ANALOG -TO - DIGITAL CONVERSION

The techniques described in previous lecture convert digital data to digital signals. Sometimes, however, we have an analog signal such as one created by a microphone or camera. We have seen that a digital signal is superior to an analog signal. The tendency today is to change an analog signal to digital data. In this section we describe pulse code modulation techniques (PCM). After the digital data are created (digitization), we can use one of the techniques described to convert the digital data to a digital signal.

Pulse Code Modulation (PCM)

The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (**PCM**). A **PCM** encoder has three processes:

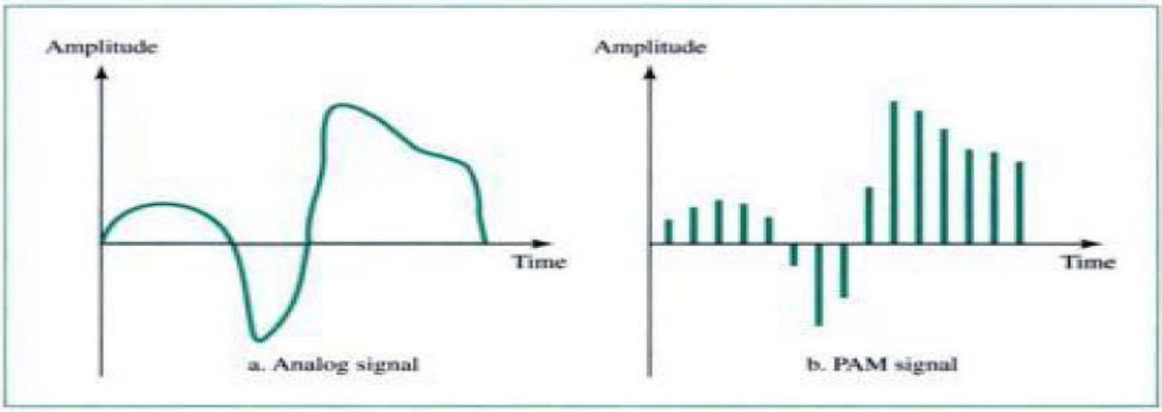
1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

Sampling

The first step in PCM is **sampling**. The term sampling means **measuring the amplitude of the signal at equal intervals**. The analog signal is sampled every **Ts s**, where **Ts** is the sample interval or period.

The sampling process is sometimes referred to as **pulse amplitude modulation** (**PAM**). We need to remember, however, that the result is still an analog signal with nonintegral values. In **PAM** , the original signal is sampled at equal intervals as shown in figure below .

Figure 4.18 *PAM*

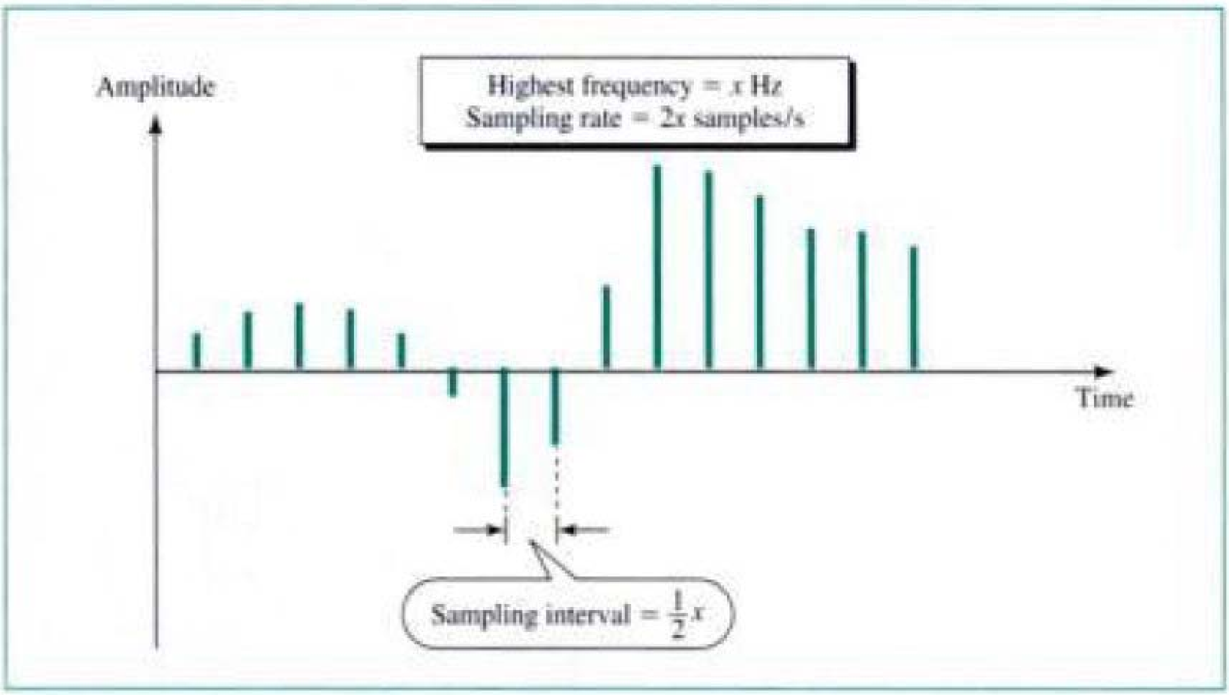


Sampling Rate

One important consideration is the sampling rate or frequency. What are the restrictions on Ts?. According to the **Nyquist theorem**, to reproduce the original analog signal, one necessary condition is that **the sampling rate must be at least twice the highest frequency in the original signal**. So if we want to sample telephone voice with a maximum frequency (4000 Hz) , we need sampling rate at (8000) sample per second.

A sampling rate of twice the frequency of x Hz means that the signal must be sampled every $1/2x$ seconds. Using the voice-over-phone-lines example above, that means one sample every 1/8000 s. Figure 4.23 illustrates the concept.

Figure 4.23 *Nyquist theorem*



Example: What a sampling rate is needed for a signal with a bandwidth of (10000 Hz) (1000 to 11000 Hz)?

Sol :

The sampling rate must be twice the highest frequency in the signal :

Sampling rate = 2 * (11000) = 22000 samples/s.

Example :

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Example

A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

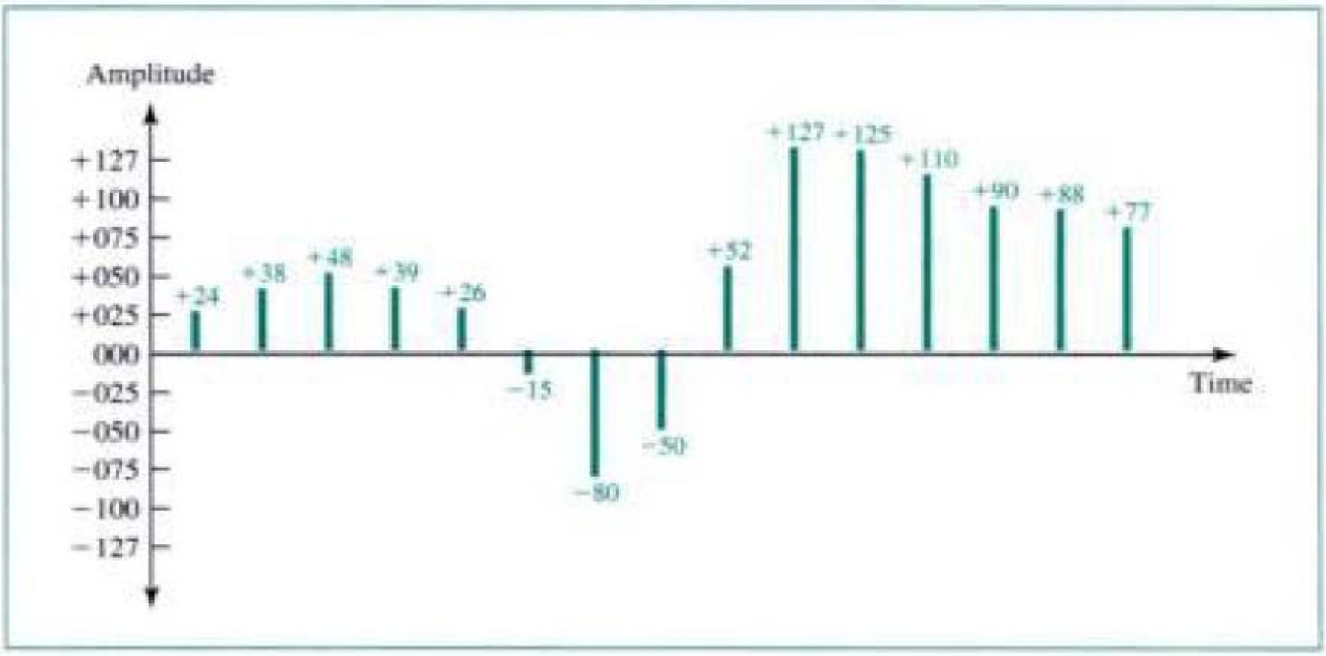
Solution

We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

Quantization

The result of sampling is a series of pulses with amplitude values between the maximum and minimum amplitudes of the signal. PCM quantized the PAM pulses. **Quantization is a method of assigning integral values in a specific range to sampled instances**, the result of quantization is shown in figure below .

Figure 4.19 *Quantized PAM signal*



Encoding

The last step in PCM is encoding. After each sample is quantized and the number of bits per sample is decided, each sample translated to its binary equivalent.

Figure 4.20 shows a simple method of assigning sign and magnitude to quantized samples. Each value is translated into its 7-bit binary equivalent. The eighth bit indicates the sign.

Figure 4.20 *Quantizing by using sign and magnitude*

+024	00011000	−015	10001111	+125	01111101
+038	00100110	−080	11010000	+110	01101110
+048	00110000	−050	10110010	+090	01011010
+039	00100111	+052	00110110	+088	01011000
+026	00011010	+127	01111111	+077	01001101

Sign bit
+ is 0 − is 1

How many Bits per sample?

After we found the sampling rate, we need to determine the number of bits to be transmitted for each sample. This depends on the level of precision needed. The choice of **L**, the number of levels, depends on the range of the amplitudes of the analog signal and how accurately we need to recover the signal. If the amplitude of a signal fluctuates between two values only, we need only two levels; if the signal, like voice, has many amplitude values, we need more quantization levels.

Example:
A signal is sampled, each sample requires at least 12 levels of precision (+0 to +5 , -0 to -5), how many bits should be sent for each sample?

Sol:
We need 4 bits, 1 bit for the sign and 3 bits for the value; a 3 bits value can represent $2^3 = 8$ levels (000 to 111) Which is more than what We need. A 2 bits value is not enough since ($2^2 = 4$). A 4 bits value is too much because ($2^4 = 16$).

Bit Rate

After finding the number of bits per sample, we can calculate the bit rate by using the following formula:

$$\text{Bit rate} = \text{sampling rate} \times \text{number of bits per sample}$$

Example 6

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate is

$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$

The bit rate can be calculated as

$$\text{Bit rate} = \text{sampling rate} \times \text{number of bits per sample} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ Kbps}$$

PCM is actually made up from four separate processes, **PAM**, **Quantization**, **binary encoding** and **Line encoding**. figure below shows the entire process.

Figure 4.22 *From analog signal to PCM digital code*