1.6 PULSE CODE MODULATION (PCM)

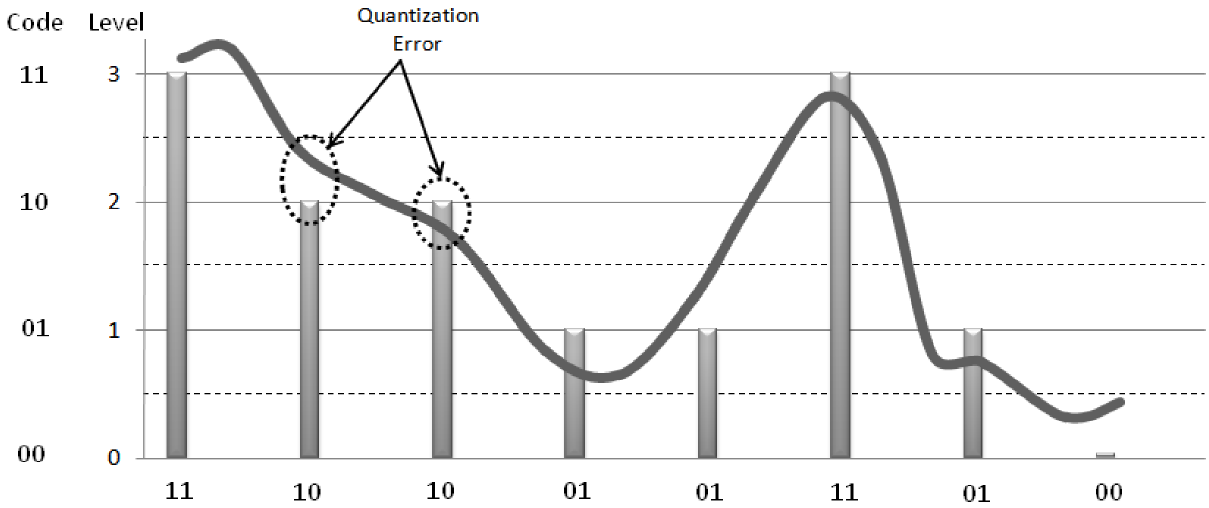
After sampling an analog signal, it is possible to code the samples using discrete symbols, such as binary. The communication through binary alphabets is more efficient than the analog transmission. If the value of a sample is sent using only two possible elements, the reception becomes more reliable. It is easier for the receiver to discriminate the reception into two possible signals than estimating the value of a continuous signal. This significantly helps in reducing the effects of distortion and of additive noise.

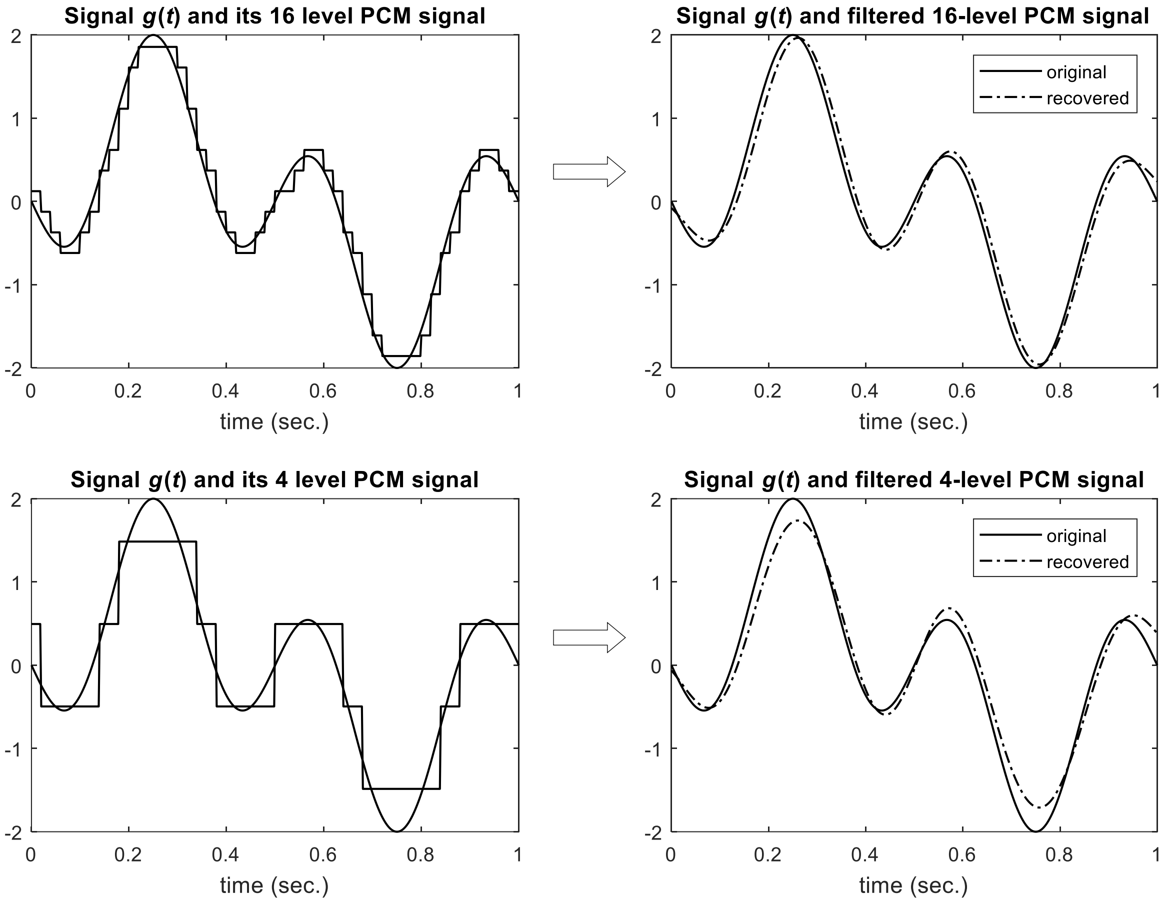
In PCM, the sample values are rounded to the values of certain levels. The rounding off operation is known as "quantization". Then each sample is coded into a binary number which is equivalent to the index of the quantization level that is closest to the sample value.

1.6.1 Quantization

When an information signal is pulse amplitude modulated, it becomes discrete in time only. It remains analogue in amplitudes since all the values within the specified range are allowed. PAM signal is said to be quantized when each pulse of the PAM signal is adjusted in amplitude to coincide with the nearest level within a finite set.

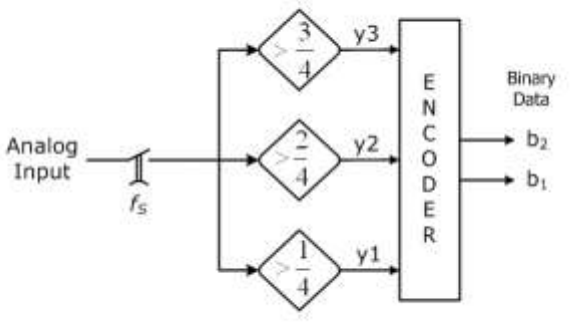
It is clear from the figure below, quantization error (noise) can be reduced by increasing the number of quantization levels (L), i.e. decreasing the intervals (q) between the levels.





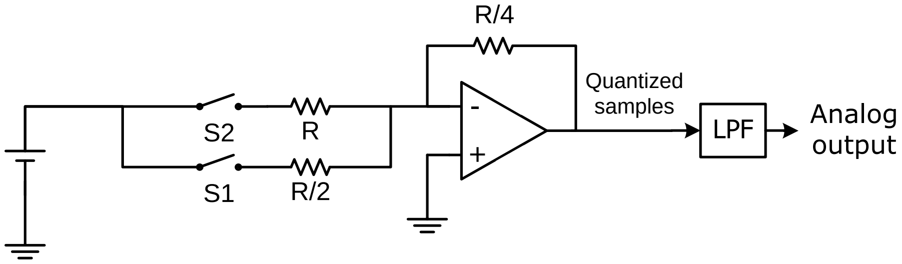
1.6.2 Encoding

The quantized samples are now to be coded with l bits per sample ($l = \log_2 L$). One of the most popular quantizer/encoder circuits is the parallel quantizer, which requires $L - 1$ comparators. The following is a 2-bit PCM coder.

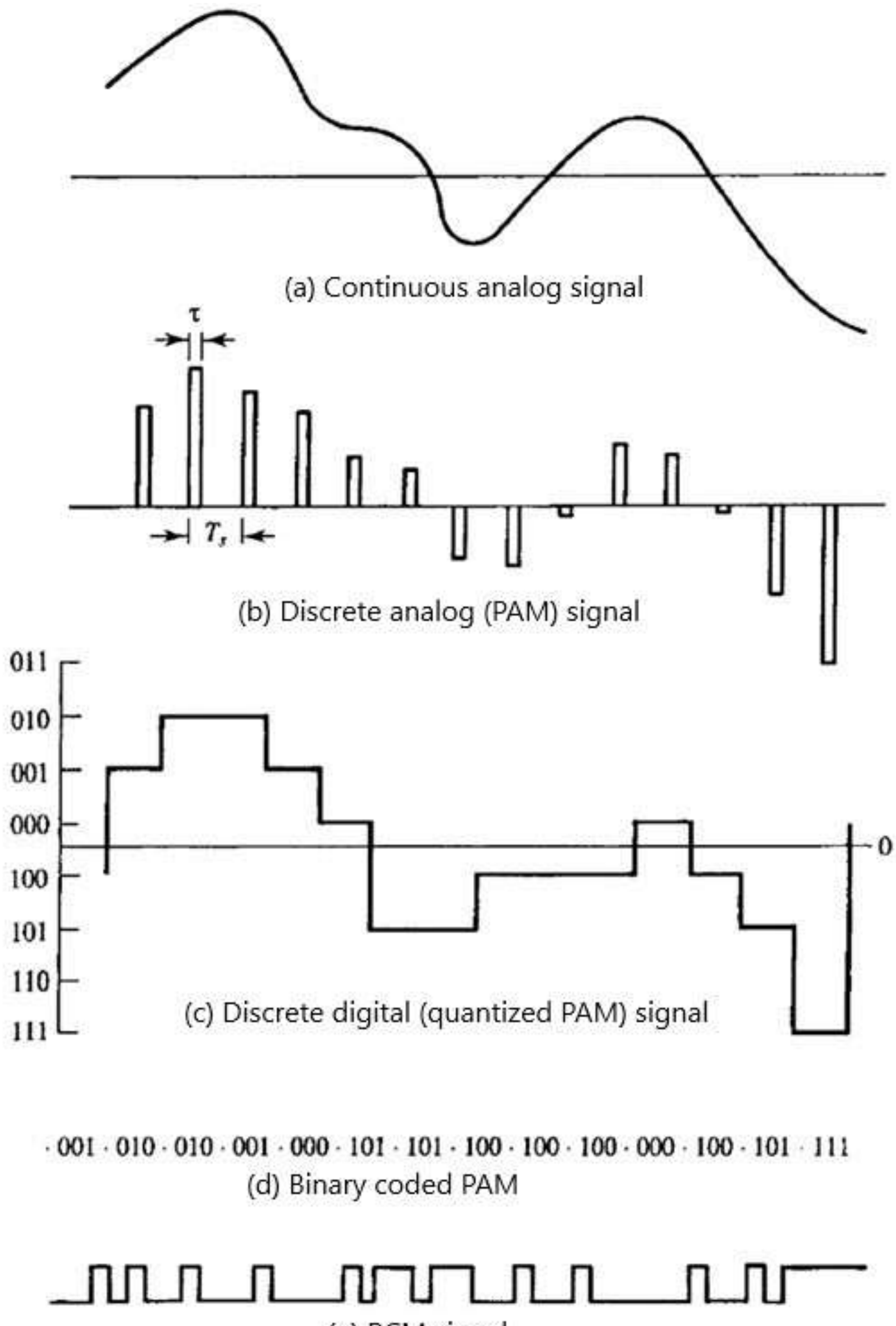


1.6.3 Decoding

The transmitter sends binary codes to the receiver via a channel. The receiver must decode the bit sequence back to a time function. This is done by associating each group of bits with the corresponding quantization level; thereby reconstructing the quantized waveform by LPF. In this simple 2-bit PCM decoder, S1 & S2 are received binary, S1=MSB and S2=LSB.



The PCM technique is considered as an Analog to Digital Converter (ADC) at the transmitter and Digital to Analog Converter (DAC) at the receiver. The following figure illustrates the ADC



(a) Continuous analog signal

(b) Discrete analog (PAM) signal

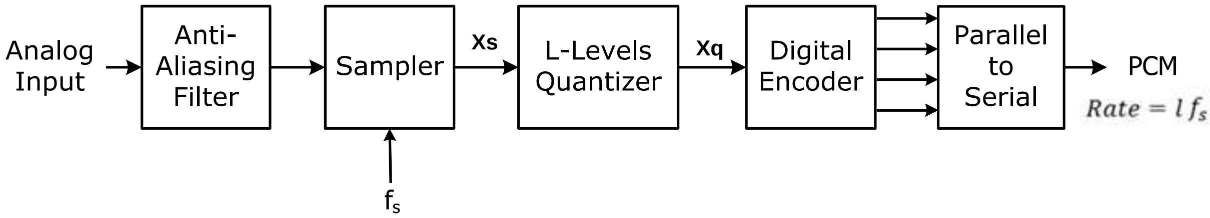
(c) Discrete digital (quantized PAM) signal

(d) Binary coded PAM

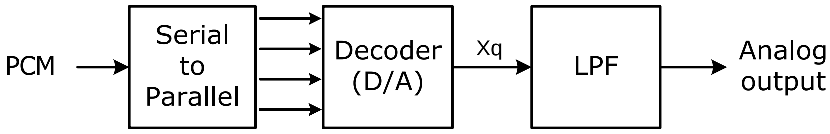
(e) PCM signal

The complete block diagram of PCM system is:

Modulator:



Demodulator:



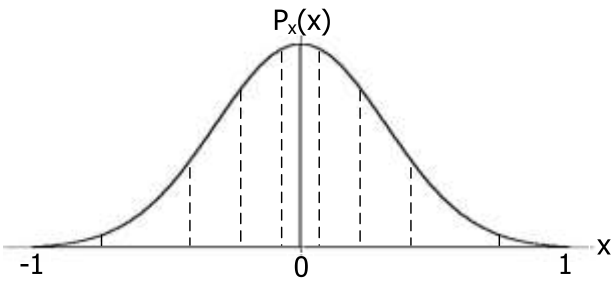
1.6.4 Non-Uniform Quantization

<u>UNEQUAL STEP-SIZE</u>

Uniform quantization assumes that the information signal has uniform PDF, i.e. all quantization levels are used equally. For most signals, it is not the case.

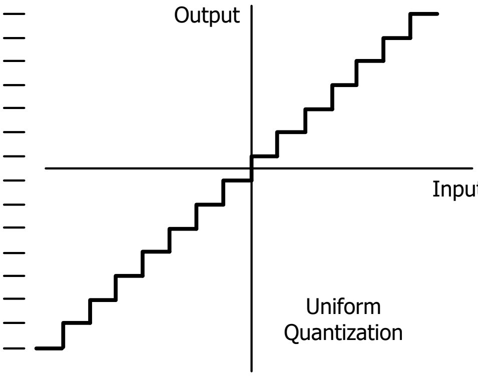
If the PDF of the information signal is not uniform (nevertheless known and constant with time), then we can optimize the locations of the quantization levels to obtain minimum quantization noise introduced.

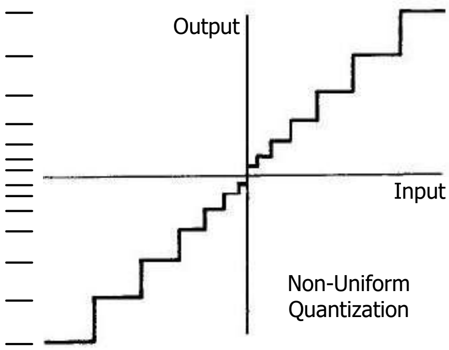
As an illustration, let the normalized signal $x(t)$ has the typical probability function $P_x(x)$:



The shape of $P_x(x)$ means $|x(t)| \ll 1$ most of the time. Therefore, we can use non-uniform quantization as indicated by the dashed lines. The quantization lines are located here close to each other near $x = 0$. They are sparse for large values of $|x(t)|$, as large $|x(t)|$ occurs infrequently.

The depictions below show the uniform and the non-uniform distribution of the quantization levels.

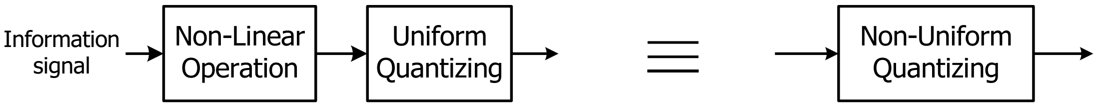




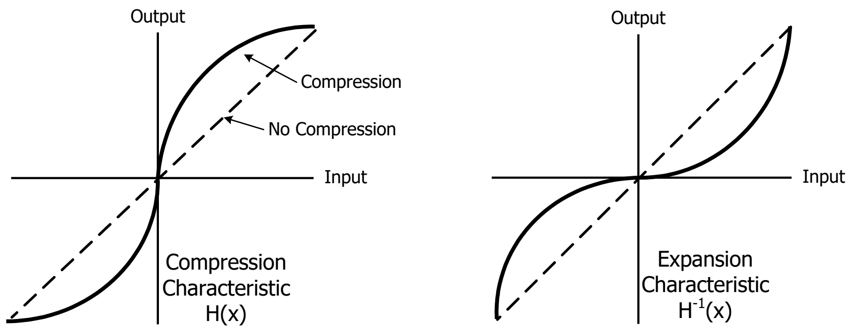
Practically, such optimization is a difficult procedure because it requires prior knowledge of the signal PDF. Problems arise if the information signal has an unknown PDF or if its PDF changes with time. However, they are similar in some signals. For example, in the case of voice signals, the PDF shape of different speakers is usually similar, but the gross level can vary widely between speakers, e.g. man is shouting and woman softly spoken. Therefore, the approach taken in practice is to use uniform quantizing after non-linear compression (the *companding*).

COMPANDING (*COMP*RESSING-EXP*ANDING*)

This is the process of *compressing* the information signal prior to linear quantization at transmission. The compression is achieved via a non-linear amplitude characteristic circuit.



The receiver *expands* the reconstructed signal with the inverse characteristic to restore the original waveform.



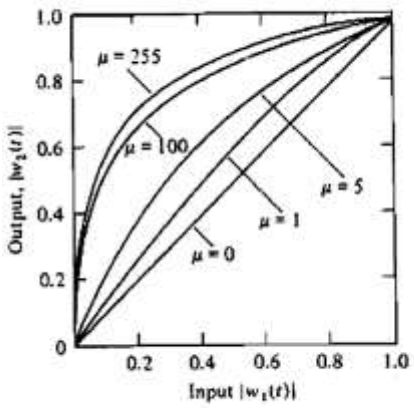
So, PCM with companding system will be:

Companding Methods

There are two types of companders that are practically and widely used for speech coding:

(1) The μ-law compander (used in the US, Canada and Japan). The parameter μ controls the amount of compression and expansion. The standard compressor uses $\mu = 255$ followed by a uniform quantizer with 128 levels (7 bits per sample).

$$H(x) = \frac{\log(1+\mu|x|)}{\log(1+\mu)}$$



(2) The A-law compander (used in most countries). A is chosen to be 87.56.

$$H(x) = \begin{cases} \dfrac{A|x|}{1+\ln A} & \text{for } 0 \le |x| \le \dfrac{1}{A} \\ \dfrac{1+\ln(A|x|)}{1+\ln A} & \text{for } \dfrac{1}{A} \le |x| \le 1 \end{cases}$$