Couch, II, L.W. “Pulse Code Modulation”
Mobile Communications Handbook
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Pulse Code Modulation[1]

Leon W. Couch, II
University of Florida

3.1 Introduction

Pulse code modulation (PCM) is analog-to-digital conversion of a special type where the information contained in the instantaneous samples of an analog signal is represented by digital words in a serial bit stream.

If we assume that each of the digital words has n binary digits, there are $M = 2^n$ unique code words that are possible, each code word corresponding to a certain amplitude level. Each sample value from the analog signal, however, can be any one of an infinite number of levels, so that the digital word that represents the amplitude closest to the actual sampled value is used. This is called **quantizing**. That is, instead of using the exact sample value of the analog waveform, the sample is replaced by the closest allowed value, where there are M allowed values, and each allowed value corresponds to one of the code words.

PCM is very popular because of the many advantages it offers. Some of these advantages are as follows.

- Relatively inexpensive digital circuitry may be used extensively in the system.
- PCM signals derived from all types of analog sources (audio, video, etc.) may be time-division multiplexed with data signals (e.g., from digital computers) and transmitted over

[1] *Source:* Leon W. Couch, II. 1997. *Digital and Analog Communication Systems*, 5th ed., Prentice Hall, Upper Saddle River, NJ. With permission.

a common high-speed digital communication system.

- In long-distance digital telephone systems requiring repeaters, a *clean* PCM waveform can be regenerated at the output of each repeater, where the input consists of a noisy PCM waveform. The noise at the input, however, may cause bit errors in the regenerated PCM output signal.
- The noise performance of a digital system can be superior to that of an analog system. In addition, the probability of error for the system output can be reduced even further by the use of appropriate coding techniques.

These advantages usually outweigh the main disadvantage of PCM: a much wider bandwidth than that of the corresponding analog signal.

3.2 Generation of PCM

The PCM signal is generated by carrying out three basic operations: sampling, quantizing, and encoding (see Fig. 3.1). The sampling operation generates an instantaneously-sampled flat-top **pulse-amplitude modulated** (PAM) signal.

The quantizing operation is illustrated in Fig. 3.2 for the $M = 8$ level case. This quantizer is said to be *uniform* since all of the steps are of equal size. Since we are approximating the analog sample values by using a finite number of levels ($M = 8$ in this illustration), *error* is introduced into the recovered output analog signal because of the quantizing effect. The error waveform is illustrated in Fig. 3.2c. The quantizing error consists of the difference between the analog signal at the sampler input and the output of the quantizer. Note that the peak value of the error (± 1) is one-half of the quantizer step size (2). If we sample at the Nyquist rate ($2B$, where B is the absolute bandwidth, in hertz, of the input analog signal) or faster and there is negligible channel noise, there will still be noise, called *quantizing noise*, on the recovered analog waveform due to this error. The quantizing noise can also be thought of as a round-off error. The quantizer output is a *quantized* (i.e., only M possible amplitude values) PAM signal.

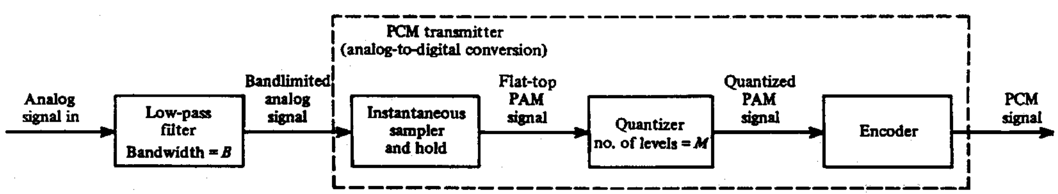

FIGURE 3.1: A PCM transmitter. *Source:* Couch, L.W. II 1997. *Digital and Analog Communication Systems*, 5th ed., Prentice Hall, Upper Saddle River, NJ, p. 138. With permission.

The PCM signal is obtained from the quantized PAM signal by encoding each quantized sample value into a digital word. It is up to the system designer to specify the exact code word that will represent a particular quantized level. If a Gray code of Table 3.1 is used, the resulting PCM signal is shown in Fig. 3.2d where the PCM word for each quantized sample is strobed out of the encoder by the next clock pulse. The Gray code was chosen because it has only 1-b change for each step change in the quantized level. Consequently, single errors in the received PCM code word will cause minimum errors in the recovered analog level, provided that the sign bit is not in error.

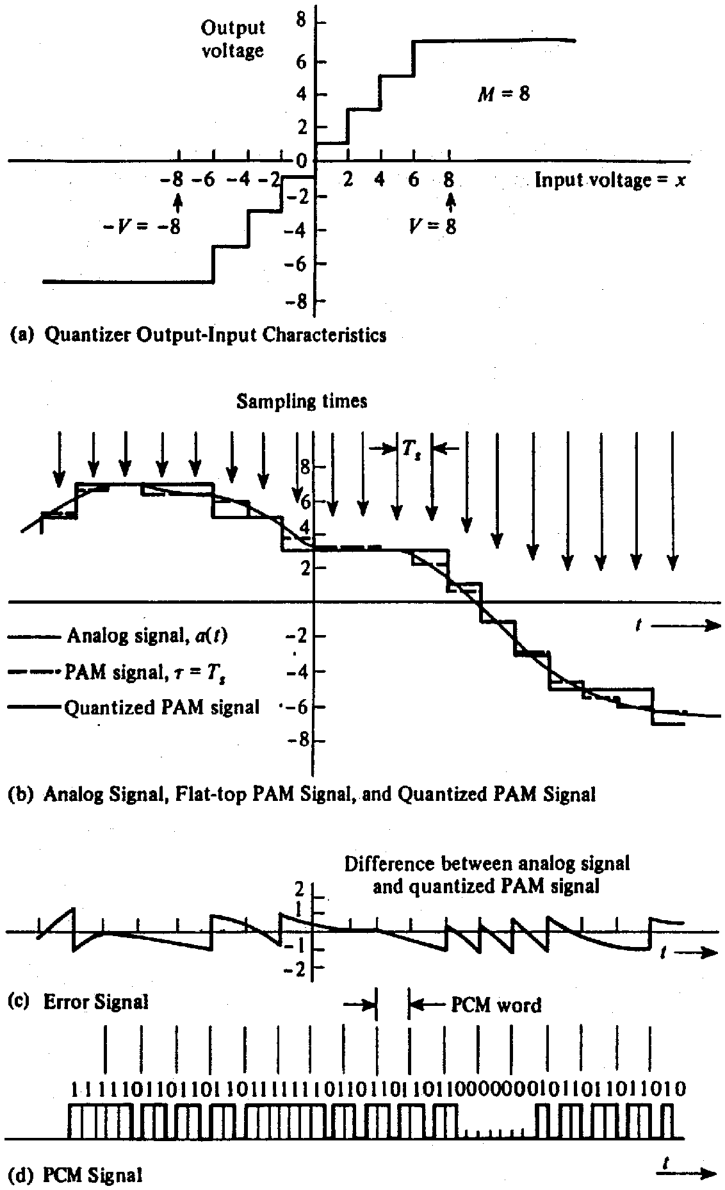

FIGURE 3.2: Illustration of waveforms in a PCM system. *Source:* Couch, L.W. II 1997. *Digital and Analog Communication Systems,* 5th ed., Prentice Hall, Upper Saddle River, NJ, p. 139. With permission.

Here we have described PCM systems that represent the quantized analog sample values by *binary* code words. Of course, it is possible to represent the quantized analog samples by digital words using other than base 2. That is, for base q, the number of quantized levels allowed is $M = q^n$, where n is the number of q base digits in the code word. We will not pursue this topic since binary ($q = 2$) digital circuits are most commonly used.

TABLE 3.1 3-b Gray Code for $M = 8$ Levels

Quantized Sample Voltage	Gray Code Word (PCM Output)	
+7	110	
+5	111	
+3	101	
+1	100	
		Mirror image except for sign bit
−1	000	
−3	001	
−5	011	
−7	010	

Source: Couch, L.W., II. 1997. *Digital and Analog Communication Systems,* 5th ed., Prentice Hall, Upper Saddle River, NJ, p. 140. With permission.

3.3 Percent Quantizing Noise

The quantizer at the PCM encoder produces an error signal at the PCM decoder output as illustrated in Fig. 3.2c. The peak value of this error signal may be expressed as a percentage of the maximum possible analog signal amplitude. Referring to Fig. 3.2c, a peak error of 1 V occurs for a maximum analog signal amplitude of $M = 8$ V as shown Fig. 3.1c. Thus, in general,

$$\frac{2P}{100} = \frac{1}{M} = \frac{1}{2^n}$$

or

$$2^n = \frac{50}{P} \tag{3.1}$$

where P is the peak percentage error for a PCM system that uses n bit code words. The design value of n needed in order to have less than P percent error is obtained by taking the base 2 logarithm of both sides of Eq. (3.1), where it is realized that $\log_2(x) = [\log_{10}(x)]/\log_{10}(2) = 3.32\log_{10}(x)$. That is,

$$n \geq 3.32\log_{10}\left(\frac{50}{P}\right) \tag{3.2}$$

where n is the number of bits needed in the PCM word in order to obtain less than P percent error in the recovered analog signal (i.e., decoded PCM signal).

3.4 Practical PCM Circuits

Three techniques are used to implement the analog-to-digital converter (ADC) encoding operation. These are the *counting* or *ramp*, *serial* or *successive approximation*, and *parallel* or *flash* encoders.

In the counting encoder, at the same time that the sample is taken, a ramp generator is energized and a binary counter is started. The output of the ramp generator is continuously compared to the sample value; when the value of the ramp becomes equal to the sample value, the binary value of the counter is read. This count is taken to be the PCM word. The binary counter and the ramp generator are then reset to zero and are ready to be reenergized at the next sampling time. This technique requires only a few components, but the speed of this type of ADC is usually limited by the speed of the counter. The Maxim ICL7126 CMOS ADC integrated circuit uses this technique.

The serial encoder compares the value of the sample with trial quantized values. Successive trials depend on whether the past comparator outputs are positive or negative. The trial values are chosen first in large steps and then in small steps so that the process will converge rapidly. The trial voltages are generated by a series of voltage dividers that are configured by (on-off) switches. These switches are controlled by digital logic. After the process converges, the value of the switch settings is read out as the PCM word. This technique requires more precision components (for the voltage dividers) than the ramp technique. The speed of the feedback ADC technique is determined by the speed of the switches. The National Semiconductor ADC0804 8-b ADC uses this technique.

The parallel encoder uses a set of parallel comparators with reference levels that are the permitted quantized values. The sample value is fed into all of the parallel comparators simultaneously. The high or low level of the comparator outputs determines the binary PCM word with the aid of some digital logic. This is a fast ADC technique but requires more hardware than the other two methods. The Harris CA3318 8-b ADC integrated circuit is an example of the technique.

All of the integrated circuits listed as examples have parallel digital outputs that correspond to the digital word that represents the analog sample value. For generation of PCM, the parallel output (digital word) needs to be converted to serial form for transmission over a two-wire channel. This is accomplished by using a parallel-to-serial converter integrated circuit, which is also known as a **serial-input-output** (SIO) chip. The SIO chip includes a shift register that is set to contain the parallel data (usually, from 8 or 16 input lines). Then the data are shifted out of the last stage of the shift register bit by bit onto a single output line to produce the serial format. Furthermore, the SIO chips are usually full duplex; that is, they have two sets of shift registers, one that functions for data flowing in each direction. One shift register converts parallel input data to serial output data for transmission over the channel, and, simultaneously, the other shift register converts received serial data from another input to parallel data that are available at another output. Three types of SIO chips are available: the *universal asynchronous receiver/transmitter* (UART), the *universal synchronous receiver/transmitter* (USRT), and the *universal synchronous/asynchronous receiver transmitter* (USART). The UART transmits and receives asynchronous serial data, the USRT transmits and receives synchronous serial data, and the USART combines both a UART and a USRT on one chip.

At the receiving end the PCM signal is decoded back into an analog signal by using a digital-to-analog converter (DAC) chip. If the DAC chip has a parallel data input, the received serial PCM data are first converted to a parallel form using a SIO chip as described in the preceding paragraph. The parallel data are then converted to an approximation of the analog sample value by the DAC chip. This conversion is usually accomplished by using the parallel digital word to set the configuration of electronic switches on a resistive current (or voltage) divider network so that the analog output is produced. This is called a *multiplying* DAC since the analog output voltage is directly proportional to the divider reference voltage multiplied by the value of the digital word. The Motorola MC1408 and the National Semiconductor DAC0808 8-b DAC chips are examples of this technique. The DAC chip outputs samples of the quantized analog signal that approximates the analog sample values. This may be smoothed by a low-pass reconstruction filter to produce the analog output.

The Communications Handbook [6, pp 107–117] and The Electrical Engineering Handbook [5, pp. 771–782] give more details on ADC, DAC, and PCM circuits.

3.5 Bandwidth of PCM

A good question to ask is: What is the spectrum of a PCM signal? For the case of PAM signalling, the spectrum of the PAM signal could be obtained as a function of the spectrum of the input analog signal because the PAM signal is a linear function of the analog signal. This is not the case for PCM. As shown in Figs. 3.1 and 3.2, the PCM signal is a nonlinear function of the input signal. Consequently, the spectrum of the PCM signal is not directly related to the spectrum of the input analog signal. It can be shown that the spectrum of the PCM signal depends on the bit rate, the correlation of the PCM data, and on the PCM waveform pulse shape (usually rectangular) used to describe the bits [2, 3]. From Fig. 3.2, the bit rate is

$$R = nf_s \tag{3.3}$$

where n is the number of bits in the PCM word ($M = 2^n$) and f_s is the sampling rate. For no aliasing we require $f_s \geq 2B$ where B is the bandwidth of the analog signal (that is to be converted to the PCM signal). The dimensionality theorem [2, 3] shows that the bandwidth of the PCM waveform is bounded by

$$B_{\text{PCM}} \geq \frac{1}{2}R = \frac{1}{2}nf_s \tag{3.4}$$

where equality is obtained if a $(\sin x)/x$ type of pulse shape is used to generate the PCM waveform. The exact spectrum for the PCM waveform will depend on the pulse shape that is used as well as on the type of line encoding. For example, if one uses a rectangular pulse shape with polar nonreturn to zero (NRZ) line coding, the first null bandwidth is simply

$$B_{\text{PCM}} = R = nf_s \ \text{Hz} \tag{3.5}$$

Table 3.2 presents a tabulation of this result for the case of the minimum sampling rate, $f_s = 2B$. Note that Eq. (3.4) demonstrates that the bandwidth of the PCM signal has a lower bound given by

$$B_{\text{PCM}} \geq nB \tag{3.6}$$

where $f_s > 2B$ and B is the bandwidth of the corresponding analog signal. Thus, for reasonable values of n, the bandwidth of the PCM signal will be significantly larger than the bandwidth of the corresponding analog signal that it represents. For the example shown in Fig. 3.2 where $n = 3$, the PCM signal bandwidth will be at least three times wider than that of the corresponding analog signal. Furthermore, if the bandwidth of the PCM signal is reduced by improper filtering or by passing the PCM signal through a system that has a poor frequency response, the filtered pulses will be elongated (stretched in width) so that pulses corresponding to any one bit will smear into adjacent bit slots. If this condition becomes too serious, it will cause errors in the detected bits. This pulse smearing effect is called **intersymbol interference** (ISI).

3.6 Effects of Noise

The analog signal that is recovered at the PCM system output is corrupted by noise. Two main effects produce this noise or distortion: 1) quantizing noise that is caused by the M-step quantizer at the PCM transmitter and 2) bit errors in the recovered PCM signal. The bit errors are caused by *channel noise* as well as improper channel filtering, which causes ISI. In addition, if the input analog signal is not strictly band limited, there will be some aliasing noise on the recovered analog signal [12]. Under

TABLE 3.2 Performance of a PCM System with Uniform Quantizing and No Channel Noise

Number of Quantizer Levels Used, M	Length of the PCM Word, n (bits)	Bandwidth of PCM Signal (First Null Bandwidth)[a]	Recovered Analog Signal Power-to-Quantizing Noise Power Ratios (dB) $(S/N)_{out}$
2	1	2B	6.0
4	2	4B	12.0
8	3	6B	18.1
16	4	8B	24.1
32	5	10B	30.1
64	6	12B	36.1
128	7	14B	42.1
256	8	16B	48.2
512	9	18B	54.2
1,024	10	20B	60.2
2,048	11	22B	66.2
4,096	12	24B	72.2
8,192	13	26B	78.3
16,384	14	28B	84.3
32,768	15	30B	90.3
65,536	16	32B	96.3

[a] B is the absolute bandwidth of the input analog signal. *Source:* Couch, L.W. II 1997. *Digital and Analog Communication Systems*, 5th ed., Prentice Hall, Upper Saddle River, NJ, p. 142. With permission.

certain assumptions, it can be shown that the recovered analog *average* signal power to the average noise power [2] is

$$\left(\frac{S}{N}\right)_{out} = \frac{M^2}{1 + 4\left(M^2 - 1\right) P_e} \tag{3.7}$$

where M is the number of uniformly spaced quantizer levels used in the PCM transmitter and P_e is the probability of bit error in the recovered binary PCM signal at the receiver DAC before it is converted back into an analog signal. Most practical systems are designed so that P_e is negligible. Consequently, if we assume that there are no bit errors due to channel noise (i.e., $P_e = 0$), the S/N due only to quantizing errors is

$$\left(\frac{S}{N}\right)_{out} = M^2 \tag{3.8}$$

Numerical values for these S/N ratios are given in Table 3.2.

To realize these S/N ratios, one critical assumption is that the peak-to-peak level of the analog waveform at the input to the PCM encoder is set to the design level of the quantizer. For example, referring to Fig. 3.2, this corresponds to the input traversing the range $-V$ to $+V$ volts where $V = 8$ V is the design level of the quantizer. Equation (3.7) was derived for waveforms with equally likely values, such as a triangle waveshape, that have a peak-to-peak value of $2V$ and an rms value of $V/\sqrt{3}$, where V is the design peak level of the quantizer.

From a practical viewpoint, the quantizing noise at the output of the PCM decoder can be categorized into four types depending on the operating conditions. The four types are overload noise, random noise, granular noise, and hunting noise. As discussed earlier, the level of the analog waveform at the input of the PCM encoder needs to be set so that its peak level does not exceed the design peak of V volts. If the peak input does exceed V, the recovered analog waveform at the output of the PCM system will have flat tops near the peak values. This produces *overload noise*. The flat tops are easily seen on an oscilloscope, and the recovered analog waveform sounds distorted since the flat topping produces unwanted harmonic components. For example, this type of distortion can be heard on PCM telephone systems when there are high levels such as dial tones, busy signals, or off-hook warning signals.

The second type of noise, *random noise*, is produced by the random quantization errors in the PCM system under normal operating conditions when the input level is properly set. This type of condition is assumed in Eq. (3.8). Random noise has a white hissing sound. If the input level is not sufficiently large, the S/N will deteriorate from that given by Eq. (3.8); the quantizing noise will still remain more or less random.

If the input level is reduced further to a relatively small value with respect to the design level, the error values are not equally likely from sample to sample, and the noise has a harsh sound resembling gravel being poured into a barrel. This is called *granular noise.* This type of noise can be randomized (noise power decreased) by increasing the number of quantization levels and, consequently, increasing the PCM bit rate. Alternatively, granular noise can be reduced by using a nonuniform quantizer, such as the μ-law or A-law quantizers that are described in Section 3.7.

The fourth type of quantizing noise that may occur at the output of a PCM system is *hunting noise.* It can occur when the input analog waveform is nearly constant, including when there is no signal (i.e., zero level). For these conditions the sample values at the quantizer output (see Fig. 3.2) can oscillate between two adjacent quantization levels, causing an undesired sinusoidal type tone of frequency $1/2 f_s$ at the output of the PCM system. Hunting noise can be reduced by filtering out the tone or by designing the quantizer so that there is no vertical step at the constant value of the inputs, such as at 0-V input for the no signal case. For the no signal case, the hunting noise is also called *idle channel noise*. Idle channel noise can be reduced by using a horizontal step at the origin of the quantizer output–input characteristic instead of a vertical step as shown in Fig. 3.2.

Recalling that $M = 2^n$, we may express Eq. (3.8) in decibels by taking $10 \log_{10}(\cdot)$ of both sides of the equation,

$$\left(\frac{S}{N}\right)_{\mathrm{dB}} = 6.02n + \alpha \tag{3.9}$$

where n is the number of bits in the PCM word and $\alpha = 0$. This equation—called the 6-dB rule—points out the significant performance characteristic for PCM: an additional 6-dB improvement in S/N is obtained for each bit added to the PCM word. This is illustrated in Table 3.2. Equation (3.9) is valid for a wide variety of assumptions (such as various types of input waveshapes and quantification characteristics), although the value of α will depend on these assumptions [7]. Of course, it is assumed that there are no bit errors and that the input signal level is large enough to range over a significant number of quantizing levels.

One may use Table 3.2 to examine the design requirements in a proposed PCM system. For example, high fidelity enthusiasts are turning to digital audio recording techniques. Here PCM signals are recorded instead of the analog audio signal to produce superb sound reproduction. For a dynamic range of 90 dB, it is seen that at least 15-b PCM words would be required. Furthermore, if the analog signal had a bandwidth of 20 kHz, the first null bandwidth for rectangular bit-shape PCM would be 2×20 kHz $\times 15 = 600$ kHz. Consequently, video-type tape recorders are needed to record and reproduce high-quality digital audio signals. Although this type of recording technique might seem ridiculous at first, it is realized that expensive high-quality analog recording devices are hard pressed to reproduce a dynamic range of 70 dB. Thus, digital audio is one way to achieve improved performance. This is being proven in the marketplace with the popularity of the digital compact disk (CD). The CD uses a 16-b PCM word and a sampling rate of 44.1 kHz on each stereo channel [9, 10]. Reed–Solomon coding with interleaving is used to correct burst errors that occur as a result of scratches and fingerprints on the compact disk.

3.7 Nonuniform Quantizing: μ-Law and A-Law Companding

Voice analog signals are more likely to have amplitude values near zero than at the extreme peak values allowed. For example, when digitizing voice signals, if the peak value allowed is 1 V, weak passages may have voltage levels on the order of 0.1 V (20 dB down). For signals such as these with nonuniform amplitude distribution, the granular quantizing noise will be a serious problem if the step size is not reduced for amplitude values near zero and increased for extremely large values. This is called nonuniform quantizing since a variable step size is used. An example of a nonuniform quantizing characteristic is shown in Fig. 3.3.

The effect of nonuniform quantizing can be obtained by first passing the analog signal through a compression (nonlinear) amplifier and then into the PCM circuit that uses a uniform quantizer. In the U.S., a μ-law type of compression characteristic is used. It is defined [11] by

$$|w_2(t)| = \frac{\ln(1+\mu\,|w_1(t)|)}{\ln(1+\mu)} \tag{3.10}$$

where the allowed peak values of $w_1(t)$ are ± 1 (i.e., $|w_1(t)| \leq 1$), μ is a positive constant that is a parameter. This compression characteristic is shown in Fig. 3.3(b) for several values of μ, and it is noted that $\mu \to 0$ corresponds to linear amplification (uniform quantization overall). In the United States, Canada, and Japan, the telephone companies use a $\mu = 255$ compression characteristic in their PCM systems [4].

Another compression law, used mainly in Europe, is the A-law characteristic. It is defined [1] by

$$|w_2(t)| = \begin{cases} \dfrac{A\,|w_1(t)|}{1+\ln A}, & 0 \leq |w_1(t)| \leq \dfrac{1}{A} \\[2ex] \dfrac{1+\ln(A\,|w_1(t)|)}{1+\ln A}, & \dfrac{1}{A} \leq |w_1(t)| \leq 1 \end{cases} \tag{3.11}$$

where $|w_1(t)| < 1$ and A is a positive constant. The A-law compression characteristic is shown in Fig. 3.3(c). The typical value for A is 87.6.

When compression is used at the transmitter, *expansion* (i.e., decompression) must be used at the receiver output to restore signal levels to their correct relative values. The *expandor* characteristic is the inverse of the compression characteristic, and the combination of a compressor and an expandor is called a *compandor.*

Once again, it can be shown that the output S/N follows the 6-dB law [2]

$$\left(\frac{S}{N}\right)_{\text{dB}} = 6.02 + \alpha \tag{3.12}$$

where for uniform quantizing

$$\alpha = 4.77 - 20\log(V/x_{\text{rms}}) \tag{3.13}$$

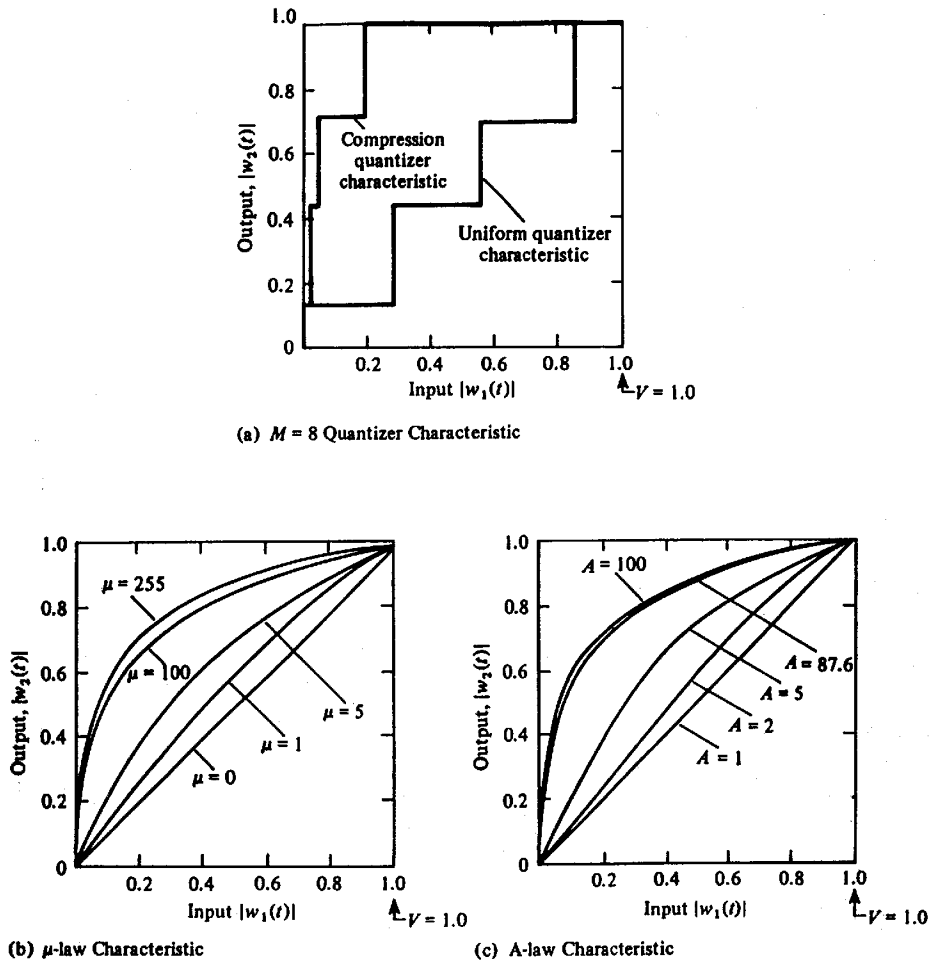

FIGURE 3.3: Compression characteristics (first quadrant shown). *Source:* Couch, L.W. II 1997. *Digital and Analog Communication Systems,* 5th ed., Prentice Hall, Upper Saddle River, NJ, p. 147. With permission.

and for sufficiently large input levels[2] for μ-law companding

$$\alpha \approx 4.77 - 20\log[\ln(1+\mu)] \tag{3.14}$$

and for A-law companding [7]

$$\alpha \approx 4.77 - 20\log[1+\ln A] \tag{3.15}$$

n is the number of bits used in the PCM word, V is the peak design level of the quantizer, and x_{rms} is the rms value of the input analog signal. Notice that the output S/N is a function of the input level

[2]See Lathi, 1998 for a more complicated expression that is valid for any input level.

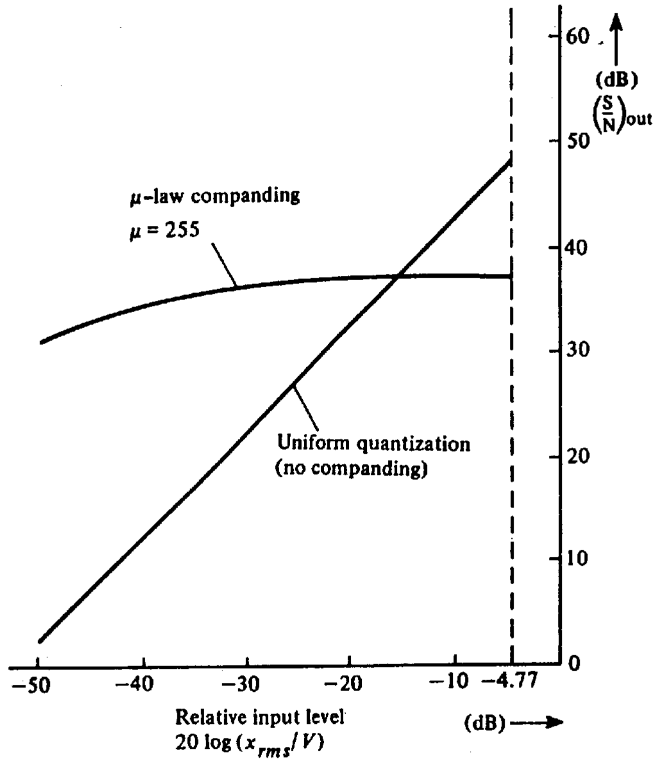

FIGURE 3.4: Output S/N of 8-b PCM systems with and without companding. *Source:* Couch, L.W. II 1997. *Digital and Analog Communication Systems,* 5th ed., Prentice Hall, Upper Saddle River, NJ, p. 149. With permission.

for the uniform quantizing (no companding) case but is relatively insensitive to input level for μ-law and A-law companding, as shown in Fig. 3.4. The ratio V/x_{rms} is called the *loading factor*. The input level is often set for a loading factor of 4 (12 dB) to ensure that the overload quantizing noise will be negligible. In practice this gives $\alpha = -7.3$ for the case of uniform encoding as compared to $\alpha = 0$, which was obtained for the ideal conditions associated with Eq. (3.8).

3.8 Example: Design of a PCM System

Assume that an analog voice-frequency signal, which occupies a band from 300 to 3400 Hz, is to be transmitted over a binary PCM system. The minimum sampling frequency would be $2 \times 3.4 = 6.8$ kHz. In practice the signal is oversampled, and in the U.S. a sampling frequency of 8 kHz is the standard used for voice-frequency signals in telephone communication systems. Assume that each sample value is represented by 8 b; then the bit rate of the PCM signal is

$$\begin{aligned} R &= \left(f_s \text{ samples}/s\right)(n \text{ b}/s) \\ &= (8\,k \text{ samples}/s)(8 \text{ b}/s) = 64 \text{ kb}/s \end{aligned} \tag{3.16}$$

Referring to the dimensionality theorem [Eq. (3.4)], we realize that the theoretically minimum absolute bandwidth of the PCM signal is

$$B_{\min} = \frac{1}{2} D = 32 \text{ kHz} \tag{3.17}$$

and this is realized if the PCM waveform consists of $(\sin x)/x$ pulse shapes. If rectangular pulse shaping is used, the absolute bandwidth is infinity, and the first null bandwidth [Eq. (3.5)] is

$$B_{\text{null}} = R = \frac{1}{T_b} = 64 \text{ kHz} \tag{3.18}$$

That is, we require a bandwidth of 64 kHz to transmit this digital voice PCM signal where the bandwidth of the original analog voice signal was, at most, 4 kHz. Using $n = 8$ in Eq. (3.1), the error on the recovered analog signal is $\pm 0.2\%$. Using Eqs. (3.12) and (3.13) for the case of uniform quantizing with a loading factor, V/x_{rms}, of 10 (20 dB), we get for uniform quantizing

$$\left(\frac{S}{N}\right)_{\text{dB}} = 32.9 \text{ dB} \tag{3.19}$$

Using Eqs. (3.12) and (3.14) for the case of $\mu = 255$ companding, we get

$$\left(\frac{S}{N}\right) = 38.05 \text{ dB} \tag{3.20}$$

These results are illustrated in Fig. 3.4.

Defining Terms

Intersymbol interference: Filtering of a digital waveform so that a pulse corresponding to 1 b will smear (stretch in width) into adjacent bit slots.

Pulse amplitude modulation: An analog signal is represented by a train of pulses where the pulse amplitudes are proportional to the analog signal amplitude.

Pulse code modulation: A serial bit stream that consists of binary words which represent quantized sample values of an analog signal.

Quantizing: Replacing a sample value with the closest allowed value.

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Further Information

Many practical design situations and applications of PCM transmission via twisted-pair T-1 telephone lines, fiber optic cable, microwave relay, and satellite systems are given in [2] and [3].