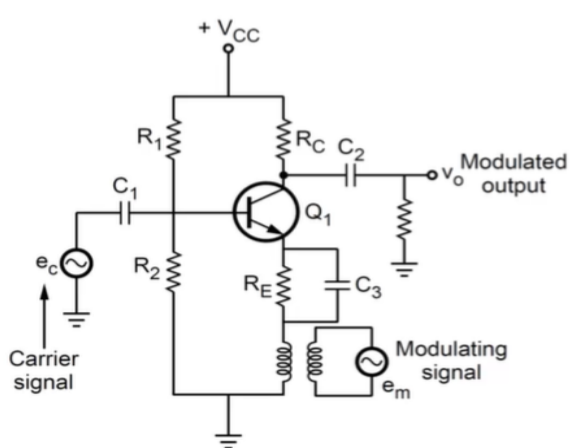
Amplitude Modulator

Amplitude modulation may be achieved in a number of ways, the most common method being to use the modulating signal to vary the otherwise steady voltage on the output electrode of an amplifier.

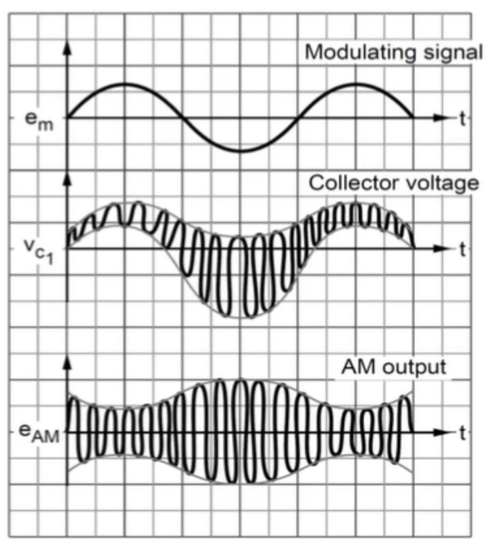
Vacuum tubes are used for very high power outputs (in the kilowatt and higher ranges), and transistors are used for lower powers. (Pentode vacuum tubes and transistors have similarly shaped output characteristics, although they operate at greatly different voltage and current levels.) The basic circuit for a BJT modulator is shown in Figure.

The transistor is normally operated in the class C mode in which it is biased well beyond cutoff. The carrier input to the base must be sufficient to drive the transistor into conduction over part of the RF cycle, during which the collector current flows in the form of pulses. These pulses are periodic at the carrier frequency and can therefore be analyzed into a trigonometric Fourier series. The tuned circuit in the collector is tuned to resonate at the fundamental component, and thus, to a close approximation, the RF voltage at the collector is sinusoidal.

Circuit diagram and waveforms



(a) Circuit diagram of low level AM modulator



(b) Waveforms

Operation

- Carrier signal is given to the base of the amplifier and modulating signal is given to emitter. In absence of modulating signal, the circuit simply operates as linear class A amplifier.
- When modulating signal is applied to an emitter, the gain of the amplifier varies according to voltage of modulating signal.
- Depending upon the gain variations, carrier signal is amplified. Thus amplitude of carrier signal is modulated by modulating signal.
- Voltage gain of emitter modulator is given as,

$$A_v = A_q[1 + m\sin(2\pi f_m t)]$$

where A_v is voltage gain with modulation.

A_q is quiescent voltage gain.

Advantages of low level modulation

i) Less modulating signal power is required to obtain high percentage modulation.

ii) Modulator circuit is to be designed at low power.

Disadvantages of low level modulation

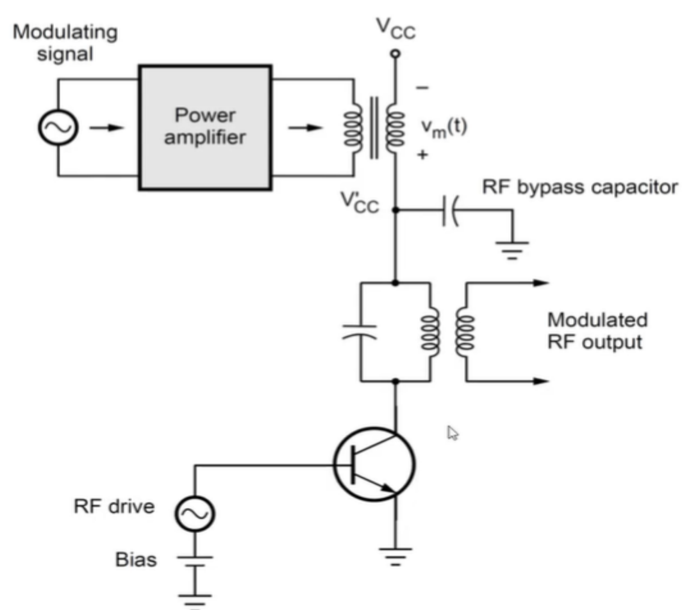
Amplifiers following modulator stage must be linear. At high operating powers linear amplifiers are very inefficient.

High Level AM Modulator

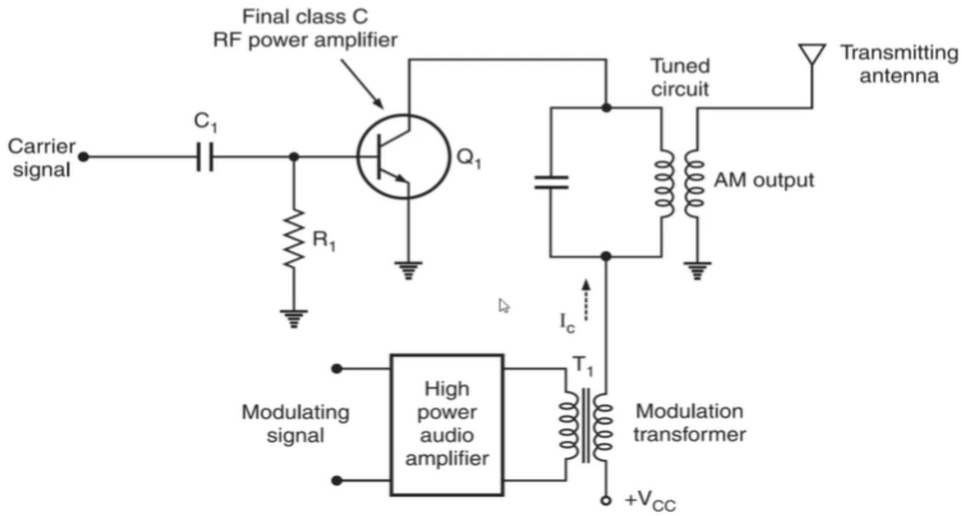
In such modulator, the modulation takes place in the final state of the transmitter. The output of the modulating stage is directly connected to an antenna.

Collector modulator

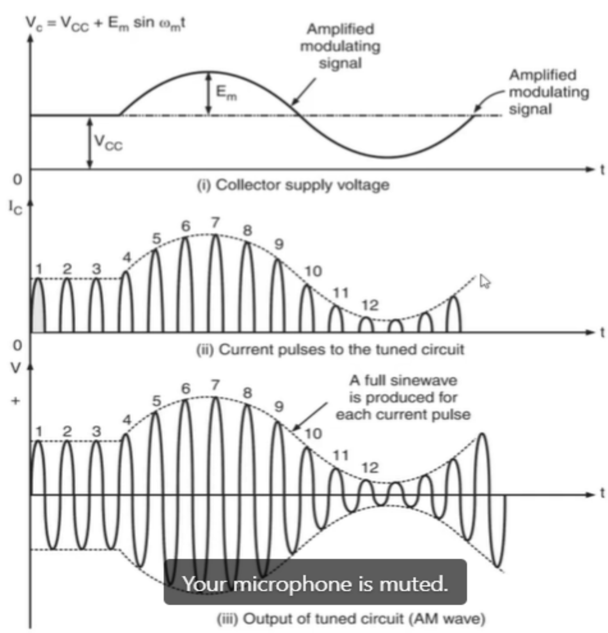
The modulated output can be obtained by making the voltage on output electrode to vary according to input modulating signal. Fig. shows the collector modulator. The transistor is biased well beyond cut-off so that it operates in class C mode. The class C mode is used because of its high efficiency. The RF drive is a carrier signal used for AM. This carrier amplitude is such that it drives transistor in conduction over part of its cycle. It is applied to the base of transistor. The modulating signal is passed through the power amplifier and applied to the collector through a low frequency transformer. This voltage is shown as $v_m(t)$ in figure. This modulating voltage is in series with the supply voltage V_{CC}. Hence the collector voltage becomes $V'_{CC} = V_{CC} + v_m(t)$. The tuned LC circuit associated tuned transformer on the collector receives the AM signal. Because of modulating voltage, the net supply voltage of transistor changes according to slow variations in $v_m(t)$. Hence the RF carrier signal amplitude is also varied according to variations in $v_m(t)$. Thus AM signal is produced across the LC circuit at the collector.



BJT collector modulator (class C amplifier)



High level collector modulator



Waveforms of collector modulator circuit

Advantages of high level modulators

i) There is no constraint of linear operation on amplifiers preceding modulator stage.

ii) Power efficiency is good.

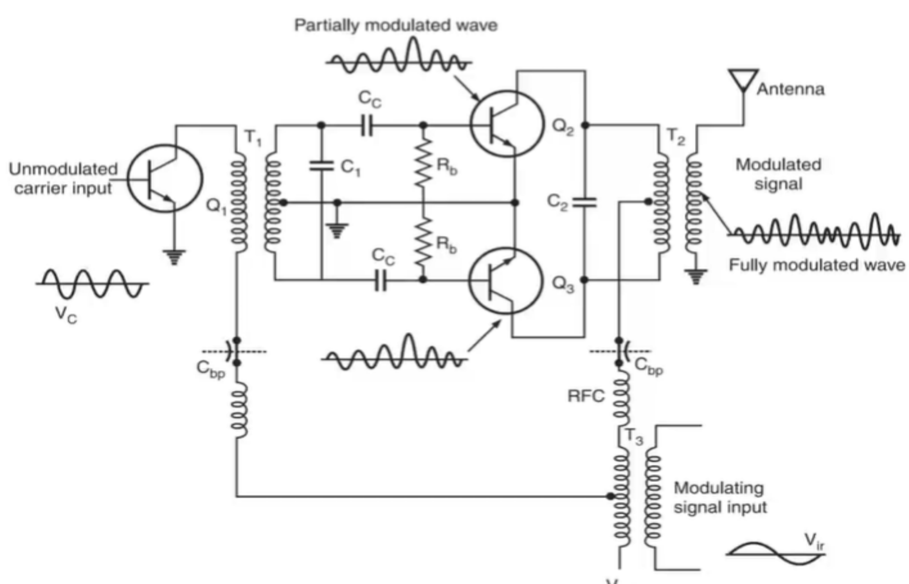
Disadvantages of high level modulators

i) High modulating power is required to achieve higher percentage modulation.

ii) Final modulating signal amplifier has to supply all the sideband power, which is around 33 % of total transmitted power.

Your microphone is muted.

Simultaneous Base and Collector Modulation



High power AM DSBFC transistor modulator

Amplitude Demodulator Circuit

Amplitude Demodulator Circuits

At the receiver, a circuit must be provided that recovers the information signal from the modulated carrier.

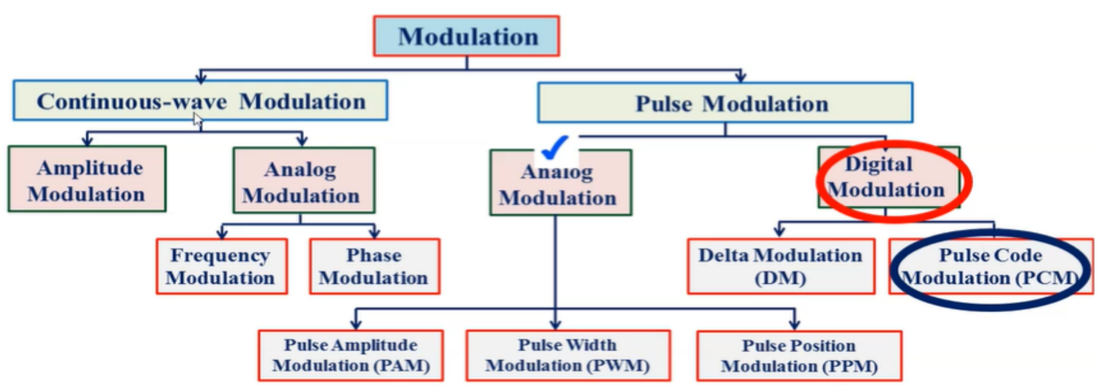
The most common circuit in use is the ***Diode Envelope Detector***, which produces an output voltage proportional to the envelope, which is the modulating or information signal. The basic circuit is shown in Fig. (a).

The diode acts as a rectifier and can be considered an ON switch when the input voltage is positive, allowing the capacitor *C* to charge up to the peak of the RF input. During the negative half of the RF cycle, the diode is off, but the capacitor holds the positive charge previously received, so the output voltage remains at the peak positive value of RF. There will, in fact, be some discharge of *C*, producing an RF ripple on the output waveform, which must be filtered out.

As the input voltage rises with the modulating cycle, the capacitor voltage has no difficulty in following this, but during the downward swing in modulation the capacitor may not discharge fast enough unless an additional discharge path is provided by the resistor *R*. The time constant of the *CR* load has to be short enough to allow the output voltage to follow the modulating cycle and yet long enough to maintain a relatively high output voltage.

Pulse Code Modulation

Types of Modulation



PCM

Pulse-code modulation (**PCM**) is a method used to digitally represent sampled analog signals.

It is the standard form of digital audio in computers, compact discs, digital telephony and other digital audio applications.

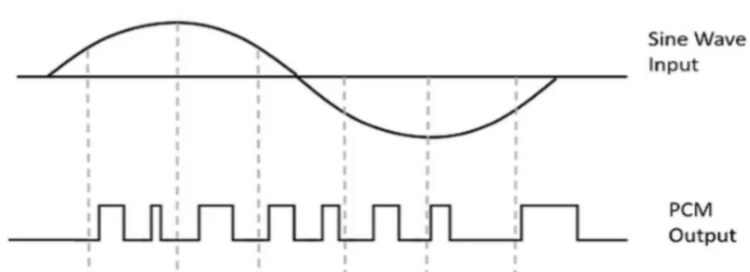
In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps.

Your microphone is muted.

PCM Signal

Pulse Code Modulation

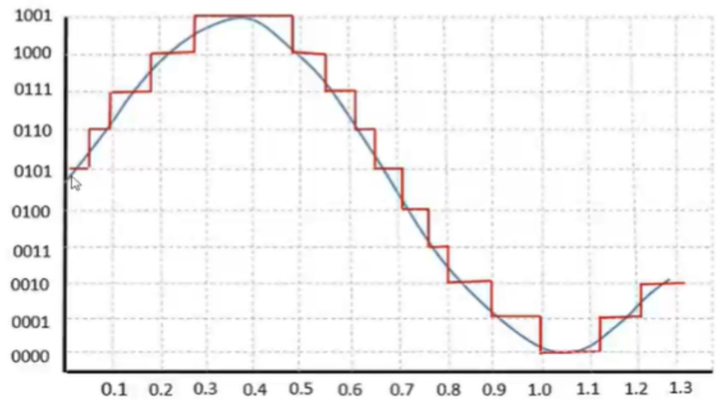
A signal is Pulse Code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a **Pulse Code Modulation (PCM)** will resemble a binary sequence. The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.



Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.

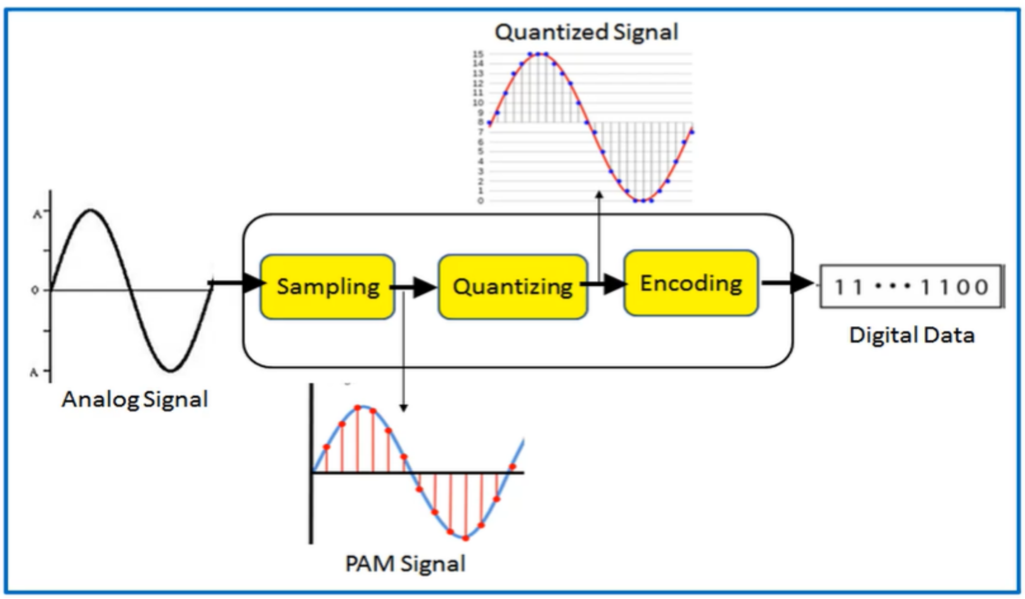
In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the red one represents the quantized signal.



Both sampling and quantization results in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as **representation levels** or **reconstruction levels**. The spacing between two adjacent representation levels is called a **quantum** or **step-size**.

Pulse Code Modulation (PCM)

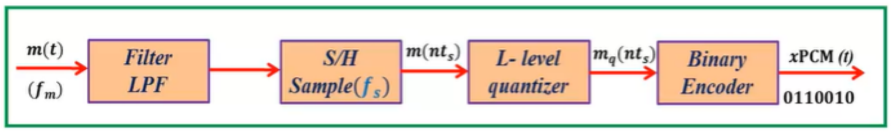


Pulse Code Modulation (PCM)

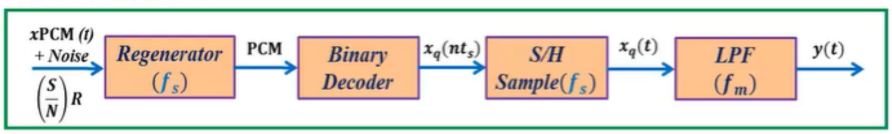
Most current digital audio systems (computers, compact discs, digital telephony etc.) use multi-bit Pulse Code Modulation (PCM) to represent the sound signal. Analog to PCM conversion consists of three steps: Sampling, Quantizing and Encoding.

Block Diagram of PCM system

PCM Transmitter



PCM Receiver



PCM Advantages and Disadvantages

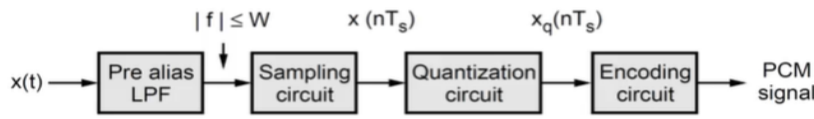
- **Advantages of PCM:**

1- Low channel noise and interference.
2- Easy multiplexing of various PCM signals.

- **Disadvantages of PCM:**

1- Complex systems.
2- Wide bandwidth is required.

- $x(t)$ is first passed through pre-alias low pass filter of cut off frequency $'W'$ Hz. Hence all the frequency components higher than 'W' Hz are blocked.



Block diagram of PCM generator

- The signal is then sampled by the sampling circuit. The sampling frequency $f_s \geq 2W$. And $T_s = \frac{1}{f_s}$.
- The quantization circuit quantizes the sampled signal to finite quantization levels. The quantized signal takes any one of the 'q' quantization levels.
- The quantized signal is then represented by finite number of digits. For example if there are 3-bits, then it will represent $2^3 = 8$ quantization levels. For example quantization level of '7' will be encoded as 111 by 3-bits. This encoded signal is called PCM signal.

Transmission Bandwidth in PCM

Let the quantizer use $'v'$ number of binary digits to represent each level. Then the number of levels that can be represented by $'v'$ digits will be,

$$q = 2^v$$

Here $'q'$ represents total number of digital levels of q-level quantizer. Each sample is converted to 'v' binary bits. i.e. Number of bits per sample = v

We know that, Number of samples per second = f_s

$\therefore$ Number of bits per second is given by,

$$\begin{aligned}(\text{Number of bits per second}) &= (\text{Number of bits per samples}) \\ &\quad \times (\text{Number of samples per second}) \\ &= v \text{ bits per sample} \times f_s \text{ samples per second}\end{aligned}$$

The number of bits per second is also called signaling rate of PCM and is denoted by 'r' i.e.,

Signaling rate in PCM : $r = v\, f_s$

Here $f_s \geq 2W$.

Bandwidth needed for PCM transmission will be given by half of the signaling rate i.e.,

Transmission bandwidth of PCM
$$\begin{cases} B_T \geq \frac{1}{2} r \\ B_T \geq \frac{1}{2} v\, f_s \quad \text{Since } f_s \geq 2\,W \\ B_T \geq v\,W \end{cases}$$

PCM Equations

1 $R_b = b f_s \quad (b/s)$

R_b : Bit rate (b/s)

b : Number of bits.

f_s: sampling frequency.

2 **Transmission bandwidth B_{PCM} of PCM**

$$B_{PCM} \geq \frac{1}{2} R_b \geq \frac{1}{2} b f_s \geq \frac{1}{2} b f_m$$

3 **Information Capacity of PCM**

$$C = B * Log_2(1 + \frac{S}{N})$$

$$C = B * Log_2(1 + \frac{12P_x}{\Delta^2})$$

PCM Numericals

Q (1):- A compact disc (CD) records audio signals digitally by using **PCM**. Assume the audio signal **15KHz**.

1- What is the ***Nyquist rate***.

2- If the samples are quantized into ***L = 65536*** levels and then binary coded, determine the number of binary digits required to encoded a sample.

3- Determine the number of binary digits per second (***bit rate***).

Solution:-

1- $f_N = f_s = 2f_m$

$$f_N = 2 * 15\ K = 30\ KHz$$

2- $L = 2^b$

$$b = \frac{Log\ L}{Log\ 2} = \frac{Log\ (65536)}{Log\ 2} = 16\ bit$$

3- $R_b = bf_s$

$$R_b = 16 * 30\ K = 480\ K\ (b/s)$$

Q (2):- A television has band limited **4.5 MHz**. This signal is sampled, quantized and binary coded to obtain a **PCM** signal.

1- Determine the sample rate if the signal is to be sampled at a rate ***20%*** above the Nyquist rate .

2- If the samples are quantized into ***L = 1024*** levels and then binary coded, determine the number of binary digits required to encoded a sample.

3- Determine the number of binary digits per second (***bit rate***) and the minimum bandwidth required to transmit this signal.

Solution:-

1- $f_N = 2f_m = 2 * 4.5\ M = 9\ MHz$

$$\therefore f_s > 20\%\ f_N$$

$$f_s > 20\% * 9\ MHz = 1.8\ MHz$$

$$f_s = 9\ M + 1.8\ M = 10.8\ MHz$$

2- $L = 2^b$

$$b = \frac{Log\ L}{Log\ 2} = \frac{Log\ (1024)}{Log\ 2} = 10\ bit$$

3- $R_b = bf_s$

$$R_b = 10 * 10.8\ M = 108\ M\ (b/s)$$

$$B_{PCM} \geq \frac{1}{2} R_b$$

$$B_{PCM} \geq \frac{1}{2} 108\ M$$

$$B_{PCM} \geq 54\ MHz$$

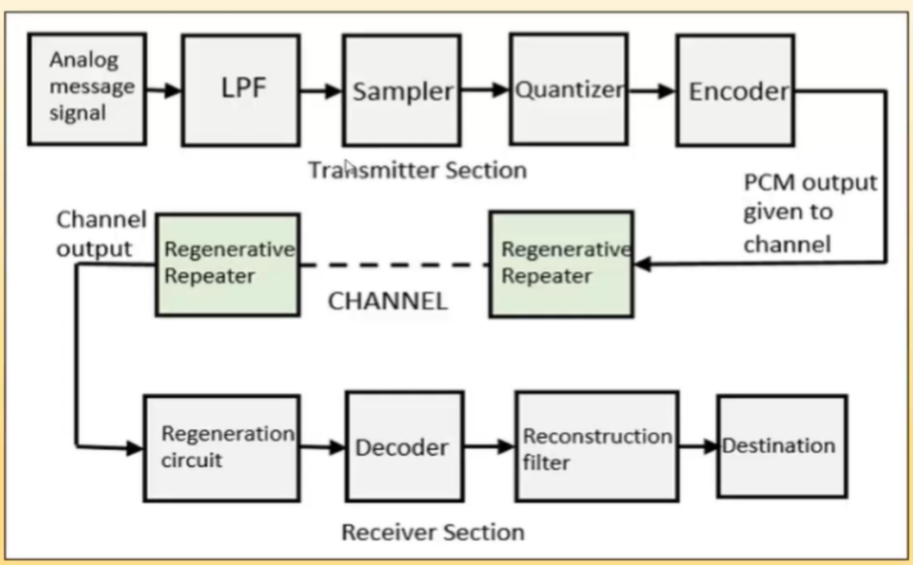
Basic Elements of PCM

The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the **analog-to-digital converter** section. The low pass filter prior to sampling prevents aliasing of the message signal.

The basic operations in the receiver section are **regeneration of impaired signals, decoding,** and **reconstruction** of the quantized pulse train. The following figure is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.

Your microphone is muted.

PCM Transmitter & Receiver



Low Pass Filter (LPF)

This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

Sampler

This is the circuit which uses the technique that helps to collect the sample data at instantaneous values of the message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component **W** of the message signal, in accordance with the sampling theorem.

Quantizer

Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

Encoder

The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections will act as an analog to the digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal.

Decoder

The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the **demodulator**.

Reconstruction Filter

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low pass filter is employed, called as the reconstruction filter to get back the original signal.

Hence, the Pulse Code Modulator circuit digitizes the analog signal given, codes it, and samples it. It then transmits in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

Quantization

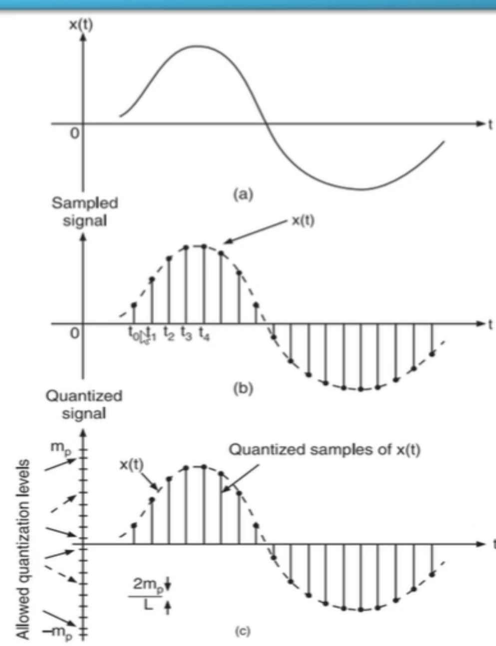
There are few modulation techniques which are followed to construct a PCM signal. These techniques like **sampling, quantization,** and **companding** help to create an effective PCM signal, which can exactly reproduce the original signal.

Quantization

The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as **Quantization**.

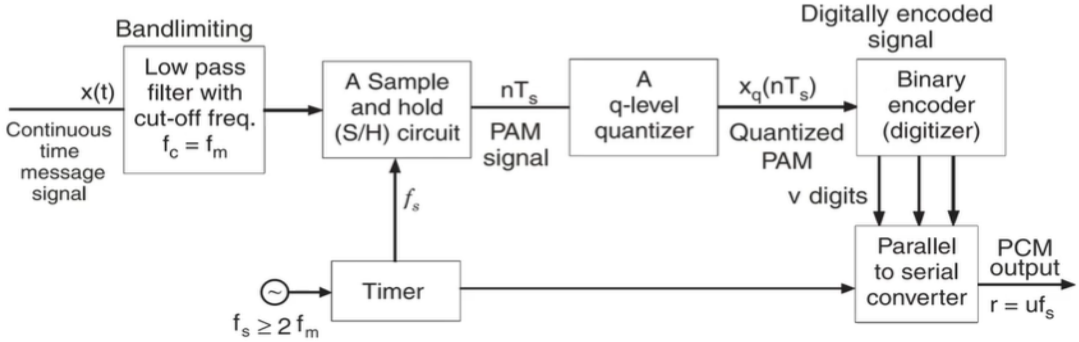
The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a **continuous-amplitude sample** into a **discrete-time signal**.

Sampling & Quantization



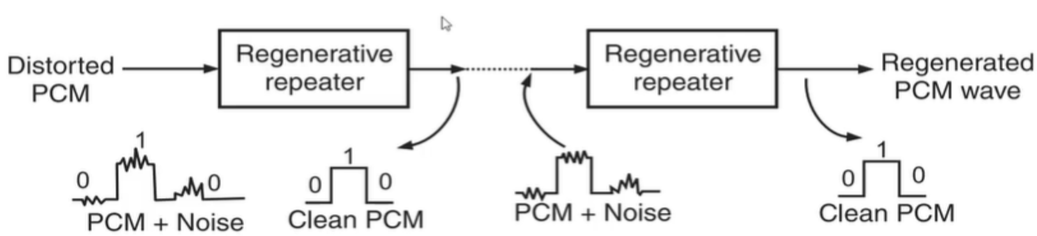
(a) An analog signal, (b) Samples of analog signal, (c) Quantization

PCM Generator



A practical PCM generator

PCM Transmission Path

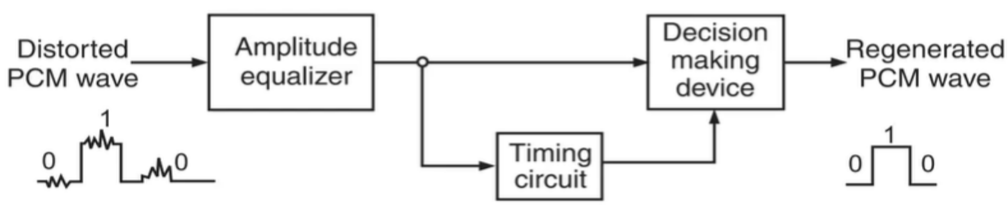


PCM transmission path

So far we have gone through different modulation techniques. The one remaining is **digital modulation**, which falls under the classification of pulse modulation. Digital modulation has Pulse Code Modulation (PCM) as the main classification. It further gets processed to delta modulation and ADM

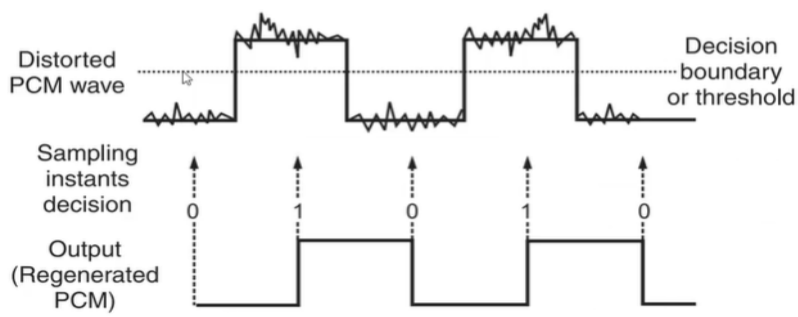
PCM (Pulse Code Modulation)

PCM – Regenerative Repeater



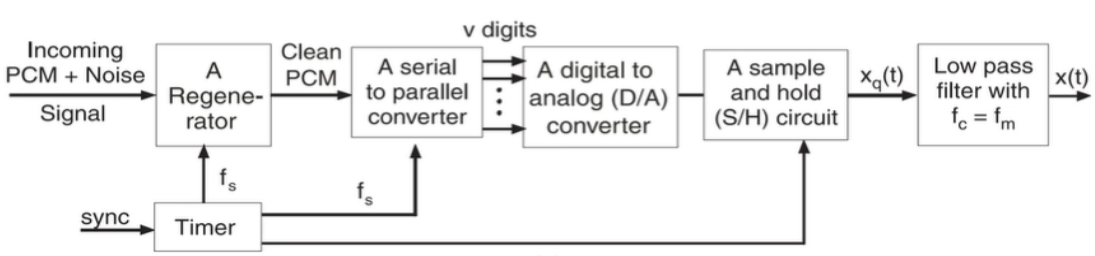
Block diagram of a regenerative repeater

PCM Regenerative Repeater Waveform



Waveforms of a regenerative repeater

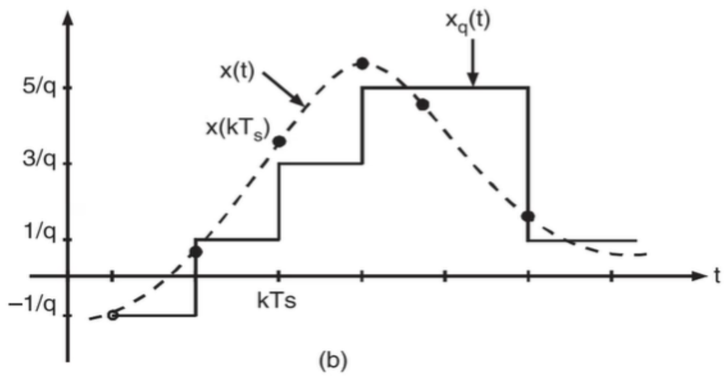
PCM Receiver



(a)

Your microphone is muted.

PCM Reconstructed Waveform

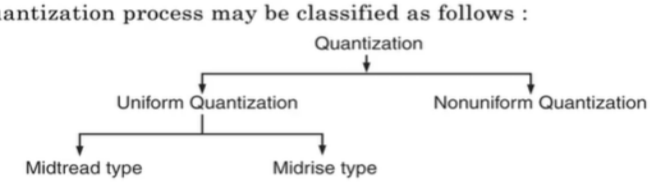


(a) PCM receiver, (b) Reconstructed waveform

Classification of Quantization Process

Classification of Quantization Process

Basically, quantization process may be classified as follows :



The quantization process can be classified into two types as under:

(i) Uniform quantization

(ii) Non-uniform quantization.

This classification is based on the step size as defined earlier.

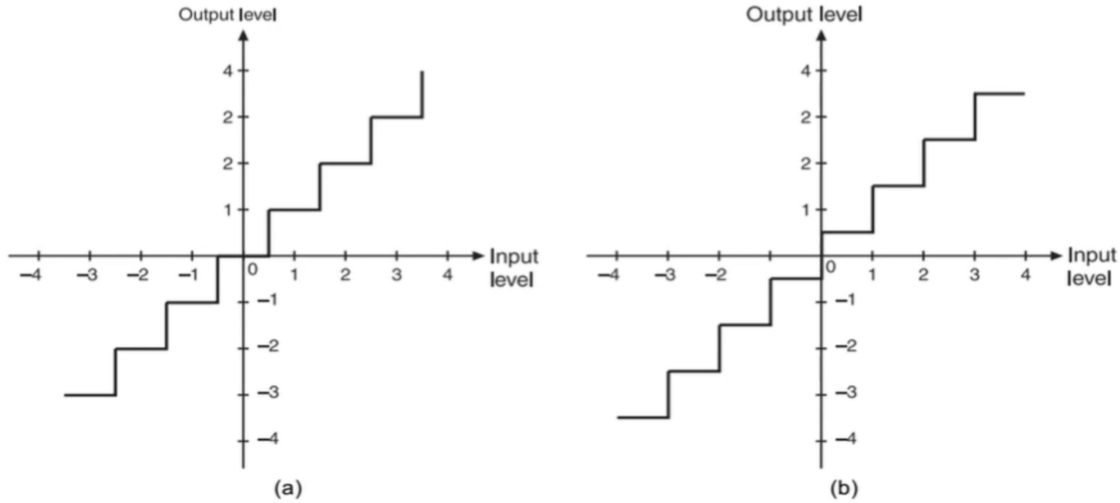
(i) Uniform Quantizer

A uniform quantizer is that type of quantizer in which the 'step size' remains same throughout the input range.

(ii) Non-uniform Quantizer

A non-uniform quantizer is that type of quantizer in which the 'step-size' varies according to the input signal values.

Uniform Quantization



Two types of Uniform quantization: (a) Midtread, and (b) Midrise

Transfer characteristic of Quantizer