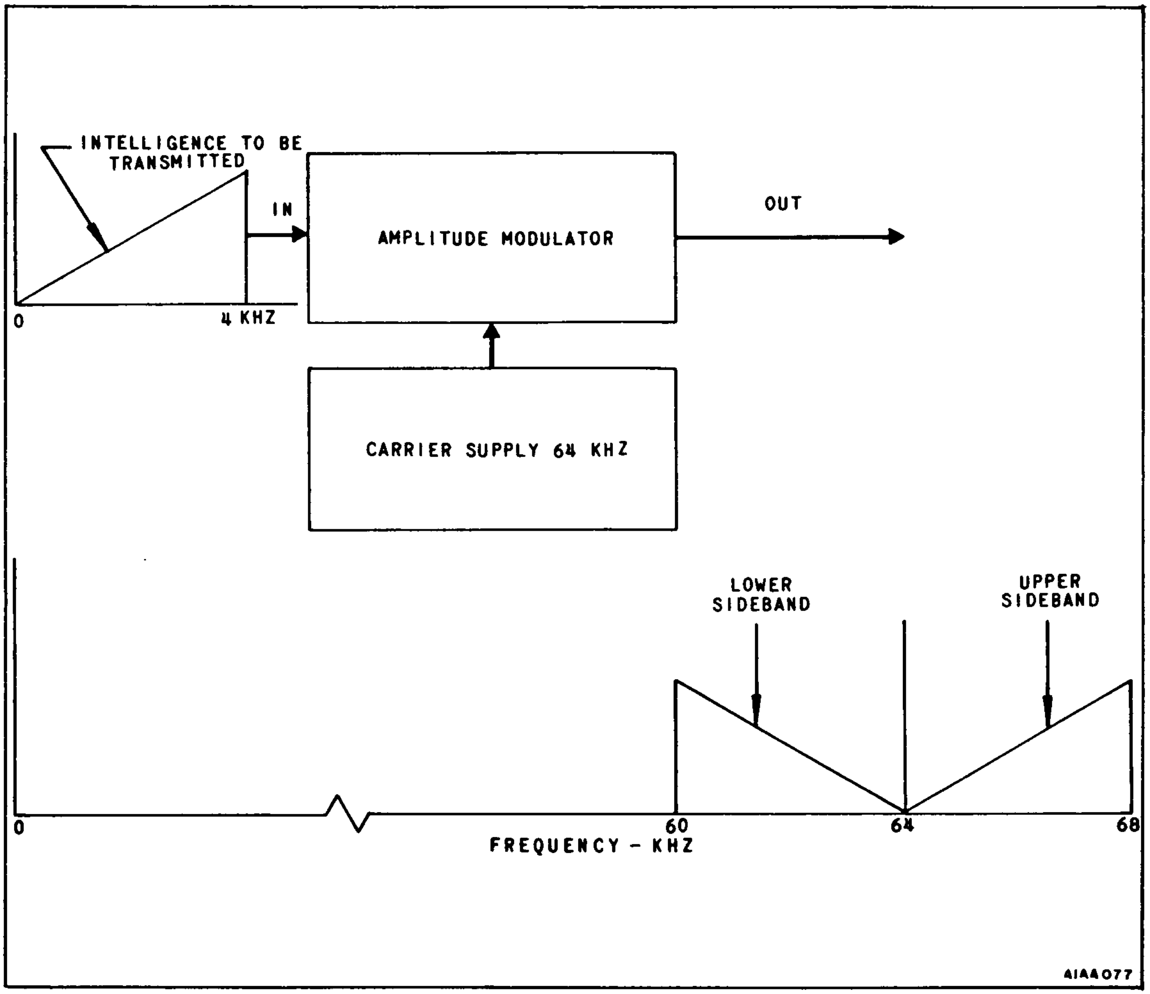


Figure 11-4. Basic Amplitude Modulation Process

- Single sideband suppressed carrier
- Double sideband suppressed carrier
- Independent sideband
- Single Sideband Suppressed Carrier (SSBSC).

With SSBSC, only one sideband is transmitted. The other sideband and the carrier are eliminated or suppressed. By transmitting only one sideband, the power required to transmit the signal is reduced. Also, the frequency band is effectively reduced to one-half of that required for a direct AM signal. It then becomes possible to transmit twice as many signal channels in the same multiplex frequency band.

Figure 11-5 illustrates a two channel SSBSC multiplex subsystem. Signals from each telephone transmitter pass through low-pass filters. These filters limit the upper end of the frequency band to about 4 kHz. The 4 kHz wide signals are applied to the balanced modulator where they are combined with their respective carriers as in conventional AM. By using a balanced modulator, the carrier is suppressed within and does not appear in the output signal spectrum. Therefore, the output of each balanced modulator contains only the upper and lower sidebands, centered around their respective suppressed carriers. Both sidebands are applied to a bandpass filter where the upper sideband is eliminated. The output of the bandpass filters are combined to form the composite FDM signal. The composite signal contains the intelligence from both telephone transmitters, and occupies an 8 kHz band between 60 and 68 kHz.

At the receiving end, the composite signals are applied to bandpass filters. The bandpass filter associated with channel 1 passes only the 60 to 64 kHz band, and that of channel 2 the 64 to 68 kHz band. The outputs of the bandpass filters are applied to balanced modulators where they combine with their respective carriers. This process is similar to that performed at the transmit end, and the carriers are suppressed within the balanced modulators. Therefore, the output of the balanced modulators contain only the upper and lower sidebands. The lower sidebands associated with each channel occupy the 0 to 4 kHz band, i.e., the original band of frequencies out of the low-pass filters at the transmit terminal. The upper sideband (associated with channel 1) occupies the 124 to 128 kHz band and that of channel 2 the 128 to 132 kHz band. The lower sidebands are then applied to the telephone receivers after the upper sidebands are removed by the low-pass filters.

The basic principles of operation for the illustrative two channel system apply to FDM subsystems that can handle hundreds of channels. This particular variation of AM is used in most high density FDM subsystems and is the standard FDM technique for international communication systems.

o Double Sideband Suppressed Carrier (DSBSC). In FDM subsystems that use the DSBSC variations of AM, both sidebands are used but the carrier is suppressed. This technique offers a saving in power required to transmit the composite multiplex signal but is still wasteful of frequency spectrum. However, DSBSC systems are less expensive to implement than the SSBSC variation. This is primarily due to the elimination of bandpass filters required to remove the unwanted sidebands in the transmit of the FDM equipment.

Both the emitted and suppressed carrier versions of the double sideband type of system find extensive use in multipair cable transmission subsystems. This is particularly true where transmission paths do not exceed 50 miles and the required number of channels is 24 or less.

The most widely used double sideband FDM subsystems fall into the category commonly known as the N-type of system. Various manufacturers offer FDM subsystems using the same frequency allocations and channel arrangements so that they coordinate back to back over the same transmission subsystem. Some systems are available that coordinate from a frequency standpoint with N-type systems, but that use single

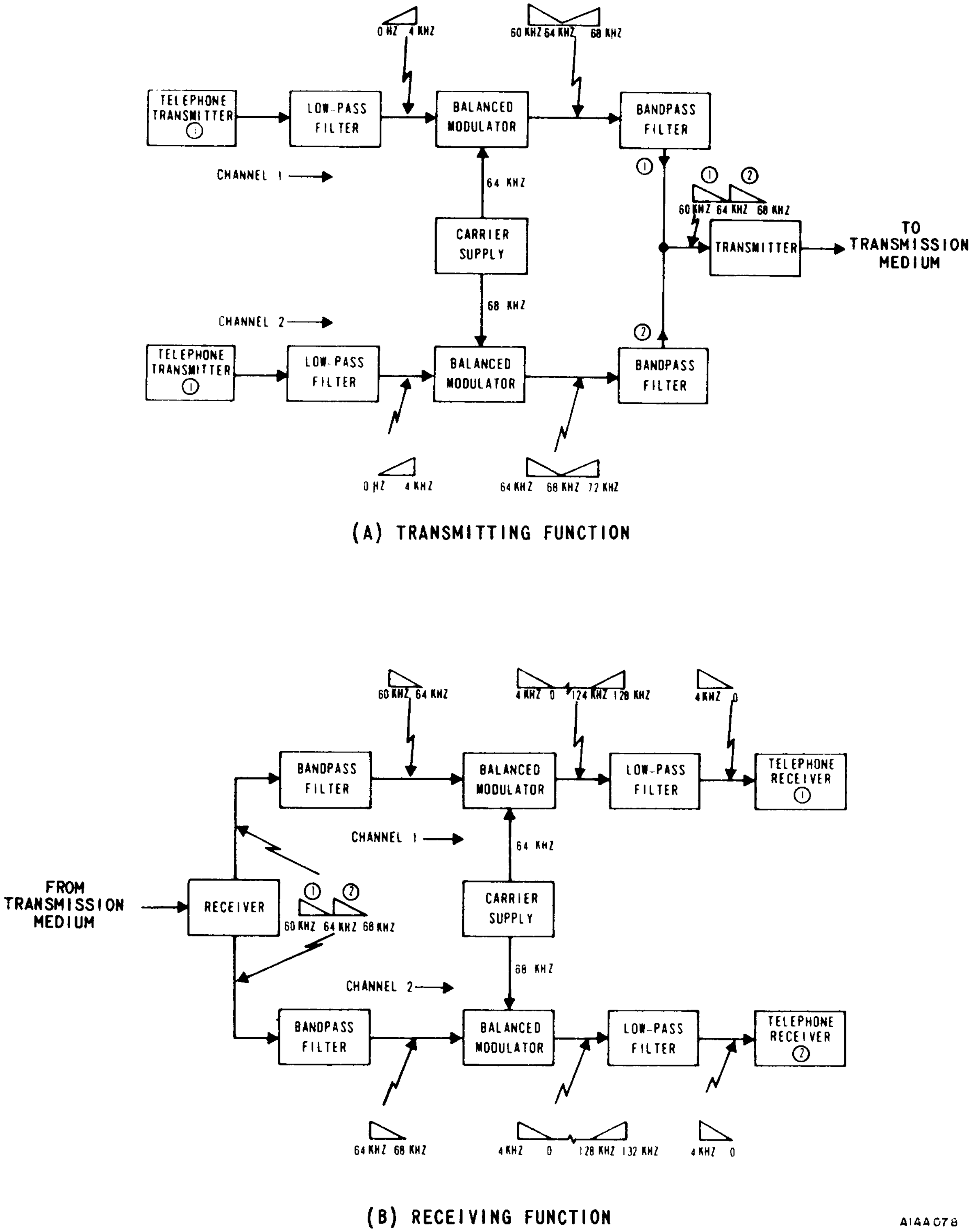


Figure 11-5. Two Channel SSBSC Multiplex Subsystem

sideband techniques. Therefore, the number of channels that can be handled on the transmission subsystem is doubled.

o Independent Sideband (ISB). With independent (twin) sideband modulation, a single carrier is used to generate two independent sidebands (the upper and lower sidebands centered around the same carrier contain different information). The same basic AM and filtering techniques described above are used to generate the ISB signal. The ISB method of modulation is used in high frequency (HF) subsystems and in certain high-density multiplexer sets.

In HF subsystems, the maximum authorized bandwidth in the HF spectrum is 12 kHz. Due to this narrow bandwidth and the need for the maximum number of channels within the 12 kHz spectrum, individual channels are limited to a bandwidth of 3 kHz. Using ISB, four channels can be accommodated within the 12 kHz spectrum. Two 2 kHz channels are first multiplexed into a 6 kHz band. Two such bands are used to modulate two independent sidebands of an HF ISB transmitter to produce four channels in the HF spectrum. Figure 11-6 illustrates the frequency spectrum of an HF ISB system.

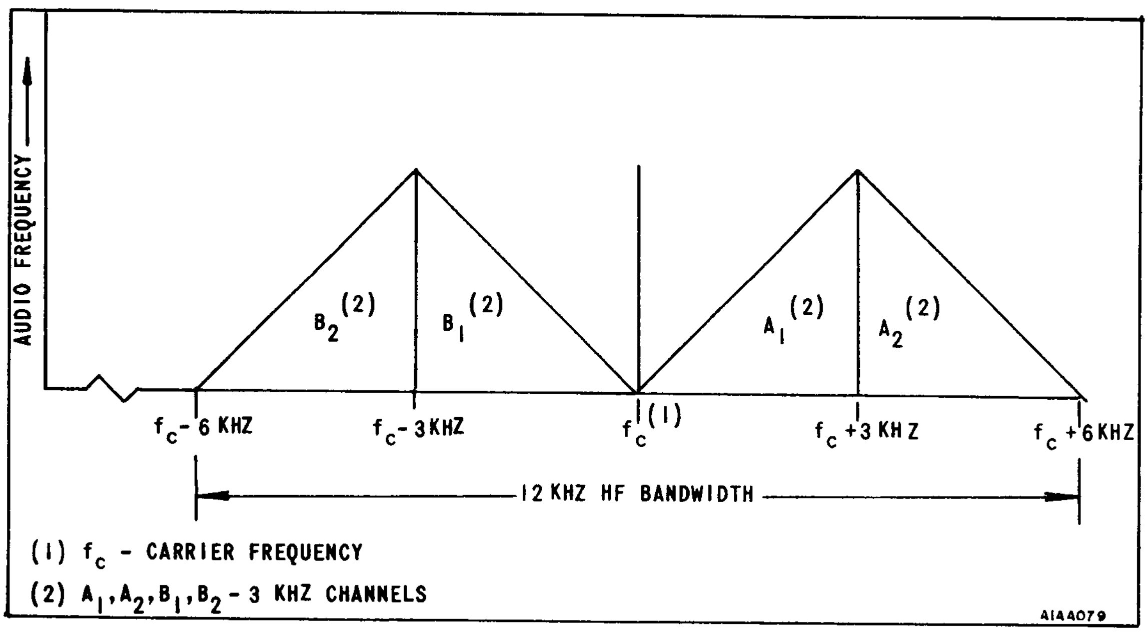


Figure 11-6. HF/ISB Frequency Spectrum

In high capacity multiplex sets (60 channels or greater) the group, supergroup, and line frequency allocations are in accordance with DCS and international standards. However, the channel multiplexing modulation is independent (twin) sideband as opposed to the all-lower sideband modulation that is the DCS and international standard. A later version of the AN/FCC-17 family, designated AN/UCC-4(V), provides an all-lower sideband capability. Figure 11-7 illustrates the difference between ISB and all-lower sideband modulation (note that 12 carriers are required to generate the standard basic group and only 6 carriers are required for the ISB basic group).

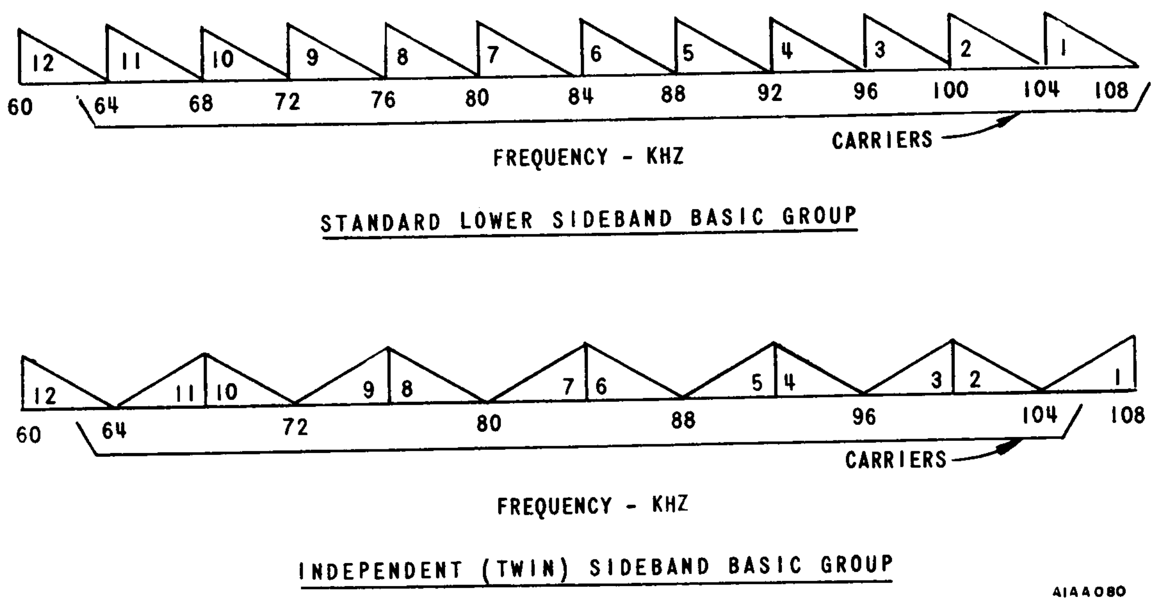


Figure 11-7. Comparison of Standard Basic Group With ISB Basic Group

11.2 TIME DIVISION MULTIPLEX (PULSE MODULATION)

11.2.1 Functional Description

In telephone TDM subsystems all voice frequency (VF) channels to be multiplexed are "divided" in the time domain. The information contained in each channel within the composite TDM signal is transmitted during a different instant of time but overlaps in a common frequency spectrum. That is, all TDM signal channels "time share" a common transmission channel and, in effect, are transmitted in a serial manner. The direct relationship of the composite signal to the original input signal found in FDM subsystems is not found in TDM subsystems due to the nature of the basic TDM process as described below.

The TDM method employs sampling techniques. If a signal is sampled at a rate twice its highest frequency component, an adequate representation of the signal may be obtained. If there are a number of channels to be sent over a common transmission path, the first channel is sampled briefly, then the second, and so on until the last channel. After the last channel is sampled, the process is repeated. Before each sample is applied to the common transmission path, some form of pulse modulation is used to form the composite TDM signal. Figure 11-8 illustrates the basic TDM transmission-process for a 12 channel telephone system.

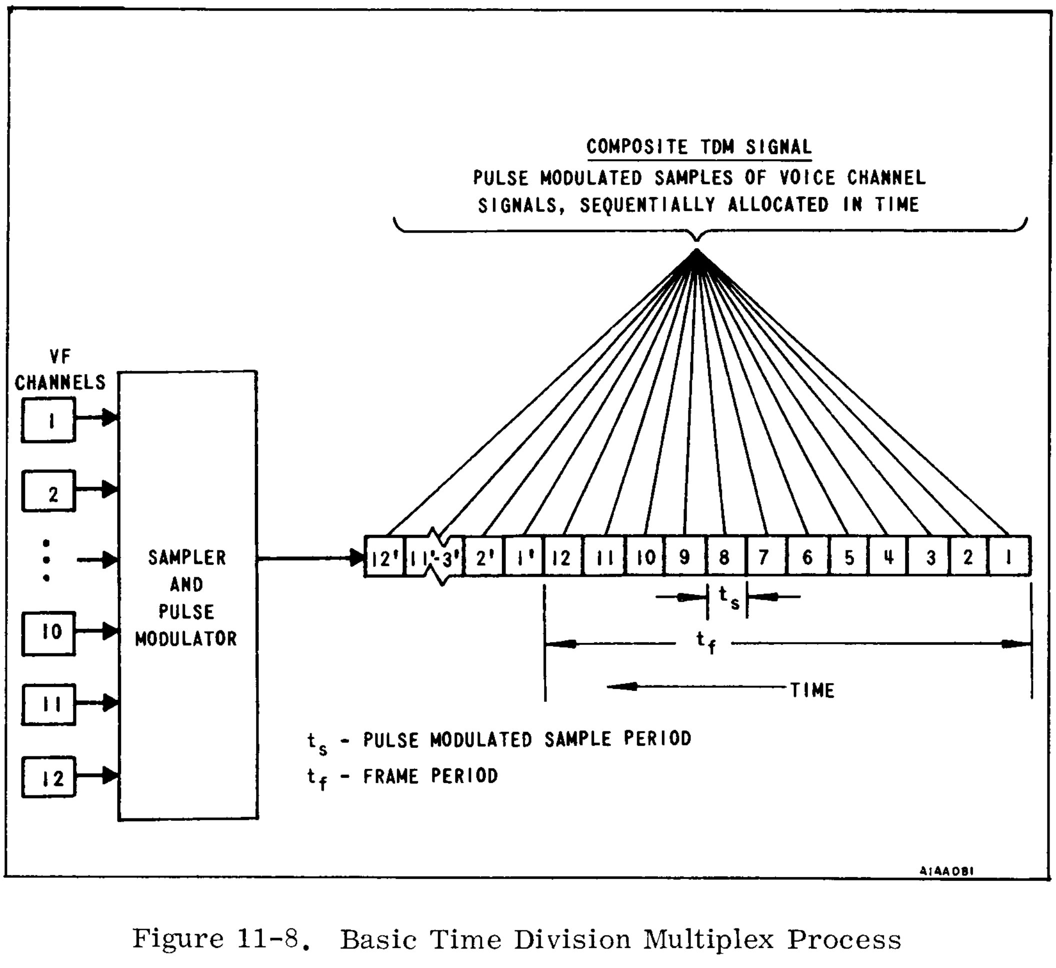


Figure 11-8. Basic Time Division Multiplex Process

Each voice channel occupies a 4 kHz bandwidth. Therefore, a sampler must scan and sample each of the voice channels at twice this figure, i.e., at 8 kHz. One complete set of samples (in this case, 12 samples) is generally called a frame. The time it takes to sample all 12 channels is called the frame period. In this example, the frame period, t_f, is equal to 125 microseconds.

$$t_f = \frac{1}{8\text{ kHz}} = \frac{1}{8 \times 10^3}\text{ seconds} = 125\text{ microseconds}$$

Each sample within a frame period is pulse modulated before it appears sequentially on the composite TDM signal channel. For this example, the time available to represent the sample in a pulse modulated form is equal to one-twelfth of the frame period. Therefore, the pulse modulated sample period, t_s, is equal to about 10.4 microseconds. The pulse modulated sample period can be broken down into smaller subperiods depending upon the type of pulse modulation used in the TDM subsystem.

$$t_s = \frac{t_f}{12} = \frac{125\text{ microseconds}}{12} = 10.4\text{ microseconds}$$

At the receiving end, the reverse process takes place. Each sample is demodulated to obtain the original voice channel sample. The voice channel samples are filtered and sequentially applied to the corresponding voice channel to yield a restored voice signal.

Pulsed systems of modulation offer an attractive means of providing much greater density of information bits per channel than has been discussed. Consider a pulsed circuit in which the highest modulating frequency is to be 5000 cycles per second, comparable to the modulation band of a broadcast transmitter. The minimum sampling rate for pulse modulating such a wave is 10,000 samples per second, or twice the frequency of the highest modulation frequency. This gives an interval per pulse of 100 microseconds, but pulses need be only one or two microseconds long to transmit the essential information. Thus, if a pulse position modulated (PPM) system has pulses of 1-microsecond duration, and the maximum time displacement at peak modulation is 4.5 microseconds, the allocation of time per pulse must be about 10 microseconds. This leaves 90 microseconds of each pulse interval that is not needed for the channel considered, and that can be allocated to other channels. If, in this instance, we divide the available 100 microseconds into 8 time blocks of 12 microseconds each, we can accommodate eight 5 kc channels, with a 20 percent (2-microsecond) guardband between each channel, plus a 4-microsecond interval in which to transmit a synchronizing pulse. This synchronizing pulse if needed at the receiver to provide a reference by use of which the individual channel pulses may be separated.

A time-multiplexed signal can be obtained by generating the pulses for each individual channel just as if it alone is involved, with the addition of timing circuits to delay the individual channel pulses so that successive channels have a progressive time difference, in the above case a difference of 12 microseconds, and the addition of a 4-microsecond synchronizing pulse prior to each sequence of channel pulses. This mixture of pulses is then used to modulate the carrier, which may be either a wire line, a radio path, or a subcarrier of such transmission media. At the receiver, after demodulation and separation of the subcarriers, the detector-output will consist of a chain of pulses identical with the one at the transmitter. This output passes through a system of time gates, one for each channel, that are controlled by the synchronizing pulse. The control is such that the gate for an individual channel is open only during the 12-microsecond interval associated with that channel.

Other pulsed systems may be time-multiplexed by the same method. If, in a given system, there are "n" time-multiplexed channels on each of "m" subcarriers, the system will have a total number of channels equal to "mn." The total number of channels is an important consideration. However, for a frequency allocation of fixed bandwidth, an increase in the number of channels reduces the bandwidth available per channel and, depending upon the application, this will dictate the maximum number of channels to be generated.

Figure 11-9 shows a time-multiplexed PAM system. Figure 11-10 shows a series of typical waveforms in the TDM/PAM system. In figure 11-9 the commutator connects the input channels in sequence to the sampling circuit at the basic repetition rate established by the clock pulse generator, and separates each series of timed samples with a synchronizing pulse. The resulting pulse train is shown in B, figure 11-10. At the receiver the pulse train in duplicated at the detector output. The pulse-width discriminator isolates the synchronizing signal, and the synchronizing signal initiates a series of time delays corresponding to the channel pulse positions in the pulse train. Each gate circuit receives all pulses at one input, and a gating pulse for a specific channel at the other input opens the gate only, during the time interval that its corresponding signal pulse appears at the multiple pulse input. The output of each gate circuit is thus a sequence of signal pulses sampled from a single channel; this modulated waveform may be recovered by using a low-pass filter or a peak detector.

Any mode of pulse modulation lends itself to TDM. The requirement being that the receiving equipment must be able to separate the several channels so that the sample pulses may be used to reconstruct the original modulating signal with distortion within acceptable limits.

The basic principles of operation of TDM telegraph equipment are similar to that of TDM telephone equipment. DC signals from a number of telegraph loops are assembled (interleaved) sequentially for transmission over a single circuit. However, since DC telegraph signals are already in pulse form, the sampling and pulse modulation required in TDM telephone equipment is not needed. The process is simply one of interleaving telegraph channels into a composite TDM signal that is compatible with the particular transmission subsystem.

In its simplest form, the operation of telegraph TDM equipment is similar to that of parallel-to-serial and serial-to-parallel converters. That is, when sending, parallel inputs are converted to a single serial output. Conversely, when receiving, a single serial input is converted to a number of parallel outputs. The modulation rate of the single serial stream depends on the number of associated telegraph loops and the loop modulation rate. For example, if 16 unit interval signals of 75 bauds are time division multiplexed, the modulation rate of the serial stream would be 16 x 75 = 1200 bauds.

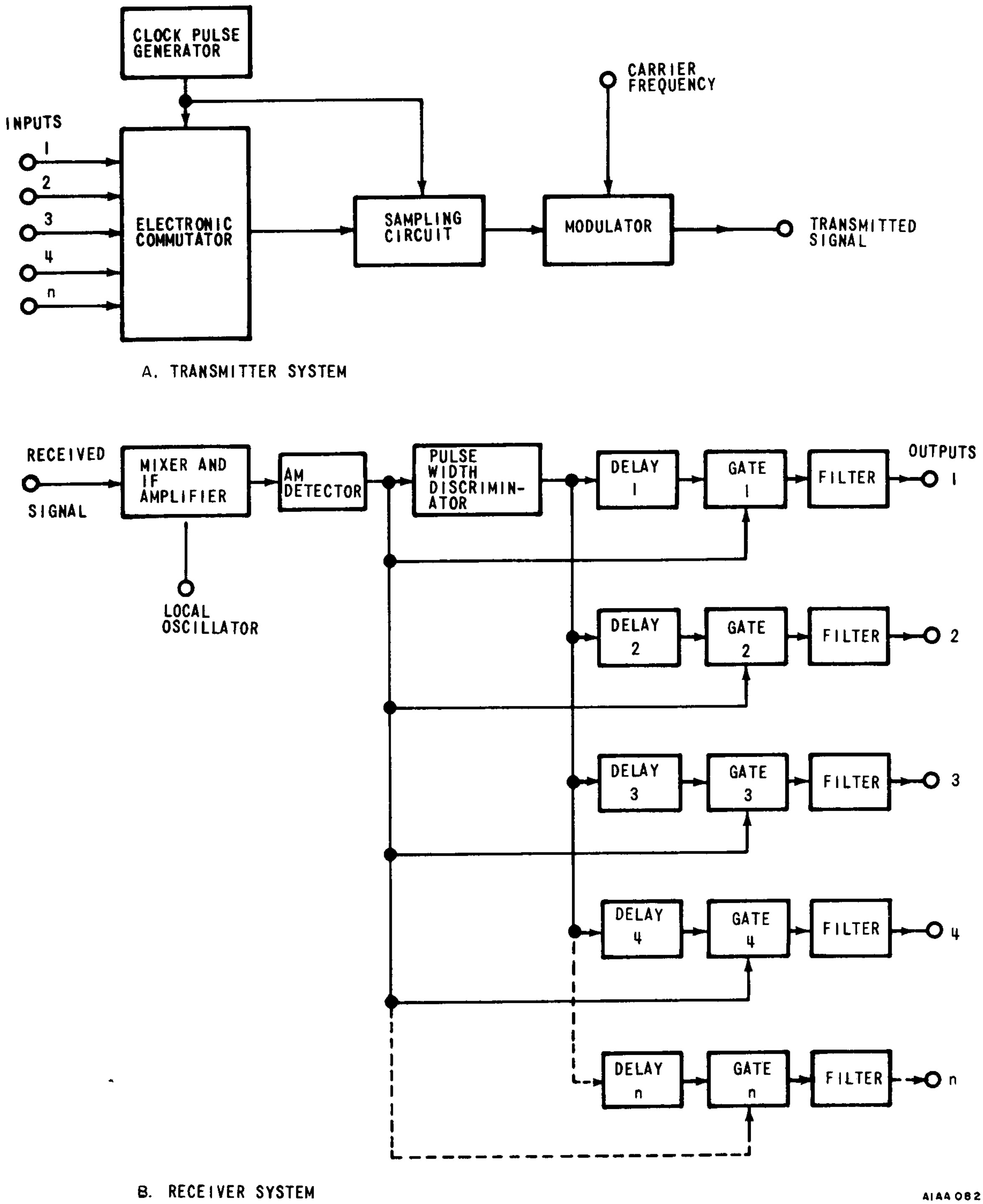


Figure 11-9. Time Multiplexed PAM System

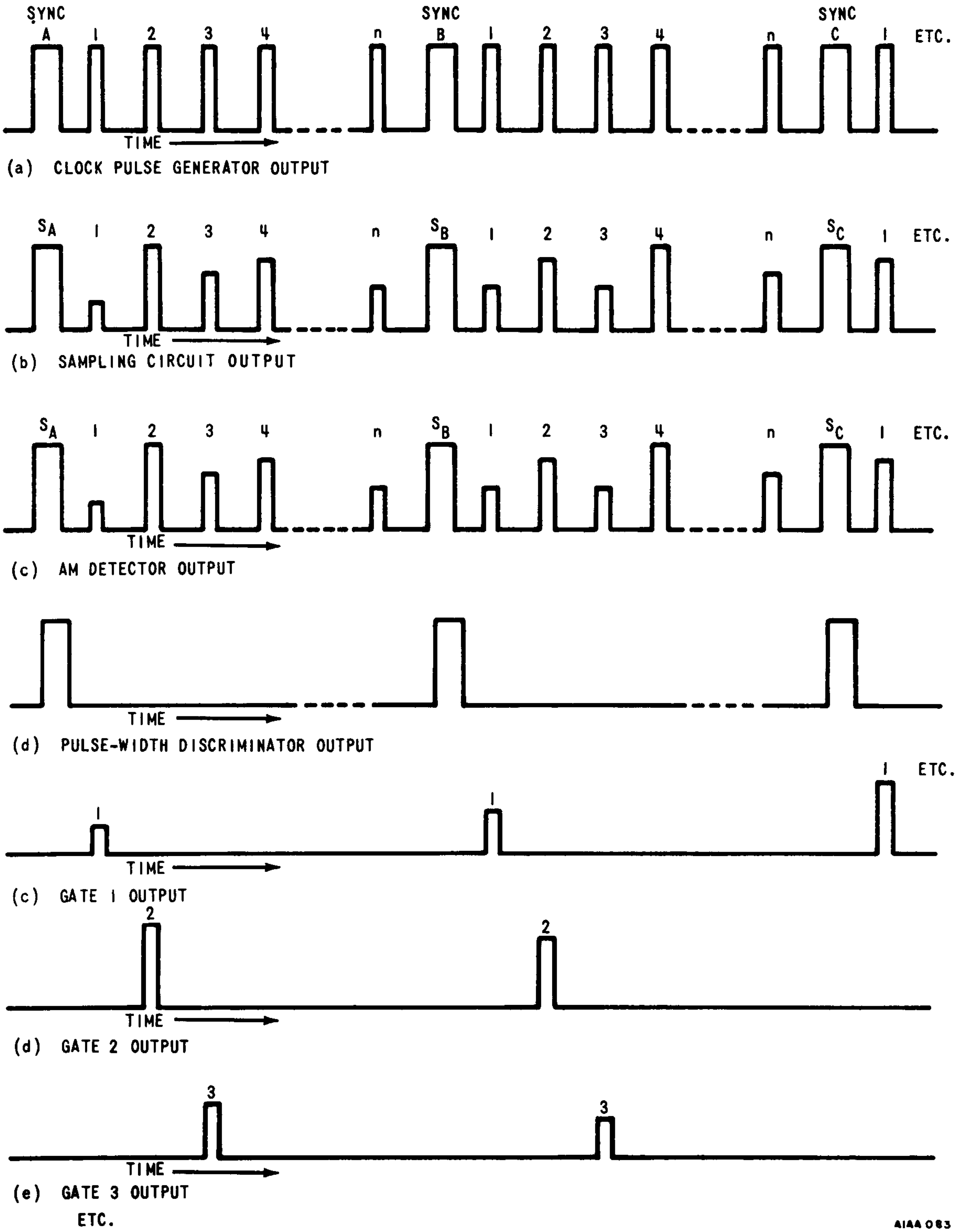


Figure 11-10. Waveforms in the FDM/PAM System

Typical telegraph TDM equipment used in the DCS can handle up to sixteen 60-, 75-, or 100-word-per-minute DC telegraph loops.

11.2.2 Theory of Operation

Earlier it was mentioned that all TDM subsystems are based on the principle of time sharing. Signals are multiplexed by sequentially allocating different time intervals for the transmission of each signal. The signals that are sequentially allocated are not a direct representation of the original signal, but only a sample of the original signal. Before each sample is applied to the common transmission path, some form of pulse modulation is used to form the composite signal. This paragraph will describe in greater detail the principles of pulse modulation techniques that can be applied to TDM subsystems.

With pulse modulation, one or more parameters of a pulse are varied in accordance with a modulating signal to transmit the desired information. The resultant modulated pulse train may then be used to modulate a carrier. This is done with AM or FM techniques, depending upon the transmission subsystem to be used. Pulse modulation that can be used to form composite TDM signals include:

o Pulse amplitude modulation (PAM)	Analog Pulse Modulation
o Pulse duration modulation (PDM)	
o Pulse position modulation (PPM) PTM	
o Pulse frequency modulation (PFM)	
o Pulse code modulation (PCM)	Digital Coding
o Delta modulation (DM)	

It should be noted at this point that PAM and PTM are truly analog pulse modulation techniques. Although the latter two, PCM and DM, are commonly referred to as pulse modulation techniques, they are really digital coding techniques. The following paragraphs discuss and compare the various pulse modulation techniques beginning with the concept of sampling.

a. Pulse Modulation Sampling Theorem. Figure 11-11 illustrates the spectrum of a sampled signal. Assume that the signal at (A), plotted as amplitude versus time, has a spectrum (B) which contains negligible energy outside some low-frequency bandwidth f_m. This is actually the case with most communication signals, though the location and width of f_m depends more or less on an arbitrary definition of "negligible energy." If the signal is now multipled by a periodic series of pulses, shown at (C), the product, shown at (E), is called a "sampled signal" and is obtained from (A) by "sampling." The period "T" between sampling pulses is called the "sampling interval" and its reciprocal 1/T is the sampling frequency, shown in (D). Inspection of the sampled spectrum at (F) shows that the free space called "margin" between the shifted

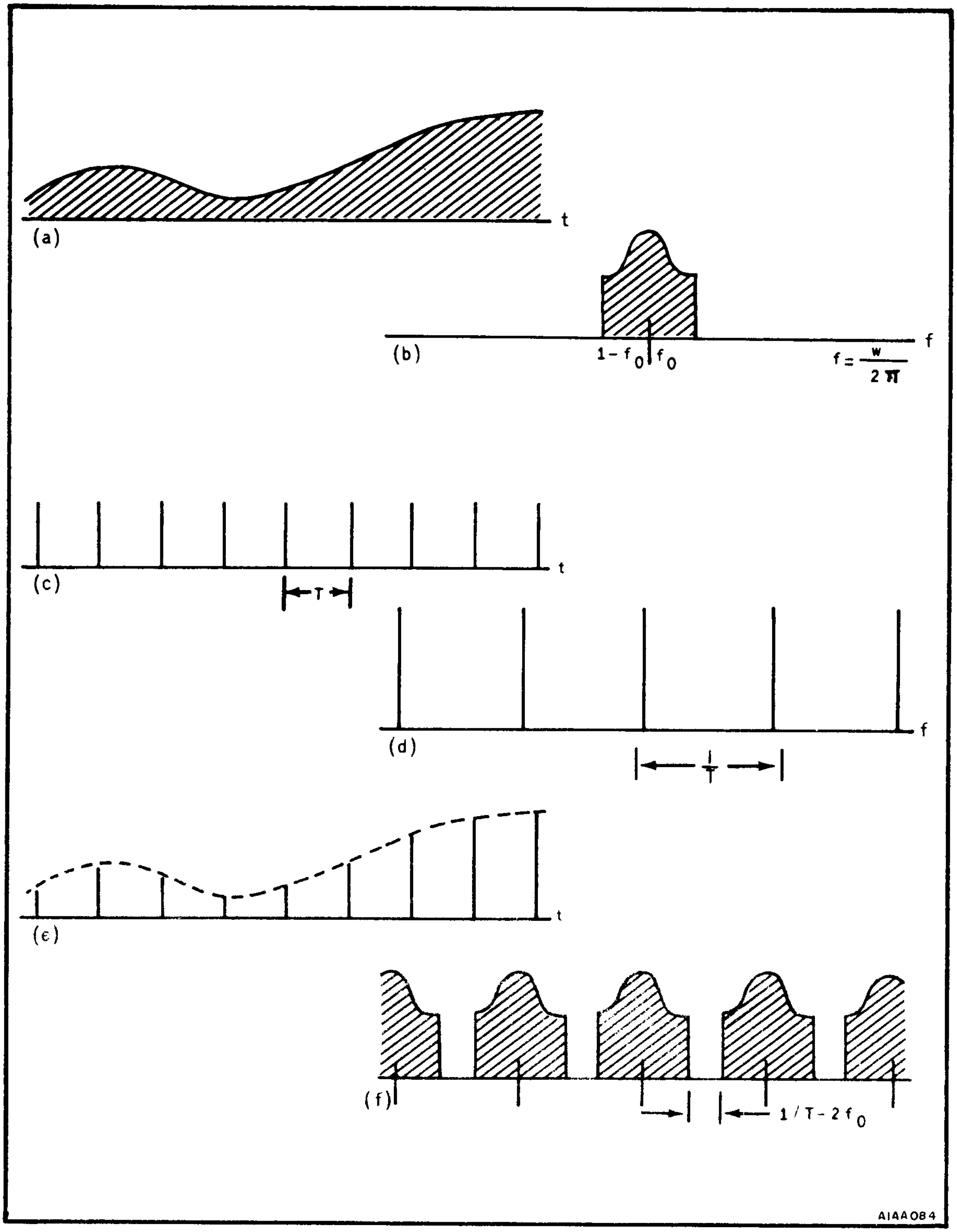


Figure 11-11. Spectral Characteristics of a Sampled Signal

replicas of the original spectrum is equal to the sampling frequency less twice the original frequency spectrum f_m. When such a margin exists -- that is, if the spectral replicas do not overlap - then spectrum (F) contains exactly the same information about the original signal as does the original spectrum (B), and the original signal (A) is recoverable from the sampled signal (E). One method of recovery is to pass the sampled signal through an electric wave filter, which passes frequencies below f_m without distortion, but rejects frequencies above f_m. When the spectral pulses of (F) overlap, the margin is negative and each pulse is contaminated by its neighbors. This contamination represents lost information concerning the original signal, and thus the original signal is no longer completely recoverable. To avoid such negative margin, the sampling frequency must be at least twice the highest modulation frequency. That is, the signal must be sampled at least twice during each cycle of its highest frequency component, in order that it may be recovered without recourse to highly complex circuitry.

b. Pulse Amplitude Modulation (PAM). The sampling pulses of a sampled signal must be varied in some characteristic by the modulating signal, in order for the intelligence of the signal to be present in the pulsed wave. Figure 11-12 shows some of the ways in which pulses may be varied; (A) represents a sine wave of intelligence to be modulated on a transmitted wave; (B) shows the timing pulses which determine the sampling interval; (C) shows Pulse Amplitude Modulation (PAM) in which the amplitude of each pulse is controlled by the instantaneous amplitude of the modulation signal at the time of each pulse. (The other patterns of figure 11-12 will be discussed in subsequent paragraphs, as PAM is the present subject.)

Pulse amplitude modulation may be either unquantized, where the pulse amplitude is varied as a continuous function of the modulation signal, or quantized, where the continuous information to be transmitted is approximated by a finite number of discrete values, one of which is transmitted by each sampling pulse. Quantization will be treated in detail later in the following paragraphs. In figure 11-11 (E) and figure 11-12 (C) are shown examples of unquantized pulse amplitude modulation, in which the amplitudes of the successive pulses are proportional to the instantaneous values of the signal wave, at the time of sampling. It is apparent that these successive pulses will quite faithfully reproduce the signal wave, and that the fidelity of reproduction will be greater with increased sampling frequency. As previously stated, the sampling frequency must be at least twice the highest modulation frequency for full signal recovery.

As indicated in figure 11-11, each harmonic of the pulse frequency is amplitude modulated by the modulating signal, resulting in sideband signals displaced above and below each harmonic by the modulating frequency. After further modulating an RF carrier, either by amplitude or angle modulation methods, transmission and demodulation at a receiver by suitable amplitude or angle detection, the modulation signal may then be recovered by filtering or by peak detection methods.

The basic process of PAM will be applied to a hypothetical 4 channel TDM subsystem. For this example, it will be assumed that each of the four channels is band limited to 4 kHz; therefore, the sampling rate must be at least 8 kHz. Using an 8 kHz sampling rate, each channel is sampled once every 125 microseconds, i.e., each frame is 125

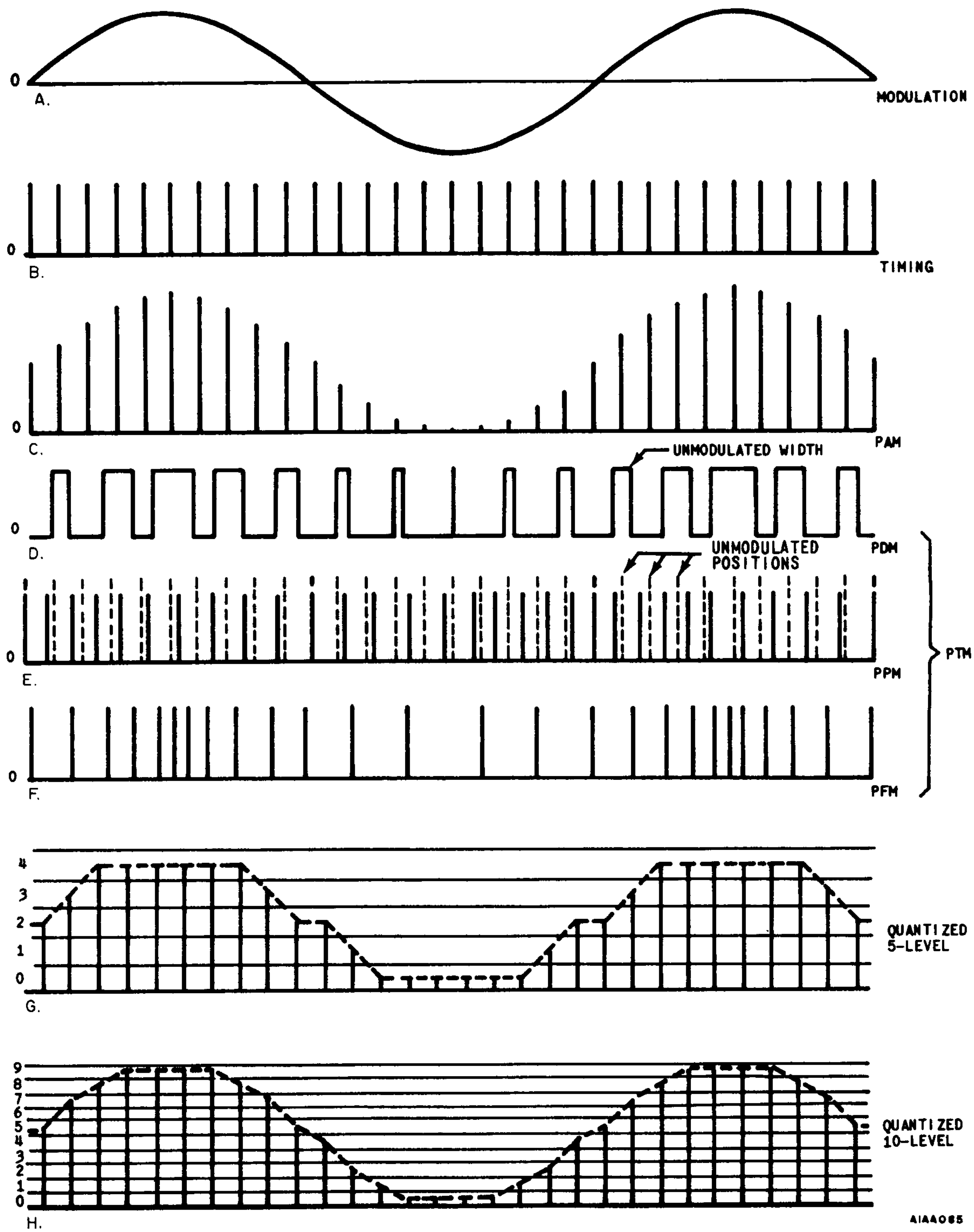


Figure 11-12. Pulse Modulation and Utilization

microseconds long. The channels are sampled sequentially with regard to signal amplitude as shown by the shaded areas S1, S2, S3, et cetera, on figure 11-13. Since there are 4 channels to be sampled during each frame, the time available to represent each signal amplitude in pulse form is 31-1/4 microseconds. As each channel is sequentially sampled, the signal amplitudes at each sampling instant amplitude modulate a repetitive pulse train. This results in the composite TDM signal shown in figure 11-14.

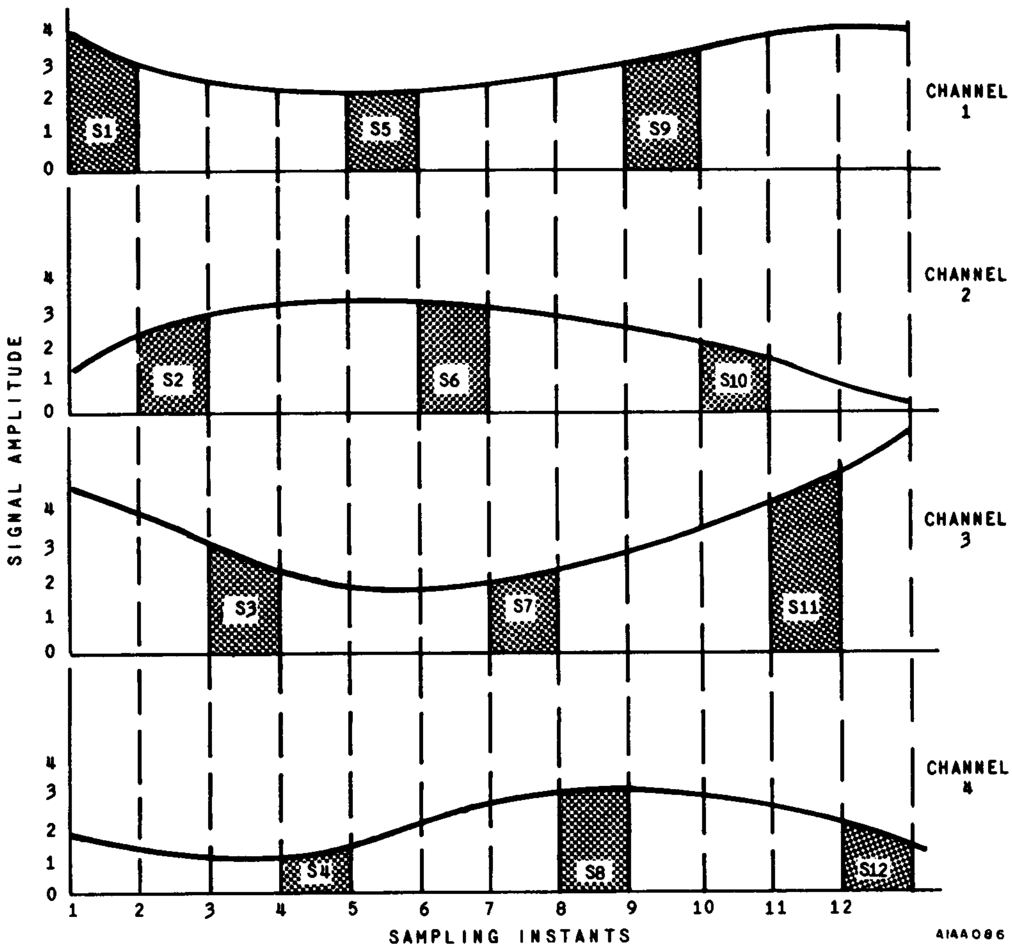
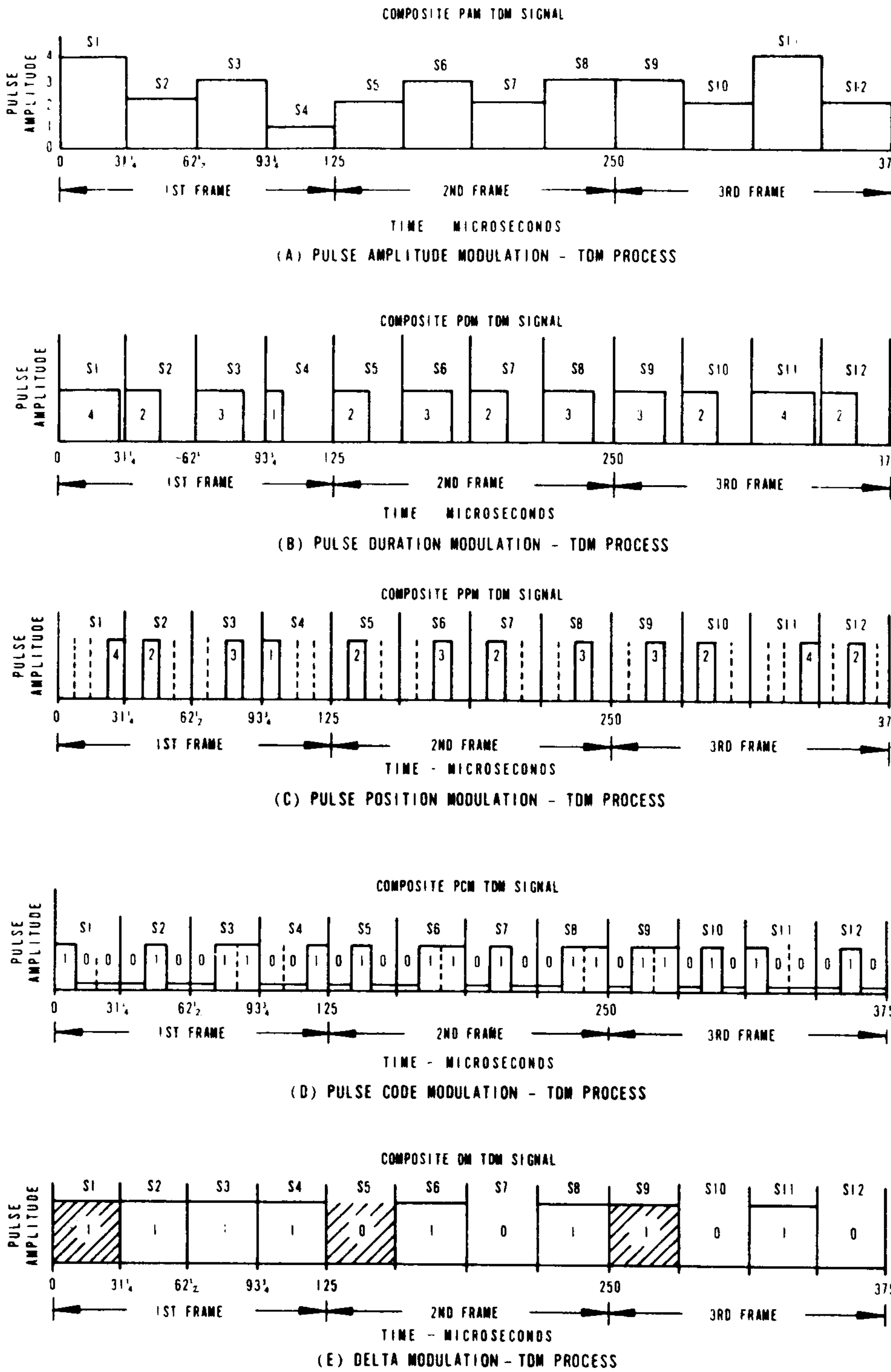


Figure 11-13. Analog Signal Channels



BINARY CODE DIGITAL SEQUENCE	SIGNAL AMPLITUDE
000	0
001	1
010	2
011	3
100	4
101	5
110	6
111	7

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Figure 11-14. Comparison of TDM Processes

c. Pulse Time Modulation (PTM). In pulse modulated systems, as in analog systems, it is possible to impress the intelligence on the carrier by varying any of its characteristics. In the preceding paragraphs, it was discussed how a pulse train was modulated by varying its amplitude. The same intelligence could be used to modify the time characteristic of the pulses. There are two time characteristics which may be affected; the time duration of the pulses, which is called Pulse Duration Modulation (PDM), or Pulse Width Modulation (PWM); and the time of occurrence of the pulses, called Pulse Position Modulation (PPM), and a specific type of PFM called Pulse Frequency Modulation (PFM). Figure 11-12 shows these types of pulse time modulation at (D), (E), and (F).

(1) Pulse Duration Modulation (PDM). Pulse Duration Modulation (PDM), Pulse Width Modulation (PWM), and Pulse Length Modulation (PLM) are all designations for a single type of modulation in which the width of each pulse in a train is made proportional to the instantaneous value of the modulating signal at the instant of the pulse. Either the leading edges, the trailing edges, or both edges of the pulses may be modulated to produce the variation in pulse width. PDM can be obtained in a number of ways, one of which is illustrated in figure 11-15. By adding the modulating signal, figure 11-15 (A), to a repetitive sawtooth, (B), the waveform at (C) is obtained. This waveform is then applied to a circuit which changes state when the input signal exceeds a specific threshold level to produce pulses whose width is determined by the length of time that the input waveform exceeds the threshold level. The resulting pulse train is then as shown at (D), and in figure 11-12 (D).

Figure 11-14(B) illustrates the PDM process as applied to TDM subsystems. In this simplified example, only four distinct signal amplitudes are used, i.e., signal amplitudes of 1, 2, 3, and 4. This was also true in the PAM example described above. Therefore, to represent the four distinct signal amplitudes in PDM form, four distinct pulse durations are required. These can be obtained by designating the full pulse duration to represent a signal amplitude of 4, three-fourths of the pulse duration to represent a signal amplitude of 3, et cetera. This is illustrated in the composite TDM signal shown in figure 11-14(B). It should be noted that the trailing edge of the pulse is modulated in this example.

Demodulation of PDM signals also may be accomplished in a number of ways. Since the average value of the pulse train varies in accordance with the modulation, the same as in the case of PAM, the intelligence may be extracted by passing the width-modulated pulses through a lowpass filter that passes only the desired modulation frequencies.

(2) Pulse Position Modulation (PPM). Pulse Position Modulation (PPM) is a method in which each instantaneous sample of a modulating wave controls the time position of a pulse in relation to the timing of a recurrent reference pulse that coincides with the position of each unmodulated signal pulse. The pulse train is shown in figure 11-12(E) as solid lines and the reference positions are shown as broken lines. PPM can be obtained in several different ways, two of which will be discussed in the following paragraphs. One system uses a method similar to that used to obtain PDM. Figure 11-16 shows the curves for this method. The modulating signal is added to a repetitive sawtooth and a pulse of fixed duration is generated each time the

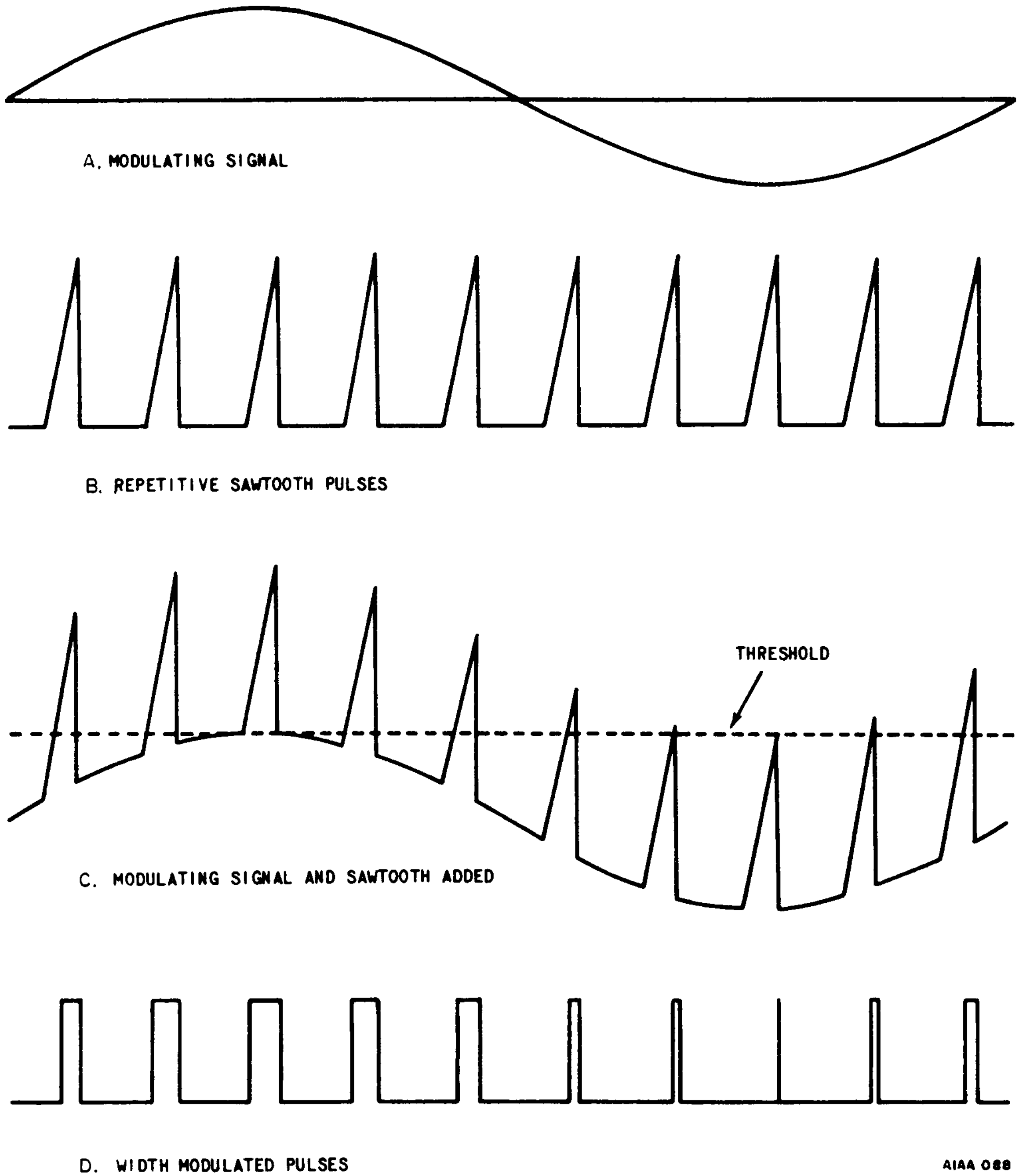

Figure 11-15. Method of Generating PDM

combined signal exceeds a fixed threshold level. PPM can also be obtained by taking a PDM pulse train that has fixed leading or trailing edges, differentiating the pulses and then using a rectifier to separate the pulses having the polarity corresponding to the differentiated modulated edge of the individual width-modulated pulses. This is illustrated in figure 11-17 using PDM with fixed leading and modulated trailing edges. The effect of pulse-position modulation upon the pulse frequency spectrum is to frequency modulate each of the harmonic components of the pulse spectrum as well as the DC term.

When the peak variation in pulse time occurrence is small compared to the interpulse period, PPM can be demodulated by passing the pulse train through a network having a frequency response with a slope of -6 dB per octave throughout the range of modulating frequencies. Alternative methods of demodulating a PPM wave are to convert the wave to either pulse-width or pulse-amplitude form, and then demodulating with a lowpass filter or a peak detector. The process involved in TDM subsystems using PPM is illustrated on figure 11-14(C). Once again, only four distinct signal amplitudes are used. Therefore, four distinct pulse positions are needed. Each sampling interval is broken up into four possible positions. A pulse present in the first position represents a signal amplitude of 1; a pulse in the second position represents a signal amplitude of 2, et cetera. This is illustrated on the composite TDM signal shown on figure 11-14(C).

(3) Pulse Frequency Modulation (PFM). Pulse Frequency Modulation (PFM) is a method of pulse modulation in which the modulating wave is used to frequency modulate a carrier wave consisting of a repetitive pulse train. The resultant pulse train is shown in figure 11-12(F). A comparison of this pulse train with that of the PPM train shown at (E) reveals that PFM is only a variation of PPM and can be demodulated by the same techniques.

d. Quantization. All of the pulse modulation systems discussed provide methods of converting analog wave shapes, such as audio, video, and facsimile systems products, to digital wave shapes; that is, pulses occurring at discrete intervals, some characteristic of which is varied as a continuous function of the analog wave. If the entire range of amplitude (or frequency or phase) values of the analog wave is arbitrarily divided into a series of standard values, and each pulse of a pulse train takes the standard value nearest its actual value when modulated, the modulating wave can be rather faithfully reproduced as shown in figure 11-12 at (G) and (H). The amplitude range has been divided into five standard values at (G), and each pulse is given whatever standard value is nearest its actual instantaneous value. At (H), the same amplitude range has been divided into 10 standard levels, and it is immediately apparent that the curve of (H) is a much closer approximation of the modulating wave (A) than is the five-level quantized curve at (G). From this it is evident that the greater the number of standard levels used, the more closely the quantized wave approximates the original. This is also made evident by the fact that an infinite number of standard levels exactly duplicates the conditions of nonquantization.

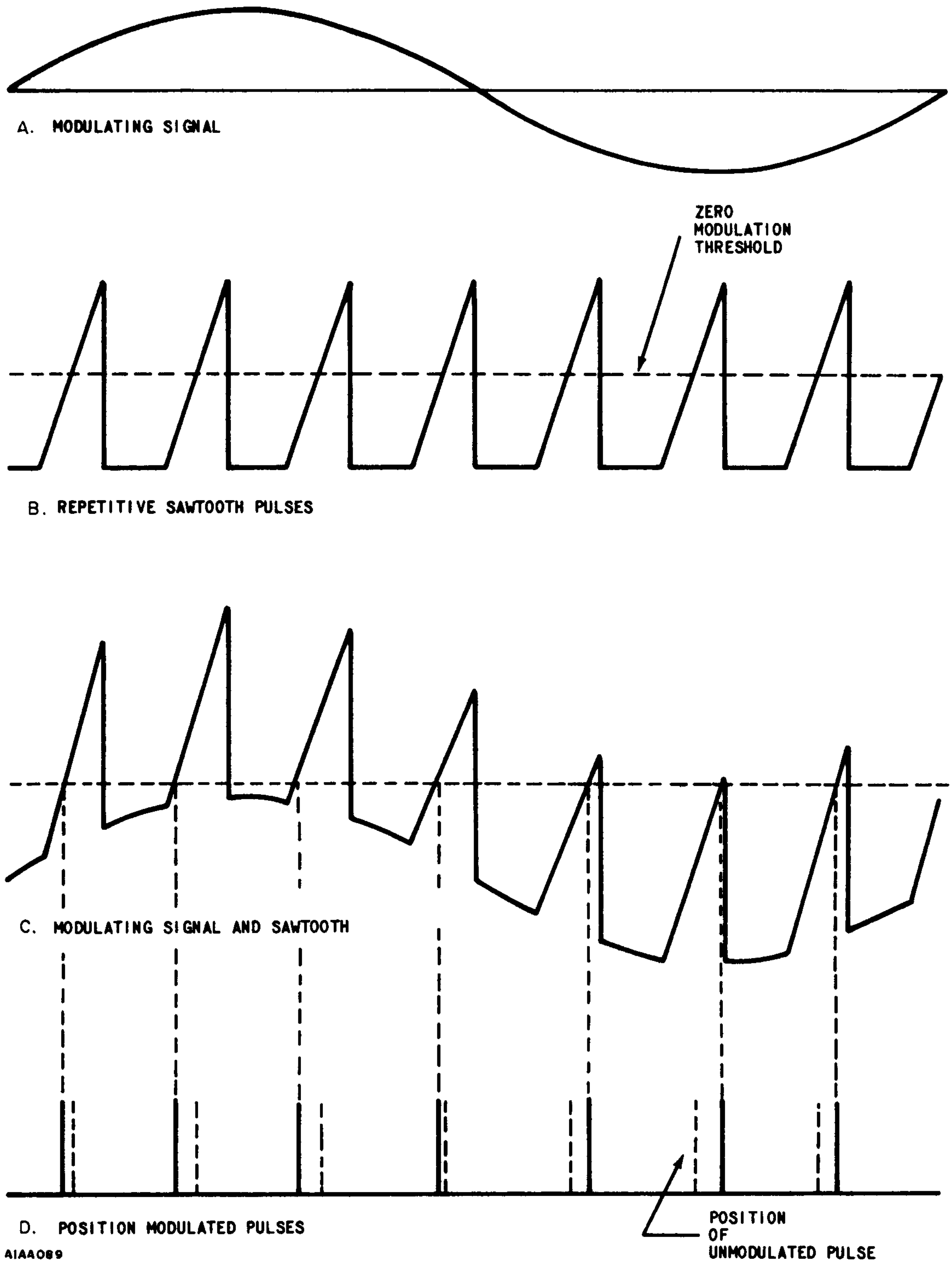

Figure 11-16. Method of Generating PPM

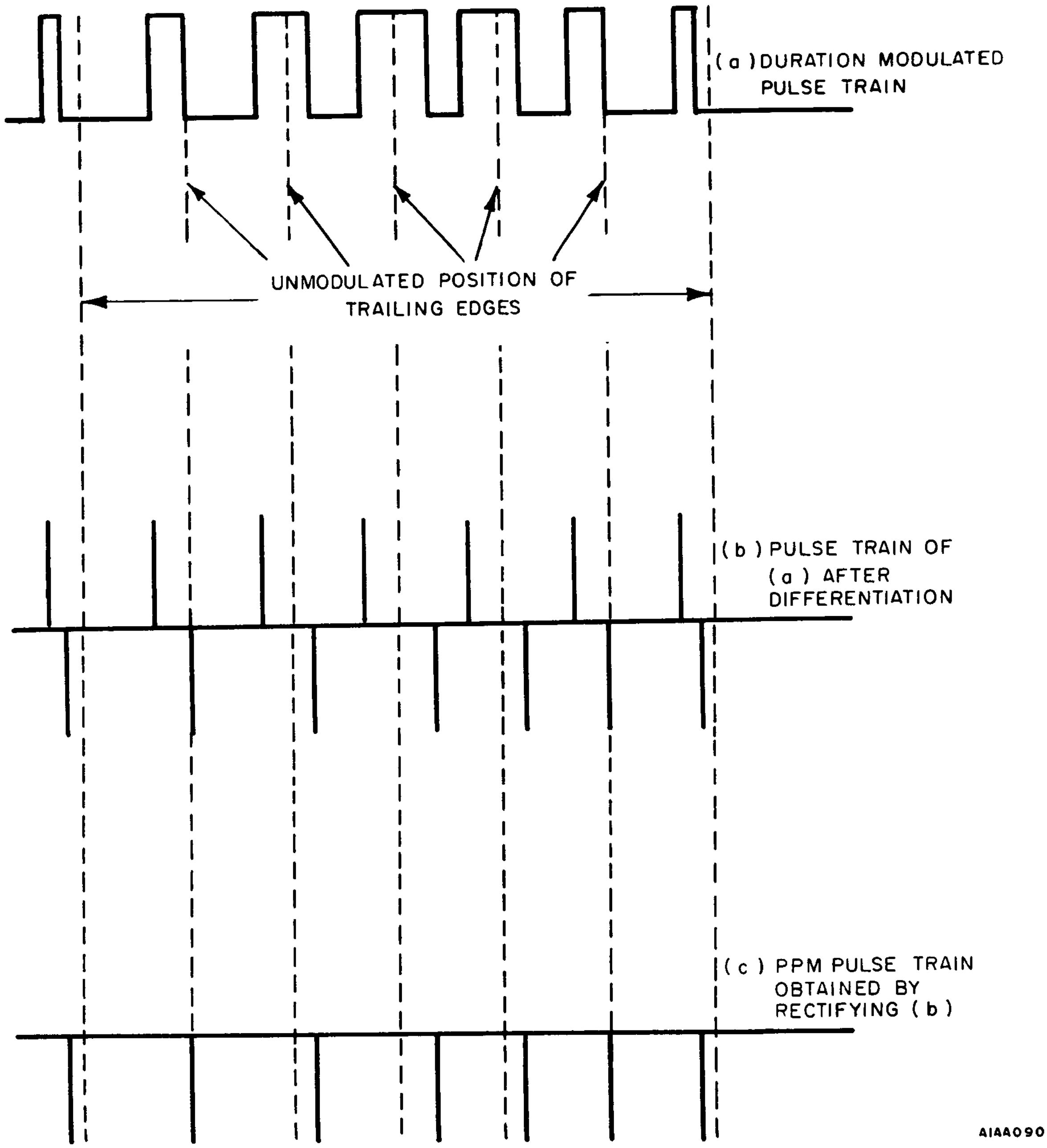

Figure 11-17. PPM Wave Obtained From PDM Wave by Differentiating

Although the quantization curves of figure 11-12 are based on five- and 10-level quantization, in actual practice the levels are usually established at some exponential value of 2, such as $4(2^2)$, $8(2^3)$, $16(2^4)$, $32(2^5)$ ... $N(2^n)$. The reason for selecting levels at exponential values of 2 will become evident in the discussion of pulse code modulation (PCM). Quantized FM is analogous in every way to quantized AM. That is, the range of frequency deviation is divided into a finite number of standard values of deviation, and each sampling pulse results in a deviation equal to the standard value nearest the actual deviation at the sampling instant. Similarly, for phase modulation, quantization establishes a set of standard values. Quantization is used mostly in amplitude- and frequency-modulated pulse systems.

e. Pulse Code Modulation (PCM). Pulse Code Modulation (PCM) refers to a system in which the standard values of a quantized wave are indicated in the modulated wave by a series of coded pulses that, when decoded, indicate the standard values of the original quantized wave so that it may be reconstructed. The codes may be binary, in which the symbol for each quantized element will consist of pulses and spaces; ternary, where the code for each element consists of any one of three distinct kinds or values, such as positive pulses, negative pulses and spaces; of N-ary in which the code for each element consists of any one of N distinct kinds or values. This discussion will be based on the binary PCM systems.

Figure 11-18 shows the relationship between decimal numbers, binary numbers, and a pulse-code waveform that represents the numbers. This is a 16-level code; that is, 16 standard values of a quantized wave could be represented by these pulse groups and only the presence or absence of the pulses are important. The next step up would be a 32-level code, with each decimal number represented by a series of five binary digits, rather than the four of figure 11-18. Six-digit groups would provide a 64-level code; seven digits a 128-level code, et cetera. Figure 11-19 shows the application of pulse coded groups to the standard values of a quantized wave.

In figure 11-19 the solid curve represents the unquantized values of a modulating sinusoid, the dashed curve is reconstructed from the quantized values taken at the sampling interval, and shows a very close agreement with the original curve. Figure 11-20 is identical to figure 11-19 except the sampling interval is four times as great, and the reconstructed curve is no longer so faithful to the original. As previously stated, the sampling rate of a pulsed system must be at least twice the highest modulating frequency in order to get a usable reconstructed modulation curve. At the sampling rate of figure 11-19, and with 4-element binary code, 128 bits (presence or absence of pulse) are transmitted for each cycle of the modulating frequency. At the sampling rate of figure 11-20, only 32 bits are used, and at the minimum sampling rate, only 8 bits are required.

As a matter of convenience, especially to simplify the demodulation of PCM, the pulse trains actually transmitted are reversed from those shown in figures 11-18, 11-19, and 11-20, that is, the pulse with the lowest binary value is transmitted first and the succeeding pulses have increasing binary values up to the code limit. Pulse coding can be performed in a number of ways, using fairly conventional circuitry or by means

DECIMAL NUMBER	BINARY EQUIVALENT				PULSE-CODE WAVEFORMS			
	2^3	2^2	2^1	2^0	2^3	2^2	2^1	2^0
0	0	0	0	0				
1	0	0	0	1				
2	0	0	1	0				
3	0	0	1	1				
4	0	1	0	0				
5	0	1	0	1				
6	0	1	1	0				
7	0	1	1	1				
8	1	0	0	0				
9	1	0	0	1				
10	1	0	1	0				
11	1	0	1	1				
12	1	1	0	0				
13	1	1	0	1				
14	1	1	1	0				
15	1	1	1	1				

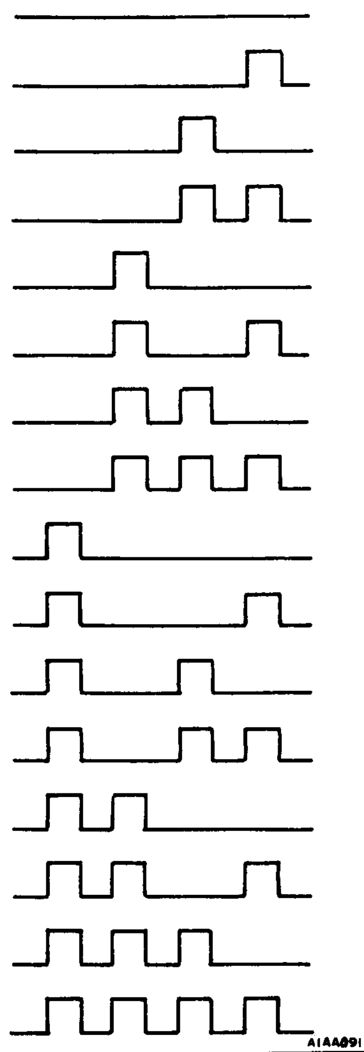

Figure 11-18. Binary Numbers and Waveform Equivalents

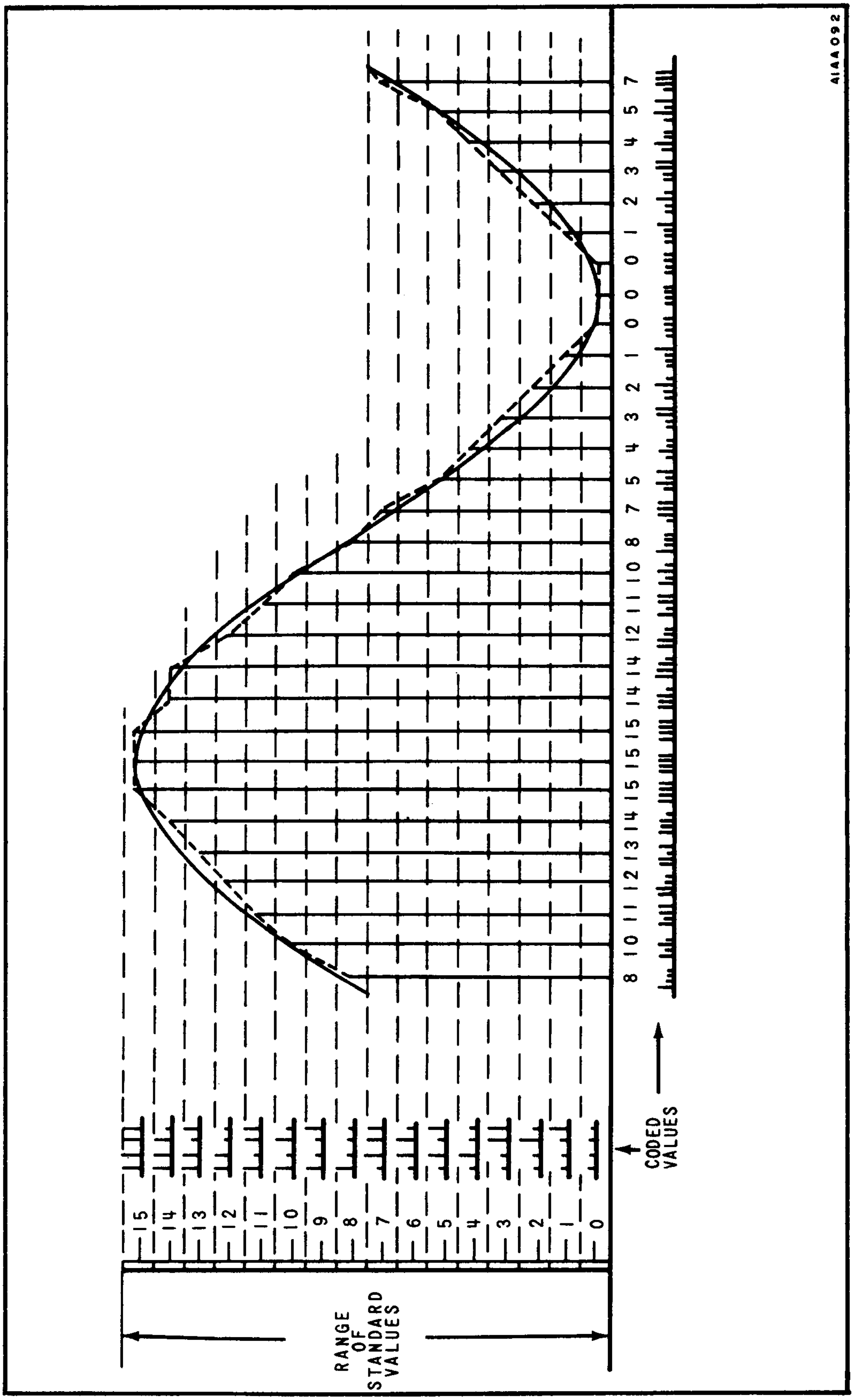

Figure 11-19. Pulse Code Modulation of a Quantized Wave (128 Bits)

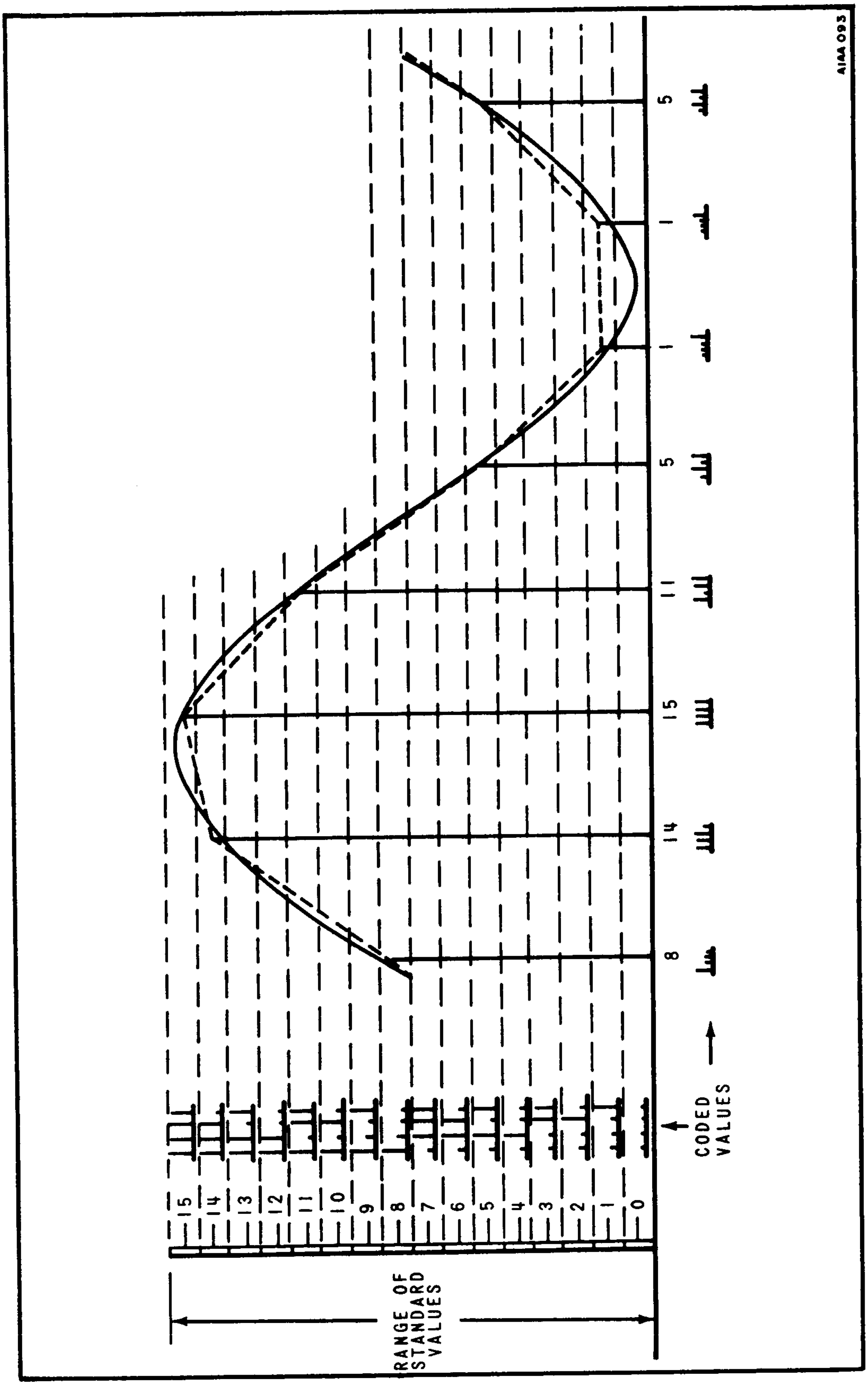

Figure 11-20. Pulse Code Modulation of a Quantized Wave (32 Bits)

of special cathode-ray coding tubes. One form of coding circuit is shown in figure 11-21. In this case, the pulse samples are applied to a holding circuit, a capacitor which stores pulse amplitude information, and the pulse duration modulator converts pulse amplitude to pulse duration. The PDM pulses are then used to gate the output of a precision pulse generator that controls the number of pulses applied to a binary counter. The duration of the gate pulse is not necessarily an integral number of the repetition pulses from the precisely timed clock-pulse generator, so the signal to the binary counter corresponding to each gate pulse may be a number of pulses plus the leading edge of an additional pulse. This "partial" pulse may have sufficient duration to trigger the counter, or it may not. The counter thus responds only to integral numbers, effectively quantizing the signal while encoding it. Each bistable stage of the counter stores a 0 or a 1 for each binary digit it represents (binary 1110) or decimal 7 is shown in figure 11-21. An electronic commutator samples the 2^0, 2^1, 2^2, and 2^3 digit positions in sequence and transmits a mark or space bit (pulse or no pulse) in accordance with the state of each counter stage. The holding circuit is always discharged and reset to zero before initiation of the sequence for the next pulse sample.

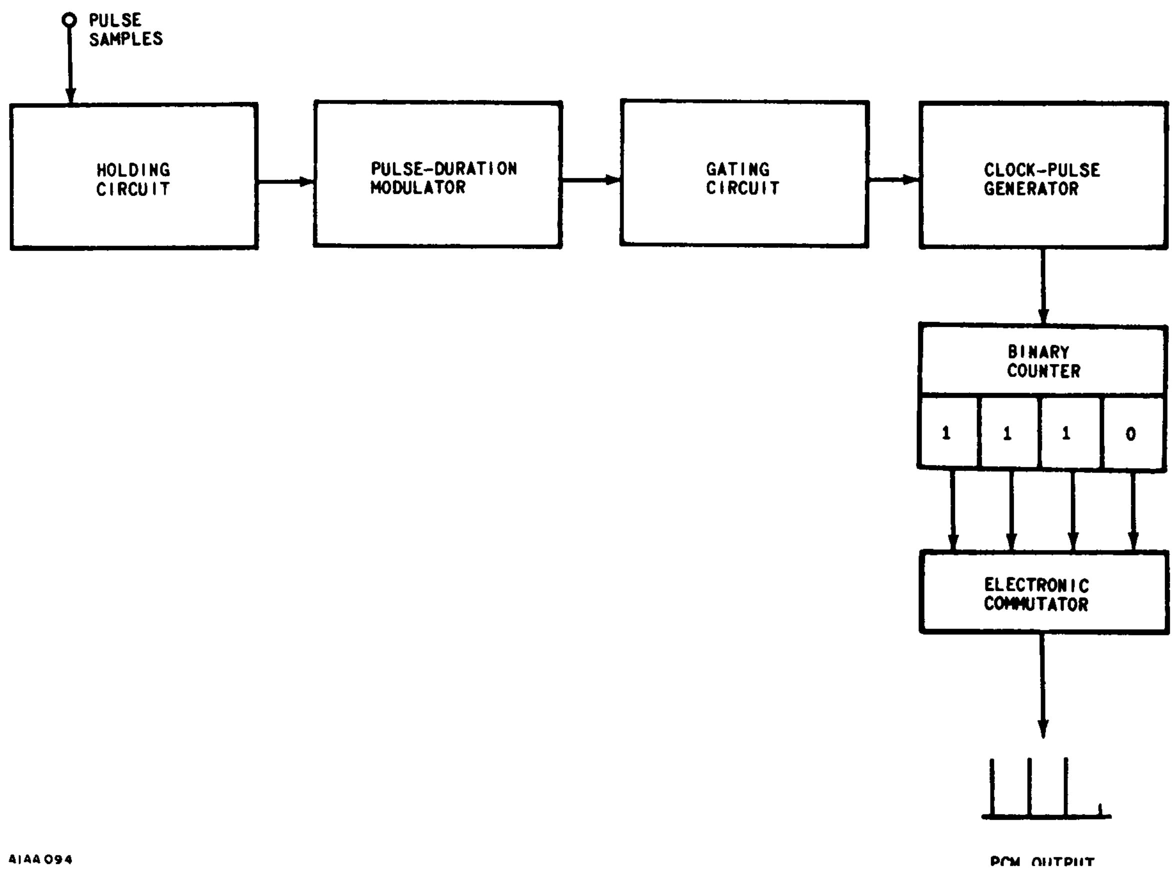

Figure 11-21. Block Diagram of a Quantizer and PCM Coder

At the receiver, after the complex wave has been demodulated and separated into discrete channels, each PCM channel has its code of pulses translated back to its corresponding standard amplitude. This is relatively simple when the pulse-code groups have been transmitted in "reverse" order; that is, if the unit pulse is transmitted first and the pulse with the highest digital value is transmitted last. A current source can be used to apply the pulse code to an RC circuit such as shown in figure 11-22 (A). The time constant of the RC circuit is such that the current leaks off to half-amplitude in the time corresponding to the interval between pulses. This value permits the capacitor charge from one pulse to decay to one-half its original value by the time of the next pulse position and to one-fourth its original value by the time of the second succeeding pulse time. The response to each of the four pulses shown in figure 11-22 (B), considered separately, are shown in (C) and represent binary numbers 0001, 0010, 0100, and 1000, equivalent to 1, 2, 4, and 8 in the decimal system. The peak amplitude of the response to each pulse, the RC time constant, and the sampling time were selected so that the responses would correspond to the decimal value of the voltage. In a linear circuit any number involving two or more pulses will have a total response equal to the sum of the individual responses at any given sampling time, after the last pulse; thus, at (d), the total response is 7 volts, composed of one volt from the first pulse (2^0 = 16 volts halved four times), two volts from the second (2^1 = 16 volts halved three times), and four volts from the third pulse (2^2 = 16 volts halved twice), and nothing from the fourth pulse position (2^3), except an additional time interval during which each of the preceding pulses is halved. The sampling time is not critical because the choice of RC time constant pulse interval assures binary weighting of the 4-digit positions at any time after the charging pulse of the highest binary position.

The process of TDM using PCM is illustrated in figure 11-14 (D). With the binary code, each signal amplitude is represented by a unique digital sequence, as shown on figure 11-14 (D). For example, the signal amplitude at sampling instant S1 is 4. Therefore, the digital sequence would be 100, as shown on the composite TDM in the S1 time interval.

The PCM demodulator will reproduce the correct standard amplitude represented by the pulse-code group provided that it is able to recognize correctly the presence or absence of pulses in each position. For this reason, noise introduces no error at all if the signal-to-noise ratio is such that the largest peaks of noise are not mistaken for pulses. When the noise is of random type (circuit and tube noise), it is possible to determine mathematically, for any ratio of signal-to-average-noise power, the probability of the appearance of a noise peak comparable in amplitude to the pulses. When this is done for 10^5 pulses per second, the approximate error rate for three values of signal-power to average-noise power is:

17 dB	-	10 errors per second
20 dB	-	1 error every 20 minutes
22 dB	-	1 error every 2000 hours

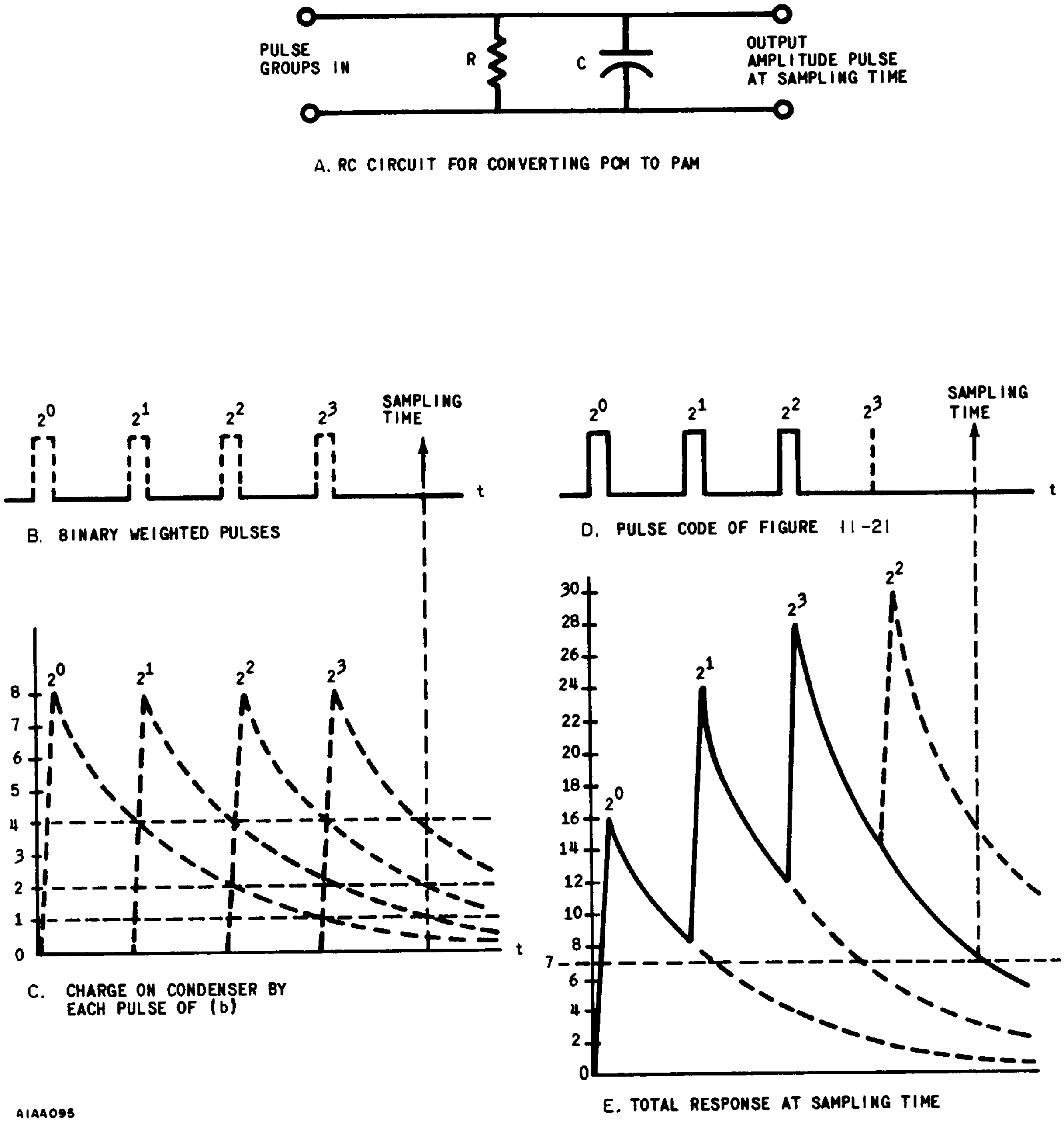

Figure 11-22. Decoding of PCM Pulse Groups

There is evidently a threshold of signal-to-noise ratio of about 20 dB above which virtually no errors occur. In all other systems of modulation, even with signal-to-noise ratios as high as 60 dB, the noise will have some effect. Moreover, the PCM signal can be retransmitted, as in a multiple relay link system, as many times as desired, without the introduction of additional noise effects; that is, noise is not cumulative at relay stations, as with other modulation systems.

There is, of course, the distortion introduced by quantizing the signal, since the standard values selected and the sampling interval both tend to make the reconstructed wave depart from the original. This distortion, called quantizing noise, is initially introduced at the quantizing and coding modulator and remains fixed throughout the transmission and retransmission processes. Its magnitude can be reduced by making the standard quantizing levels closer together. The relationship of the quantizing noise to the number of digits in the binary code is given approximately by the following relationship:

$$\frac{\text{Peak Signal Power}}{\text{Average quantizing noise power where n is the number of digits in the binary code}} = (10.8 + 6n) \text{ dB}$$

Thus, with the 4-digit code of figure 11-19 and figure 11-20, the quantizing noise will be about 35 dB weaker than the peak signal which the channel will accommodate.

The advantages of Pulse Code Modulation are twofold. First, there is the almost complete elimination of noise interference when the pulse signals exceed noise levels by a value of 20 dB or more, since only the presence or absence of each pulse need be determined to find the exact value of the transmitted signal. Second, the signal may be received and retransmitted (relayed) as many times as may be desired without introducing progressive distortion and deterioration of the signal.

f. Delta Modulation (DM). The process of delta modulation is based on the comparison of signal amplitudes at two consecutive sampling instants. Based on this comparison, a determination is made as to whether to transmit a pulse, or to inhibit a pulse in a repititive pulse train. In its simplest form, a pulse is inhibited if the comparison shows that the signal level at the present sampling instant has decreased from the last sampling instant. If the comparison shows an increase, a pulse is transmitted.

The DM process as applied to TDM is shown on figure 11-14 (E). For the purpose of illustration, the sequence of pulses for channel 1 will be described. The comparison of the signal amplitude at S5 with that at S1 shows a decrease in signal amplitude. Therefore, during sampling interval S5 on the composite TDM signal, a pulse is inhibited. The comparison of amplitudes at S9 with S5 shows an increase, resulting in the transmission of a pulse during sampling interval S9. This sequence is shown cross-hatched on figure 11-14 (E).

11.3 COMPARISON OF MULTIPLEX TECHNIQUES

In single sideband FDM subsystems, the bandwidth of the composite FDM signal is equal to the number of channels times the bandwidth of each single channel. For example, in a 12 channel system, with 4 kHz bandwidths per channel, the composite signal will have a bandwidth of 48 kHz. In a double sideband FDM system, this bandwidth will be doubled, i.e., 96 kHz. These bandwidths are relatively narrow when compared to that required in practical TDM subsystems. In a 64 level PCM/TDM system handling twelve 4 kHz channels, the composite TDM signal would have frequency components exceeding 576 kHz, or 48 kHz per channel. This is 12 times greater than that required for single sideband FDM, and six times greater than the bandwidth required for double sideband FDM. When the available transmission subsystem bandwidth is restricted due to technical or economic reasons, FDM would be the proper choice. This would be true where the available bandwidth of the transmission subsystem is derived at substantial cost, such as present day tropospheric scatter radio and submarine cable systems. In cases where the available bandwidth is not restricted, TDM may be a better choice, since it is capable of providing better performance with regard to overall circuit noise.

11.3.1 Noise

From a performance standpoint, the noise in FDM systems increases as the system length is increased, i.e., noise is cumulative. The reason for this is that as noise is introduced into the system, it is added to, and amplified with, the signal at all repeater stations and terminals. As the system length is increased, more repeaters and terminals are required, resulting in high overall circuit noise between originating and terminating user terminals.

In TDM systems, the elements (bits) in the composite TDM signal are usually regenerated at each repeater station or terminal. As used here, regeneration refers to the process of generating a "clean" pulse upon receipt of a "noisy" pulse. Noise in TDM systems will stay relatively constant between terminals regardless of distance. This is true as long as the noise in the transmission subsystem links does not exceed the threshold of recognition for proper regeneration of the TDM pulse.

11.3.2 Future Prospects of TDM

The above comments were made with regard to the present state-of-the-art of TDM and applicable pulse modulation techniques. As new techniques in TDM become available, further comparison with FDM will be required. Furthermore, improvements in future transmission subsystems may provide wider bandwidths for transmission of information. If this indeed becomes a reality, the fact that TDM techniques are wasteful of frequency spectrum may become inconsequential.