Unit 3

Chapter 1

3-1. List and briefly describe the four most common methods *of pulse transmission.*

Answer:-

PULSE MODULATION

Pulse modulation consists essentially of sampling analog information signals and then converting those samples into discrete pulses and then transporting the pulses from a source to a destination over a physical transmission medium. The four predominant methods of pulse modulation include *pulse width modulation* (PWM), *pulse position modulation* (PPM), *pulse amplitude modulation* (PAM), and *pulse code modulation* (PCM).

1. *PWM:-* This method of pulse modulation is sometimes called *pulse duration modulation* (PDM) or *pulse length modulation* (PLM), as the width (active portion of the duty cycle) of a constant amplitude pulse is varied proportional to the amplitude of the analog signal at the time the signal is sampled. PWM is shown in Figure 5-1c. As the figure shows, the amplitude of sample 1 is lower than the amplitude of sample 2. Thus, pulse 1 is narrower than pulse 2. The maximum analog signal amplitude produces the widest pulse, and the minimum analog signal amplitude produces the narrowest pulse. Note, however, that all pulses have the same amplitude.

2. *PPM:-* With PPM, the position of a constant-width pulse within a prescribed tint slot is varied according to the amplitude of the sample of the analog signal. PPM is shows in Figure 5-1d. As the figure shows, the higher the amplitude of the sample, the farther to the right the pulse is positioned within the prescribed time slot. The highest amplitude sample produces a pulse to the far right, and the lowest amplitude sample produces a pulse to the far left.

3. *PAM:-* With PAM, the amplitude of a constant-width, constant-position pulse is varied according to the amplitude of the sample of the analog signal. PAM is shown ii Figure 5- le, where it can be seen that the amplitude of a pulse coincides with the amplitude of the analog signal. PAM waveforms resemble the original analog signal more than the waveforms for PWM or PPM.

4. PCM:- With PCM, the analog signal is sampled and then converted to a serial n-bit binary code for transmission. Each code has the same number of bits and requires the same length of time for transmission. PCM is shown in Figure 5-1f.

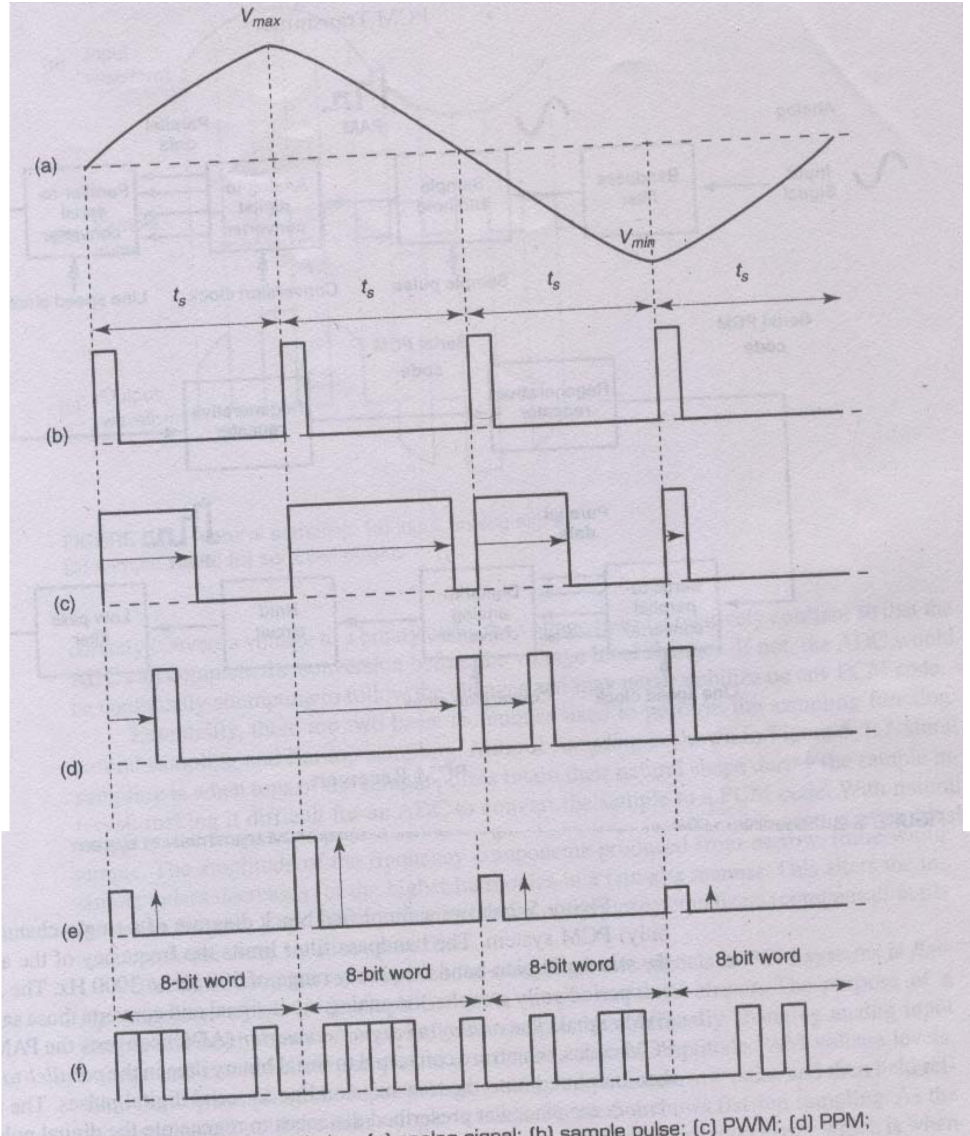


FIGURE 5-1 Pulse modulation: (a) analog signal; (b) sample pulse; (c) PWM; (d) PPM; (e) PAM; (f) PCM

3-2. List and describe the primary components of a single-channel PCM system.
Answer:-

Figure *5-2* shows a simplified block diagram of a single-channel, simplex (one-way only) PCM system. The band pass filter limits the frequency of the analog input signal to the standard voice-band frequency range of 300 Hz to 3000 Hz. The *sample-and-hold* circuit periodically samples the analog input signal and converts those samples to a multilevel PAM signal. The *analog-to-digital converter* (ADC) converts the PAM samples to parallel PCM codes, which are convened to serial binary data in the *parallel-to-serial converter* and then outputted onto the transmission line as serial digital pulses. The transmission line repeaters are placed at prescribed distances to regenerate the digital pulses.

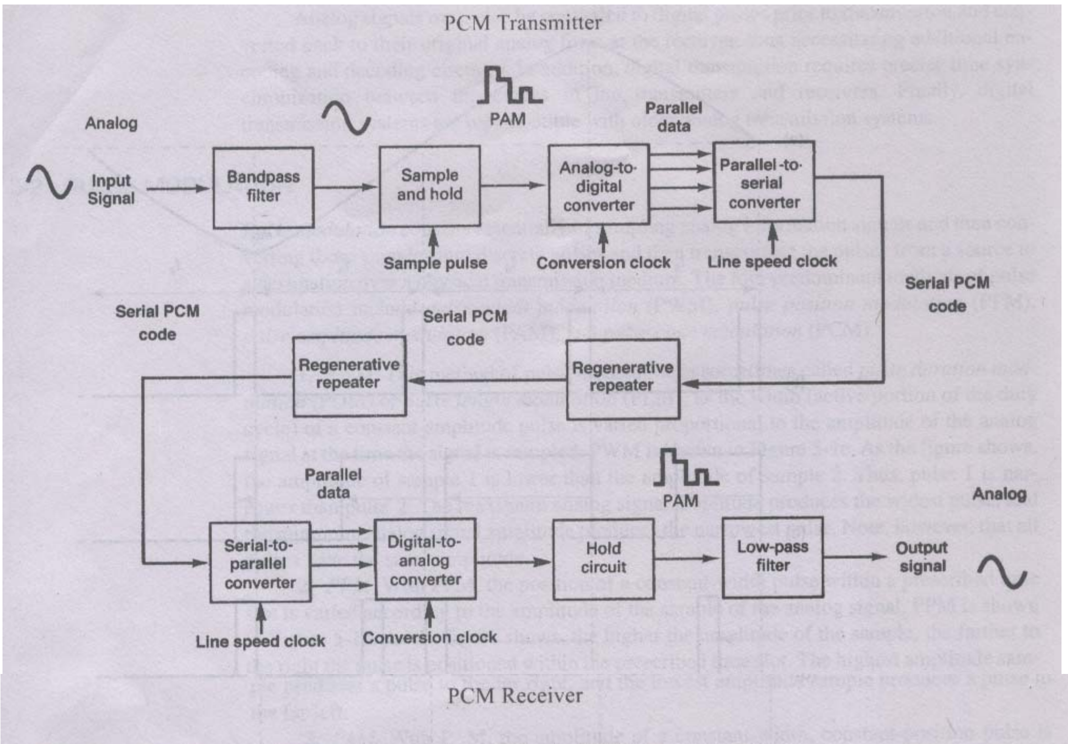


FIGURE 5-2 Simplified block diagram of a single-channel, simplex PCM transmission system

In the receiver, the *serial-to-parallel converter* converts serial pulses received from the transmission line to parallel PCM codes. The *digital-to-analog converter* (DAC) converts the parallel PCM codes to multilevel PAM signals. The hold circuit is basically a low pass filter that converts the PAM signals back to its original analog form.

Figure *5-2* also shows several clock signals and sample pulses that will be explained in later sections of this chapter. An integrated circuit that performs the PCM encoding and decoding functions is called a codec (coder/decoder).

3-3. What is *natural* and *flat-top sampling?*

Answer:-
There are two basic techniques used to perform the sampling function: 1.Natural sampling 2.flat-top sampling. ***Natural sampling*** is shown in Figure 5-3. Natural sampling is when tops of the sample pulses retain their natural shape during the sample interval, making it difficult for an ADC to convert the sample to a PCM code. With natural samplings the frequency spectrum of the sampled output is different from that of an ideal sample. The amplitude of the frequency components produced from narrow- finite-width sample pulses decreases for the higher harmonics in a (sin *x)/x* manner. This alters the information frequency spectrum, requiring the use of frequency equalizers (compensation filters) before recovery by a low-pass filter.

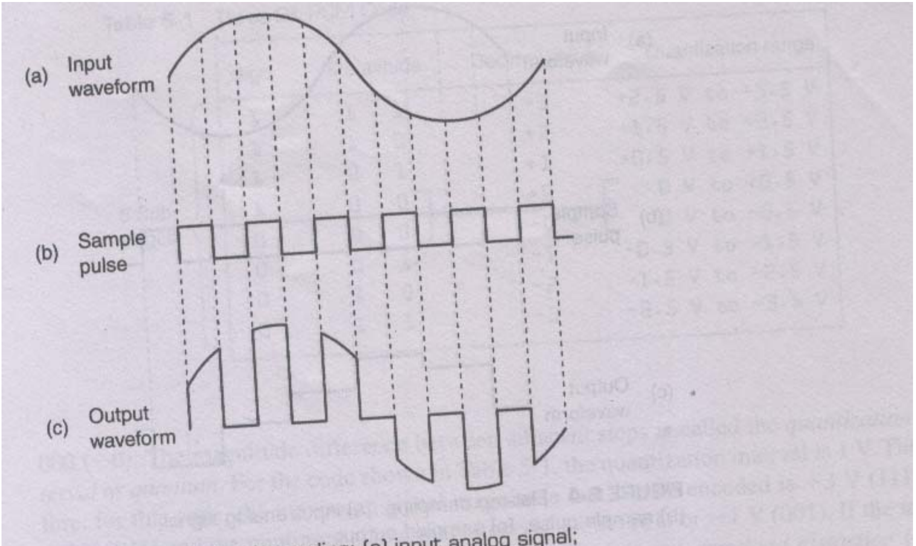


FIGURE 5-3 Natural sampling: (a) input analog signal; (b) sample pulse; (c) sampled output

Flat top sampling:-The most common method used for sampling voice signals in PCM systems is *flat- top sampling,* which is accomplished in a *sample and hold circuit.* The purpose of a sample and hold circuit is to periodically sample the continually changing analog input voltage and convert those samples to a series of constant amplitude PAM voltage levels. With flat-top sampling, the input voltage is sampled with a narrow pulse and then held relatively constant until the next sample is taken.

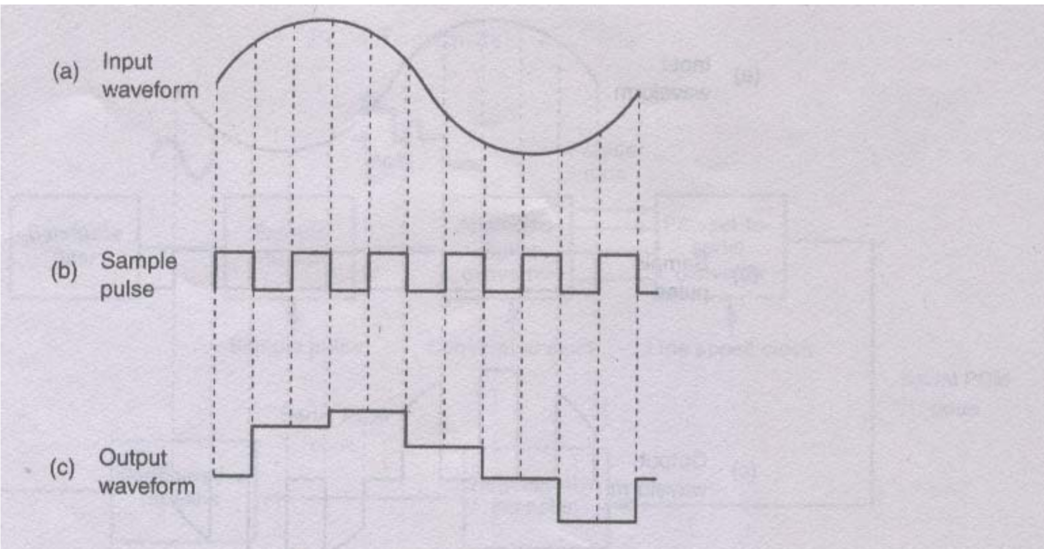


FIGURE 5-4 Flat-top sampling: (a) input analog signal; (b) sample pulse; (c) sampled output

Figure 5-4 shows flat-top sampling. As the figure shows, the sampling process introduces an error called *aperture error,* which is when the amplitude of the sampled signal changes during the sample pulse time. This prevents the recovery circuit in the PCM receiver from exactly reproducing the original analog signal voltage. The magnitude of error depends on how much the analog signal voltage changes while the sample is being taken and the width (duration) of the sample pulse. Flat-top sampling, however, introduces less aperture distortion than natural sampling and can operate with a slower analog to digital converter.

3-4. What is the *Nyquist sampling rate and fold over distortion?*
Answer:-
The *Nyquist sampling theorem* establishes the *minimum sampling rate (f_s)* that can be used for a given PCM system. For a sample to be reproduced accurately in a PCM receiver, each cycle of the analog input signal *(f_a)* must be sampled at least twice. Consequently, the mini mum sampling rate is equal to twice the highest audio input frequency. If f_s is less than two times f_a, an impairment called *alias or fold over distortion* occurs. Mathematically, the mini mum Nyquist sampling rate is

$$f_s \geq 2f_a$$

where f_s = minimum Nyquist sample rate (hertz)

f_a = maximum analog input frequency (hertz)

A sample-and-hold circuit is a nonlinear device with two inputs: the sampling pulse and the analog input signal. Consequently, nonlinear mixing occurs between these two signals. As long as the sampling pulse is at least twice the highest analog input signal, fold over distortion does not occur, and the samples contain all the information from the original signal. However, if the analog input frequency exceeds twice the sampling rate, information is lost, and fold over distortion occurs.

3-5. Define Q*uantization, Quantization interval, folded binary code* Q*uantization interval,* O*verload distortion, Resolution and Quantization error.*
Answer:-

***Quantization**:- It* is the process of converting an infinite number of possibilities to a finite number of conditions. Analog signals contain an infinite number of amplitude possibilities. Thus, converting an analog signal to a PCM code with a limited number of combinations requires quantization. In essence, quantization is the process of rounding off the amplitudes of flat-top samples to a manageable number of levels. For example, a sine wave with peak amplitude of 5 V varies between +5 V and -5, converting samples of a sine wave to PCM requires some rounding off.
Folded binary code:-With quantization, the total voltage range is subdivided into a smaller number of sub- ranges, as shown in Table 5-1.

Table 5-1 Three-Bit PCM Code

	Sign	Magnitude		Decimal value	Quantization range
8 Sub ranges	1	1	1	+3	+2.5 V to +3.5 V
	1	1	0	+2	+1.5 V to +2.5 V
	1	0	1	+1	+0.5 V to +1.5 V
	1	0	0	+0	0 V to +0.5 V
	0	0	0	-0	0 V to -0.5 V
	0	0	1	-1	-0.5 V to -1.5 V
	0	1	0	-2	-1.5 V to -2.5 V
	0	1	1	-3	-2.5 V to -3.5 V

The PCM code shown in Table 5-1 is a three-bit sign- magnitude code with eight possible combinations (four positive and four negative). The leftmost bit is the sign bit (1 = + and 0 = -), and the two rightmost bits represent magnitude. This type of code is called a folded binary code because the codes on the bottom half of the table is a mirror image of the codes on the top half, except for the sign bit. If the negative codes were folded over on top of the positive codes, they would match perfectly. Also, with a folded binary code there are two codes assigned to zero volts: 100 (+0) and 000 (—0).

Q***uantization interval:***-The magnitude difference between adjacent steps is called the *quantization interval* or *quantum.* For the code shown in Table *5-1,* the quantization interval is 1 V. Therefore, for this code, the maximum signal magnitude that can be encoded is +3 V (111) or -3 V (011) and the minimum signal magnitude is +1 V (101) or-1 V (001).

O*verload distortion*:- If the magnitude of the sample exceeds the highest quantization interval, *overload distortion* (also called *peak limiting)* occurs.

Resolution*:-**Assigning PCM codes to absolute magnitudes is called quantizing. The magnitude of a quantum is also called the *resolution.* The resolution is equal to the voltage of the *least significant bit* *(V_{lsb})* of the PCM code, which is the minimum voltage other than 0 V that can be decoded by the digital-to-analog converter in the receiver. The resolution for the PCM code shown in Table 5-1 is 1 V. The smaller the magnitude of a quantum, the better (smaller) the resolution and the more accurately the quantized signal will resemble the original analog sample. ***Quantization error:-Any round-off errors in the transmitted signal are reproduced when the code is converted back to analog by the DAC in the receiver. This error is called the *quantization error (Q_e).* The quantization error is equivalent to additive white noise as it alters the signal amplitude. Consequently, quantization error is also called *quantization noise.*

***3-6*. Explain the relationship between *dynamic range, resolution,* and the number of bits in a PCM code.**

The number of PCM bits transmitted per sample is determined by several variables, which include maximum allowable input amplitude, resolution, and dynamic range. *Dynamic range* (DR) is the ratio of the largest possible magnitude to the smallest possible magnitude (other than 0 V) that can be decoded by the digital to analog converter in the receiver, Mathematically, dynamic range is

$$DR = V_{max} / V_{min} \qquad (5\text{-}2)$$

where *DR* = dynamic range (unit less ratio)

V_{min} = the quantum value (resolution)

V_{max} = the maximum voltage magnitude that can be discerned by the DACs in the receiver .

Equation *5-2* can be rewritten as

$$DR = V_{max}/\text{resolution} \qquad (5.3)$$

The number of bits used for a PCM code depends on the dynamic range. The relationship between dynamic range and the number of bits in a PCM code is

$$2^n - 1 \geq DR \qquad (5.4)$$

and for a minimum number of bits $2^n - 1 = DR$

where , n= number of bits in a PCM code, excluding the sign bit

DR= absolute value of dynamic range

One positive and one negative PCM code is used for 0 V. which is not considered for dynamic range. Therefore,

$2^n = DR + 1$

Dynamic range can be expressed in decibels as

$DR(dB) = 20\log(V_{max}/V_{min})$ or

$DR(dB) = 20\log(2^n - 1)$

where *n* is the number of PCM bits.

3-7. Define *companding.*

Companding is the process of *compressing* and then *expanding. With* companded systems, the higher amplitude analog signals are compressed (amplified less than the lower. amplitude signals) prior to transmission and then expanded (amplified more than the lower. amplitude signals) in the receiver. Companding is a means of improving the dynamic range of a communications system.

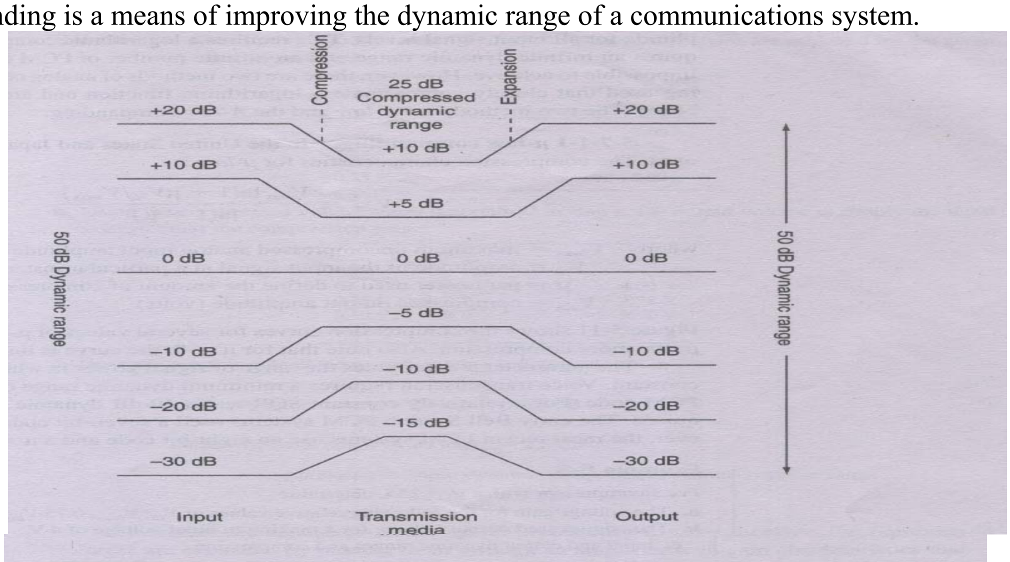


FIGURE 5-9 Basic companding process

Figure 5-9 illustrates the process of companding. An analog input signal with a dynamic range of 50dB is compressed to 25 dB prior to transmission and then in the receiver expanded back to its original dynamic range of 50 dB. With PCM, companding may be accomplished using analog or digital techniques. Early PCM Systems used analog companding, whereas more modern systems use digital companding.

3-8. What does the parameter μ determine?
Answer:-

μ -law companding:- In the United States and Japan, μ -law companding is used. The compression characteristics for μ *-law* is

$$v_{out} = v_{max} \ln\left(1 + \mu \frac{V_{in}}{V_{max}}\right) / \ln(1 + \mu)$$

where V_{max} = maximum uncompressed analog input amplitude (volts)

V_{in}=amplitude of the input signal at a particular instant of time (volts)

μ = parameter used to define the amount of compression (unitless)

V_{out}=compressed output amplitude (volts)

3-9. Briefly explain the process of *digital companding.*

Answer:-

Digital Companding

Digital companding involves compression in the transmitter after the input sample has been converted to a linear PCM code and then expansion in the receiver prior to PCM decoding.

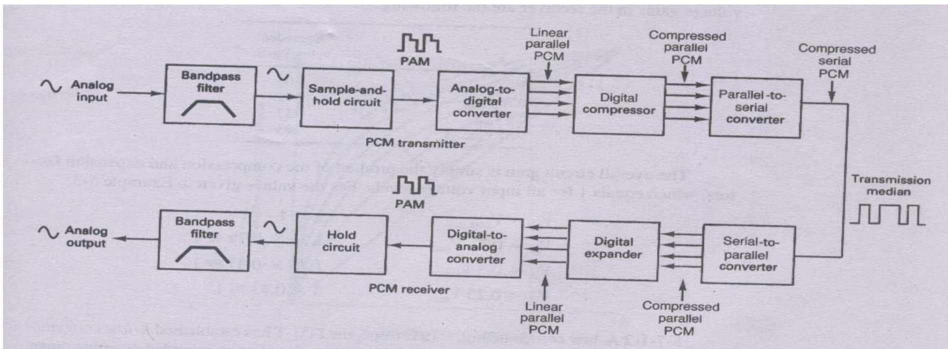


FIGURE 5-12 Digitally companded PCM system

Figure 5-12 shows the block diagram for a digitally companded PCM system. With digital companding, the analog signal is first sampled and converted to a linear PCM code, and then the linear code is digitally compressed. In the receiver, the compressed PCM code is expanded and then decoded (i.e., converted back to analog). The most recent digitally compressed PCM systems use a 12-bit linear PCM code and an eight-bit compressed PCM code. The compression and expansion curves closely resemble the analog μ-law curves with a μ = 255 by approximating the curve with a set of eight straight-line segments (segments 0 through 7). The slope of each successive segment is exactly one-half that of the previous segment.

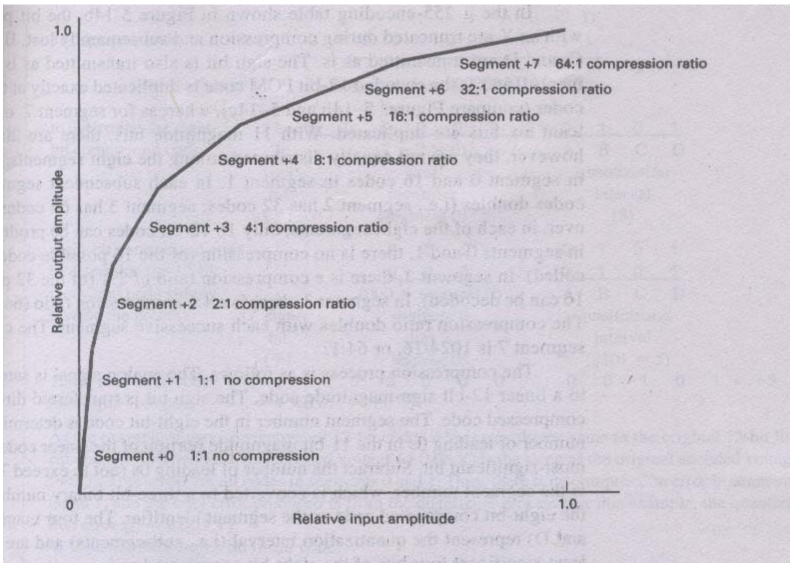
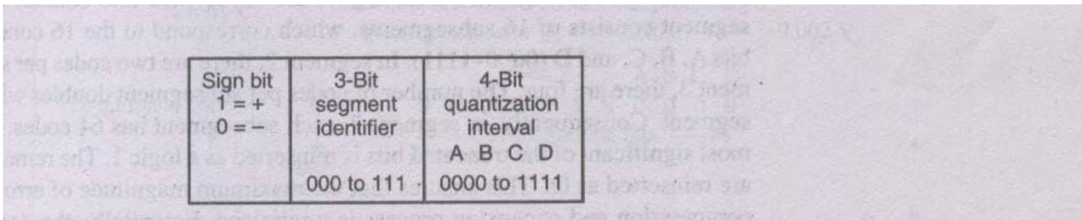


FIGURE 5-13 μ255 compression characteristics (positive values only)

Figure *5-13* shows the 12-to-eight-bit digital compression curve for positive values only. The curve for negative values is identical except the inverse. Although there are 16 segments (eight positive and eight negative), this scheme is often called *I3-segment compression* because the curve for segments +0, +1,-0, and -0 is a straight line with a constant slope and is considered as one segment. The digital companding algorithm for a 12-bit linear to eight-bit compressed code is actually quite simple. The eight-bit compressed code consists of a sign bit, a three-bit segment identifier, and a five-bit magnitude code that specifies the quantization interval within the specified segment (see Figure 'a').



(a)