PULSE AMPLITUDE MODULATION

In the PAM systems the width and the timing of the pulse is maintained constant, but the Amplitude of each pulse is varied in accordance with instantaneous value of the analog signal (massage Signal).

Sampling theorem provides us the basis to convert analog signal to discrete time.

The signal is sampled at regular intervals such that each sample is proportional to the amplitude of the signal at that sampling instant.

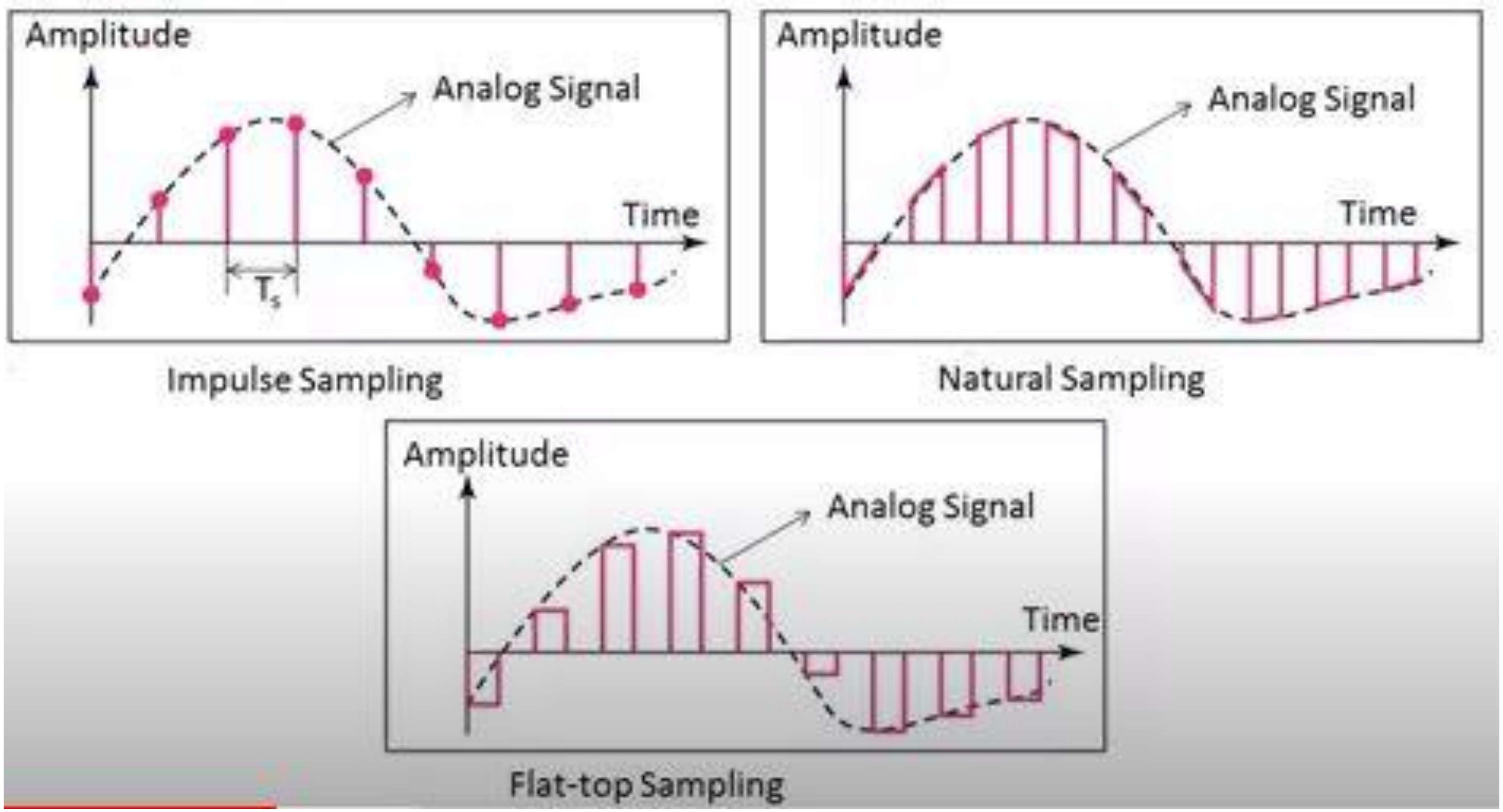
Sampling can be performed with the help of sampler.

Depending upon different types of sampling techniques we have different types of samplers:

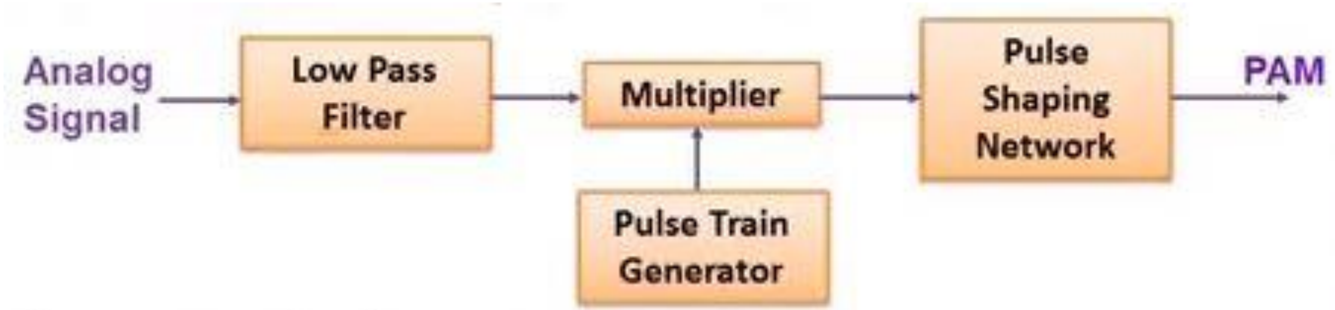
- **Impulse sampler**
- **Natural sampler**
- **Flat Top sampler**

Impulse sampling is not used for information exchange as it has very law power.

Flat Top sampling is most widely used due to its noise mitigation capability.



Pulse Amplitude Modulator – Flat Top Sampled



The message signal is passed to the LPF in order to make it band limited whose cutoff frequency is fm. The LPF is used to avoid aliasing effect.

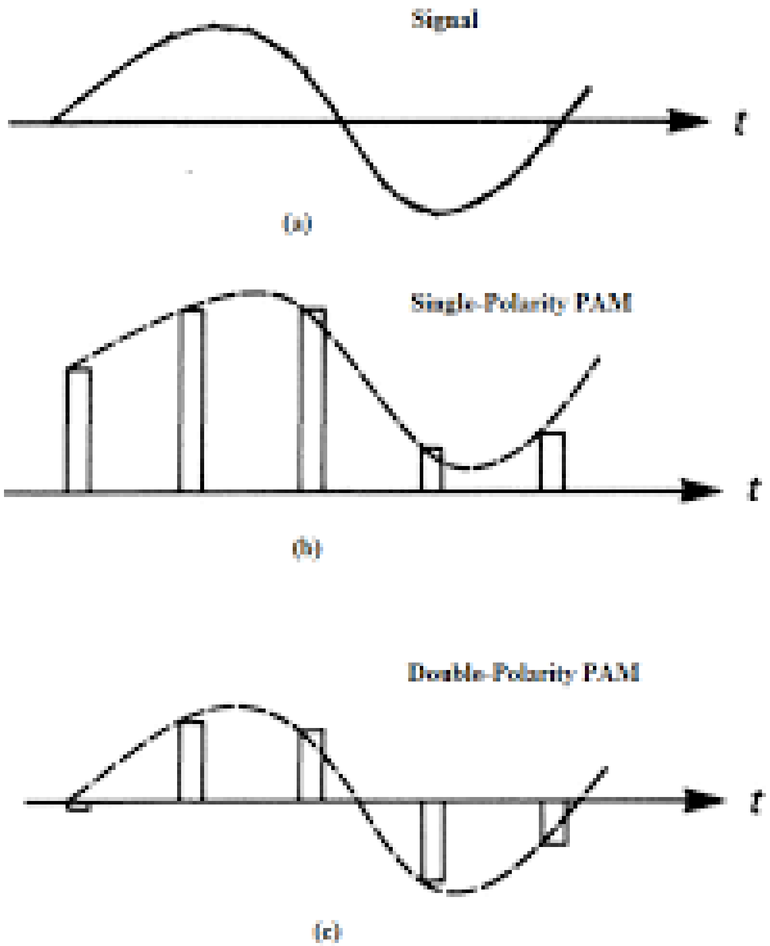
This band limited signal is then sampled at the multiplier with the help of pulse train generator sampled at Ts sec.

It is then passed through a pulse shaping network which produces flat top sampled pulses.

PAM could be:

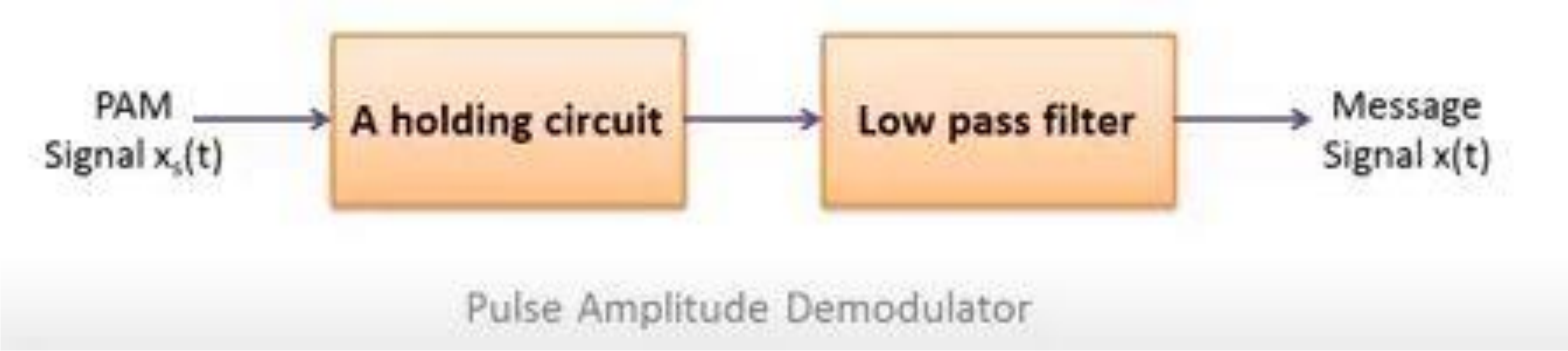
(i) Single polarity PAM: A suitable fixed DC bias is added to the signal to ensure that all the pulse are positive.

(ii) Double polarity PAM: In this the pulses are both positive and negative.

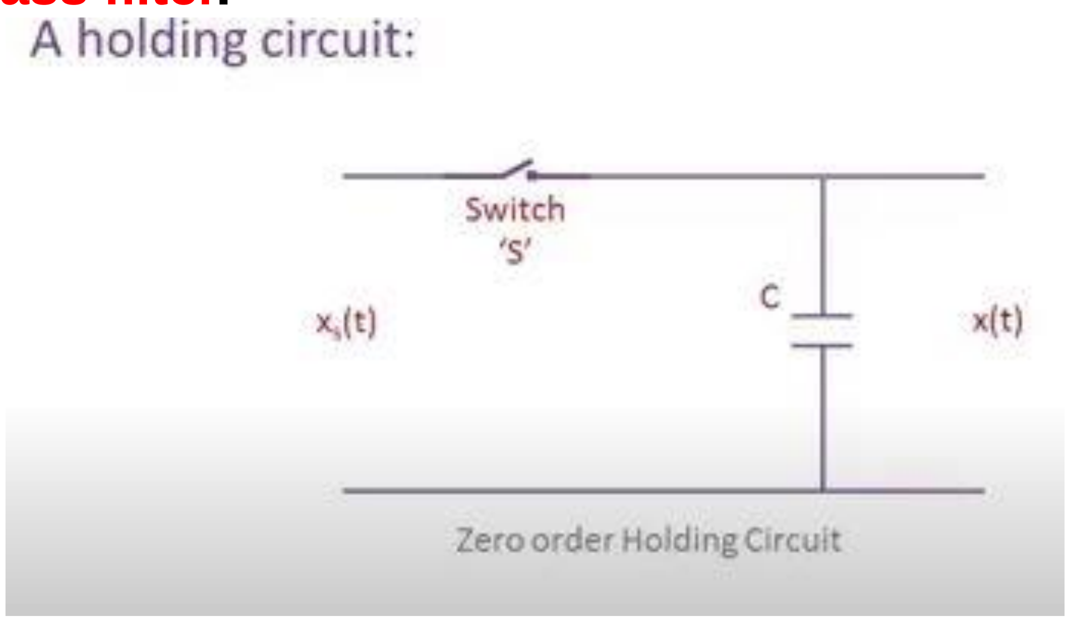


RECEPTION OF PAM SIGNAL

Figure shows the system for recovering message signal x(t) from PAM signal Xs(t).



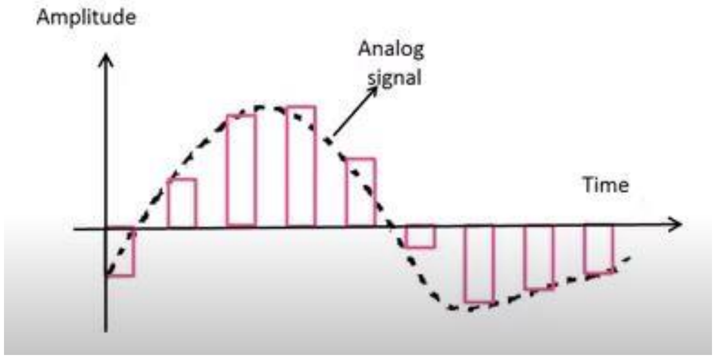
The PAM Signal is passed through the holding circuit and then with a law pass filter.

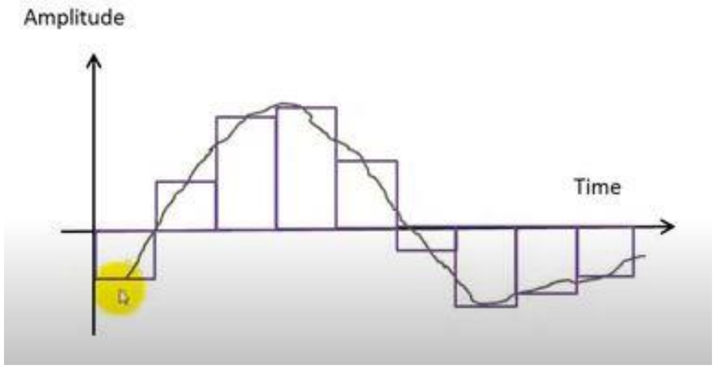


The switch ‘S’ is closed after the arrival of the pulse and it is opened at the end of pulse.

Hence the capacitor ‘C’ is charged to the pulse amplitude value and hold this value till the next pulse arrives.

After that the signal is smoothened by a LPF in order to get the message signal back.





PAM Signal Reception: The PAM signal, denoted as Xs(t), is received at the input of the reception system. This PAM signal carries the modulated information and needs to be demodulated to recover the original message signal. After that the signal is smoothened by a LPF in order to get the message signal back.

Holding Circuit: The PAM signal is first passed through a holding circuit. The purpose of this circuit is to capture and store the pulse amplitude value. When a pulse arrives, the switch 'S' is closed, allowing the capacitor to charge to the amplitude value of the incoming pulse. This value is held by the capacitor until the next pulse arrives.

Low-Pass Filter (LPF): After the holding circuit, the signal is then passed through a low-pass filter (LPF). The LPF is used to smoothen the signal. In the context of PAM demodulation, the LPF serves to remove any high-frequency components or noise that may have been introduced during transmission or reception.

Message Signal Recovery: The output of the LPF should be a reconstructed version of the original message signal x(t). The LPF effectively removes the discrete nature of the PAM signal and smooths it into a continuous signal. This reconstructed signal can then be considered an estimate or approximation of the original message signa

Transmission of PAM signals

For PAM signals to be transmitted through space using antennas, they must be amplitude/frequency/ phase modulated by a high frequency carrier and only then they can be transmitted.

Thus the overall system is PAM-AM. PAM-FM or PAM-PM and at receiving end, AM/ FM/PM detection is first employed to get the PAM signal and then message signal is recovered.

Transmission BW of PAM Signal

In Pulse amplitude modulated signal the pulse duration Γ is considered to be very small in comparison to time period(Sampling period) Ts between any two samples.

Γ << Ts

Now, if the maximum frequency of the modulating signal x(t) is fm, then according to the sampling theorem the sampling frequency fs must be greater than or equal to the Nyquist rate that is :

$$f_s \geq 2 f_m$$

We can get as; $\frac{1}{T_s} \geq 2f_m \qquad \because f_s = \frac{1}{T_s}$

$$T_s \leq \frac{1}{2f_m}$$

But as per the assumption:

$$\tau << T_s$$

$$\because \tau << T_s \leq \frac{1}{2f_m}$$

If we consider the 'ON' and 'OFF' time of PAM signal is same, then the maximum frequency of PAM will be:

$$f_{max} = \frac{1}{\tau + \tau} \Rightarrow \frac{1}{2\tau}$$

The Bandwidth required for the transmission of a PAM signal would be greater than or equal to the maximum frequency fmax is given by:

$$BW \geq f_{max} \qquad \text{but } f_{max} = \frac{1}{2\tau}$$

$$BW \geq \frac{1}{2\tau}$$

But since $\tau << \frac{1}{2f_m}$

$$\because BW \geq \frac{1}{2\tau} >> f_m$$

$$BW >> f_m$$

Drawbacks of PAM

Large Bandwidth Requirement: One of the significant drawbacks of PAM is its requirement for a large bandwidth for transmission. The bandwidth of a PAM signal is directly related to the maximum frequency present in the modulating signal. This means that in order to transmit signals with high-frequency components accurately, a wide bandwidth is needed. This can be inefficient in terms of spectrum utilization.

Noise Interference: PAM signals are susceptible to noise interference, especially when the amplitude of the PAM pulses varies in accordance with the modulating signal. Variations in amplitude make PAM signals more vulnerable to noise, and this can result in signal degradation and reduced signal-to-noise ratio (SNR) at the receiver.

Peak Power Variations: In PAM, the peak power required by the transmitter varies with the modulating signal. This can pose challenges for the transmitter's power amplifier, which must be capable of handling the highest peak power levels. Managing peak power variations can be complex and may require additional components and design considerations.

Inefficient Power Usage: PAM is not very power-efficient, particularly when compared to other modulation schemes like Pulse Code Modulation (PCM) or Quadrature Amplitude Modulation (QAM). Since PAM represents information solely through amplitude variations, it does not make efficient use of both amplitude and phase as QAM does.

Limited Data Rate: The maximum data rate achievable with PAM is limited by the bandwidth available and the signal-to-noise ratio. To transmit higher data rates, one may need to employ higher-order modulation schemes, which can be more complex but offer better spectral efficiency.

Lack of Robustness: PAM signals may not be as robust in noisy environments or over long-distance transmission as some other modulation techniques. The reliance on amplitude variations makes them susceptible to amplitude distortions, which can affect signal quality.

Demodulation of PAM

PAM signal sampled at Nyquist rate can be reconstructed at the receiver end , by passing it through an efficient Low Pass Filter (LPF) with exact cut off frequency of fs/2. This is known as Reconstruction or Interpolation Filter.

The low pass filter eliminates the high-frequency ripples and generates the demodulated signal. This signal is then applied to the inverting amplifier to amplify its signal level to have the demodulated output with almost equal amplitude with the modulating signal

PULSE WIDTH MODULATION: PWM

Pulse Width Modulation (PWM) / Pulse Duration Modulation (PDM) / Pulse Time Modulation (PTM) is an analog modulating scheme in which the width or duration or time of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.

Modulation Principle: PWM is a method of encoding analog information (the message signal) into a digital signal (the pulse train) by varying the width or duration of pulses. The amplitude of the pulse train remains constant.

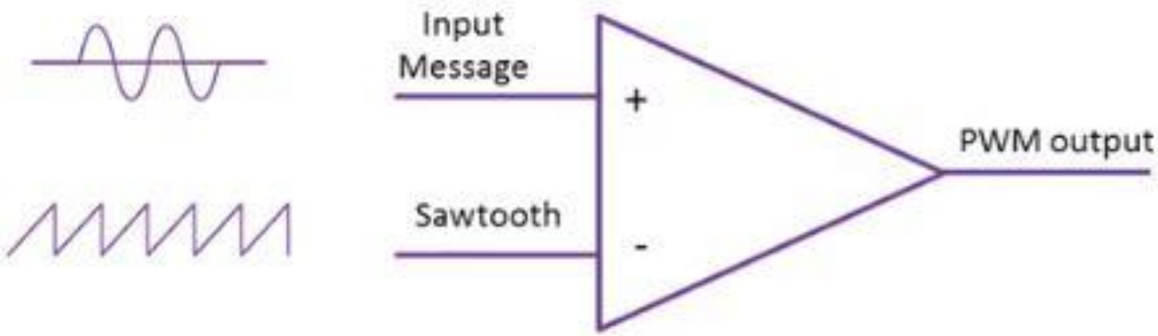
Amplitude Limiting: To maintain a constant amplitude in the pulse train, amplitude limiters are often used. These circuits clip or limit the amplitude of the signal to a desired level. This is important to prevent distortion and to keep the signal within specified bounds.

Power Control: PWM is primarily used for power control. By rapidly switching a signal (commonly referred to as a carrier signal) on and off at a high frequency, the average power delivered to a load can be controlled. The duty cycle, which is the ratio of the pulse width to the total period, determines the average power.

Average Value Control: The average value of voltage (or current) supplied to the load can be controlled by changing the duty cycle of the PWM signal. When the duty cycle is increased, the average voltage is higher, and when it's decreased, the average voltage is lower. This control mechanism is widely used in applications such as motor speed control, voltage regulation, and LED dimming.

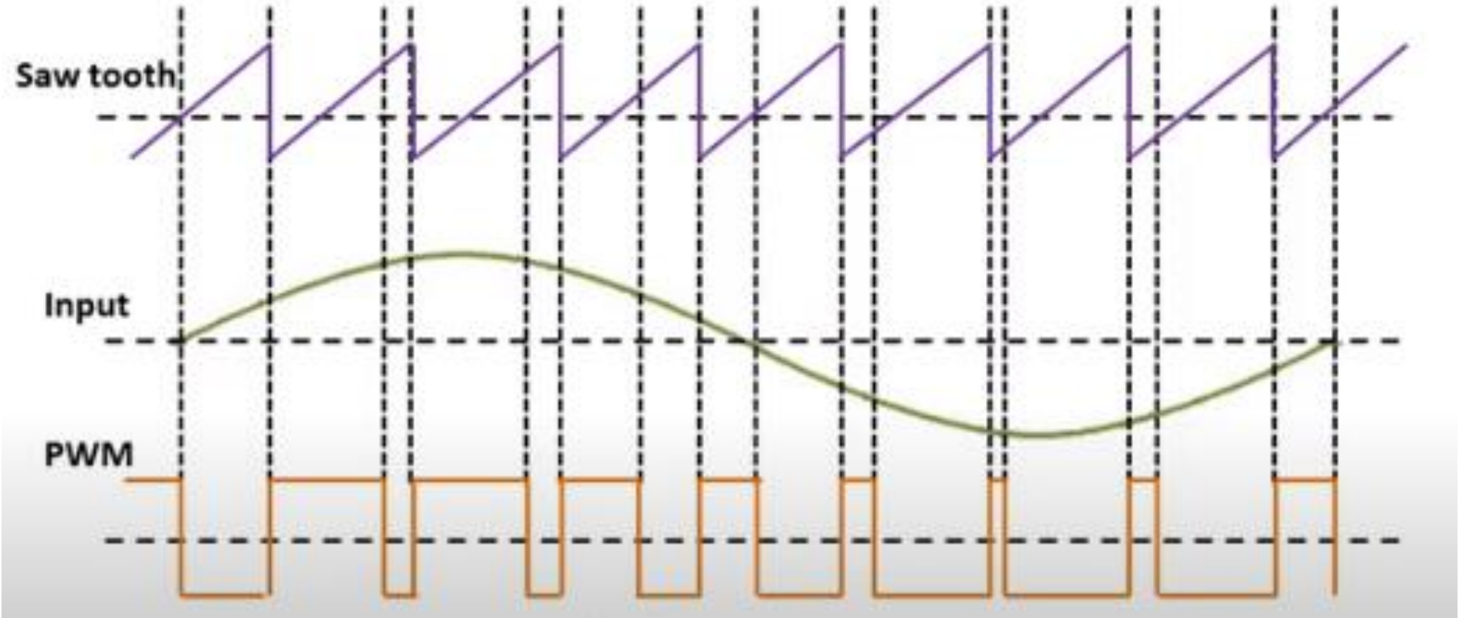
Efficiency: PWM is efficient in controlling power because it doesn't dissipate as much heat as linear regulators. Instead of reducing voltage through resistance (which causes heat dissipation), PWM switches the voltage on and off rapidly, reducing power loss.

Generation of PWM by a Comparator



Generation of PWM by a comparator

When this method is used to generate PWM, the peak value of the message signal should be less than the peak value of the Sawtooth wave form.



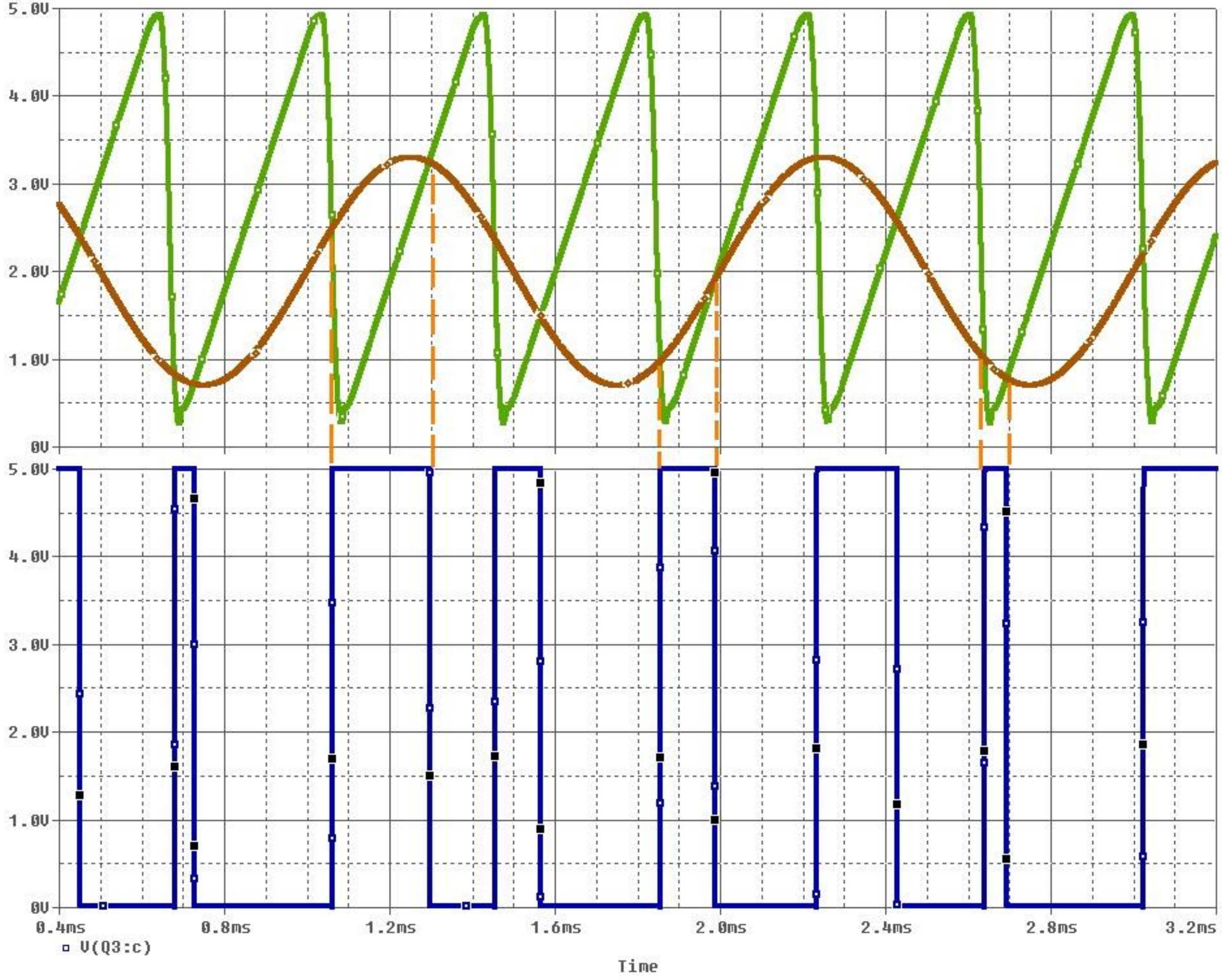
Principle of PWM generation

The Comparator produces the output depending upon the inputs (Message Signal) applied. Ex High(+ve) or law (-ve)

It produces the +ve or –ve pulses depending upon the magnitude of the message signal and the Saw-Tooth Pulses.

The Width of the pulse is dependent on the amplitude of the message Signal.

When The Input voltage is lesser than the Sawtooth voltage then the Out Put Pulse will be Negative and Vise versa.

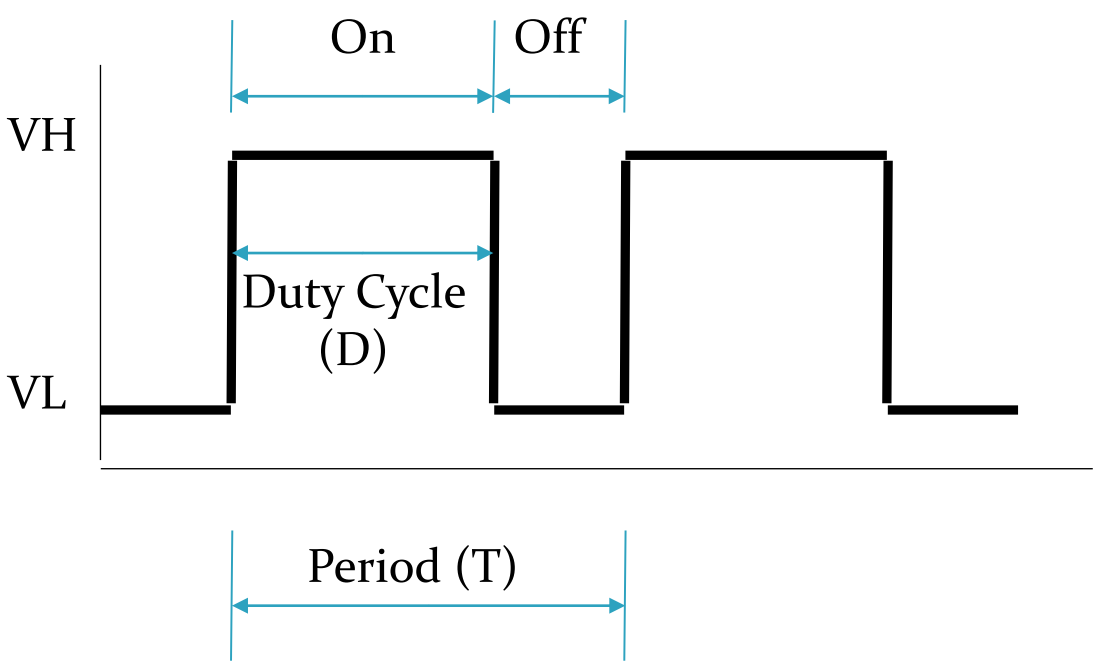

5.0V
4.0V
3.0V
2.0V
1.0V
0V
0.4ms
0.8ms
1.2ms
1.6ms
2.0ms
2.4ms
2.8ms
3.2ms
V(Q3:c)
Time

Duty Cycle

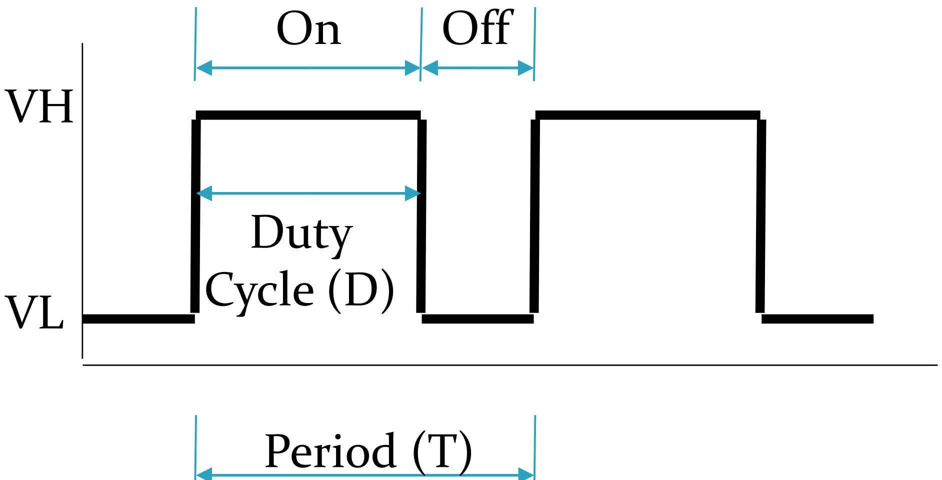
Definition: The Duty Cycle is a measure of the time the modulated signal is in its “high” state.

It is generally recorded as the percentage of the signal period where the signal is considered on.

The term duty cycle describes the proportion of 'on' time to the regular interval or 'period' of time; a low duty cycle corresponds to low power, because the power is off for most of the time.

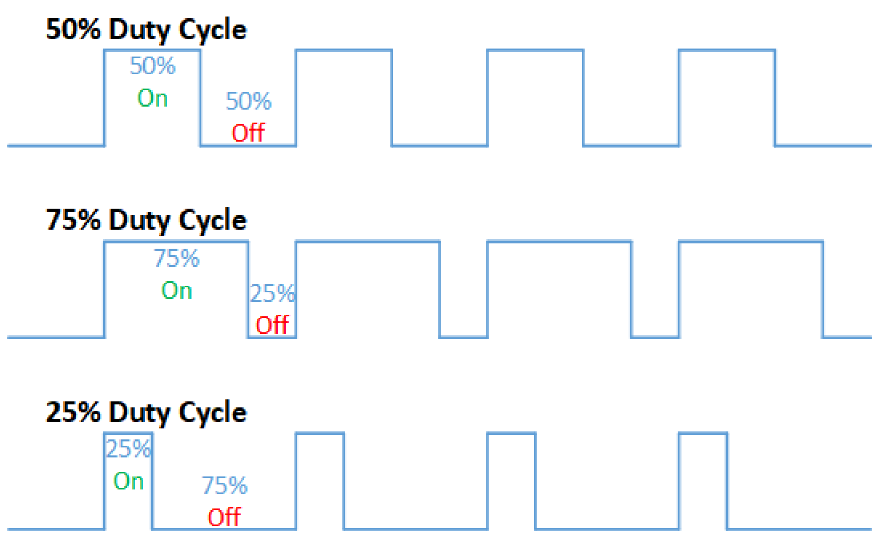


Duty Cycle Formulation

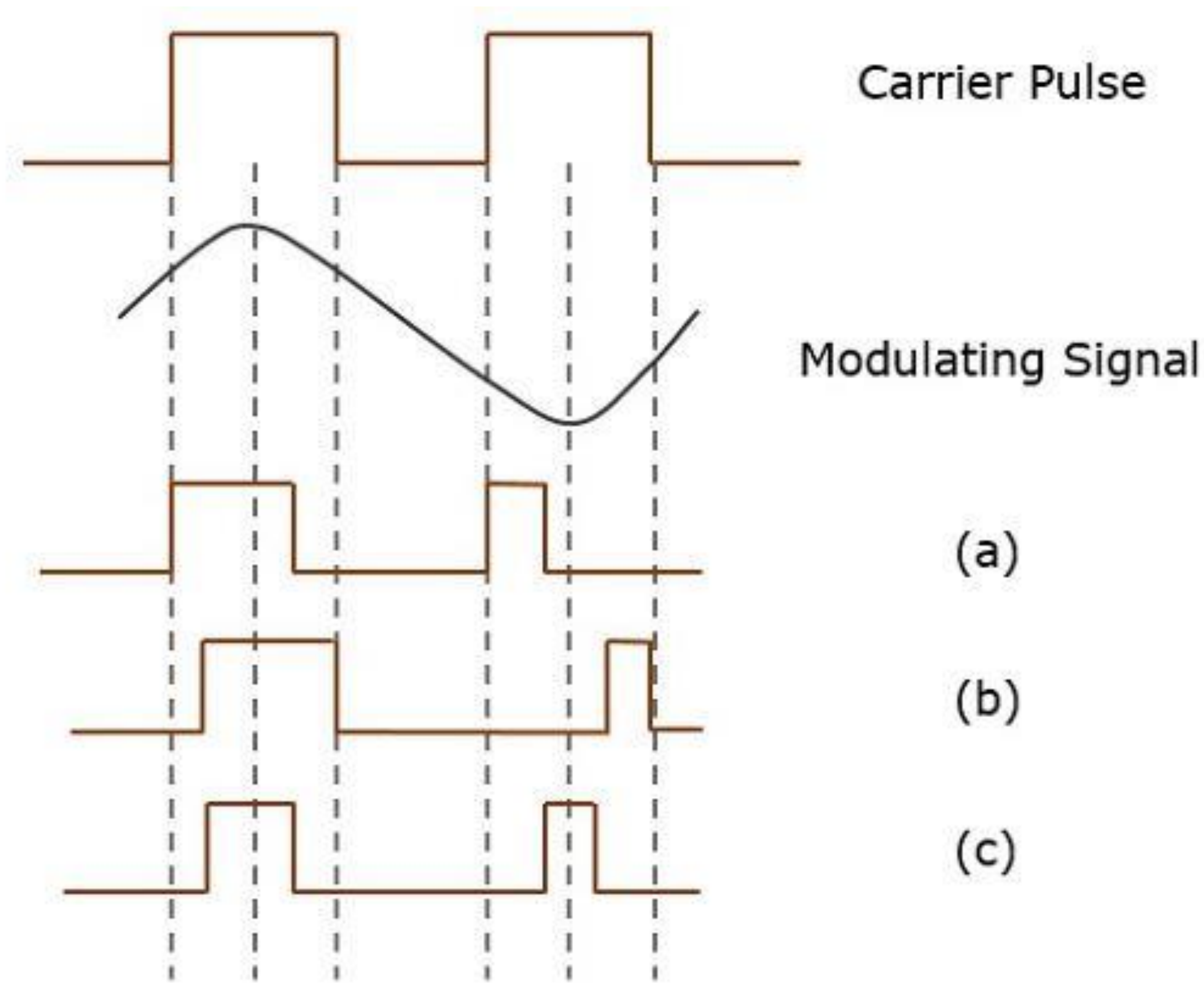


Duty Cycle is determined by:

$$Duty\ Cycle = \frac{On\ Time}{Period} \times 100\%$$



The following figures explain the types of Pulse Width Modulations.



There are three variations of PWM. They are −

(a) The leading edge of the pulse being constant, the trailing edge varies according to the message signal.

(b) The trailing edge of the pulse being constant, the leading edge varies according to the message signal.

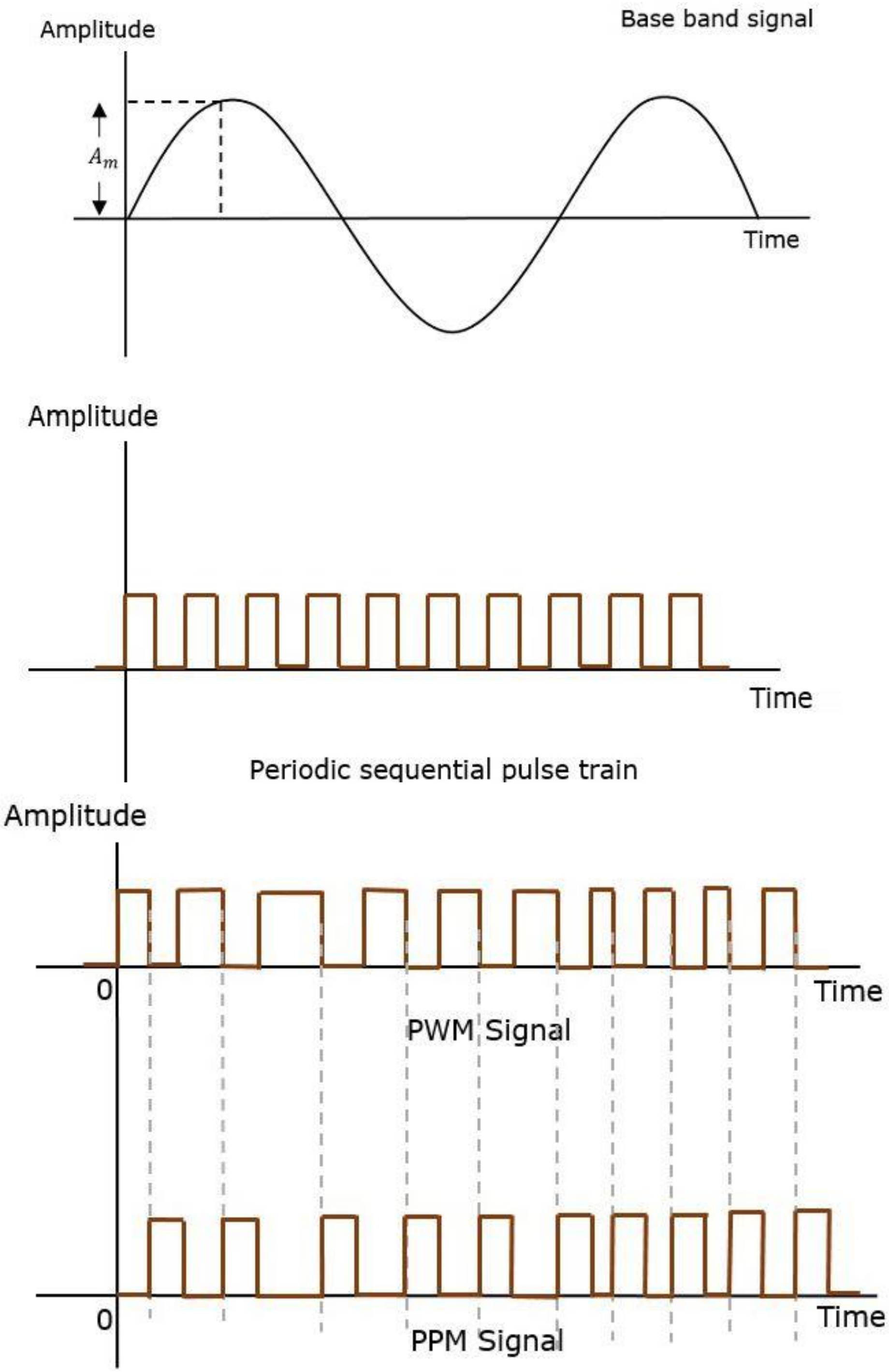
(c) The center of the pulse being constant, the leading edge and the trailing edge varies according to the message signal.

These three types are shown in the above given figure, with timing slots.

PULSE POSITION MODULATION - PPM

Pulse Position Modulation (PPM) is an analog modulating scheme in which the amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal.

The transmitter has to send synchronizing pulses (or simply sync pulses) to keep the transmitter and receiver in synchronism. These sync pulses help maintain the position of the pulses. The following figures explain the Pulse Position Modulation.


Amplitude
Base band signal
A_m
Time
Amplitude
Time
Periodic sequential pulse train
Amplitude
0
Time
PWM Signal
0
Time
PPM Signal

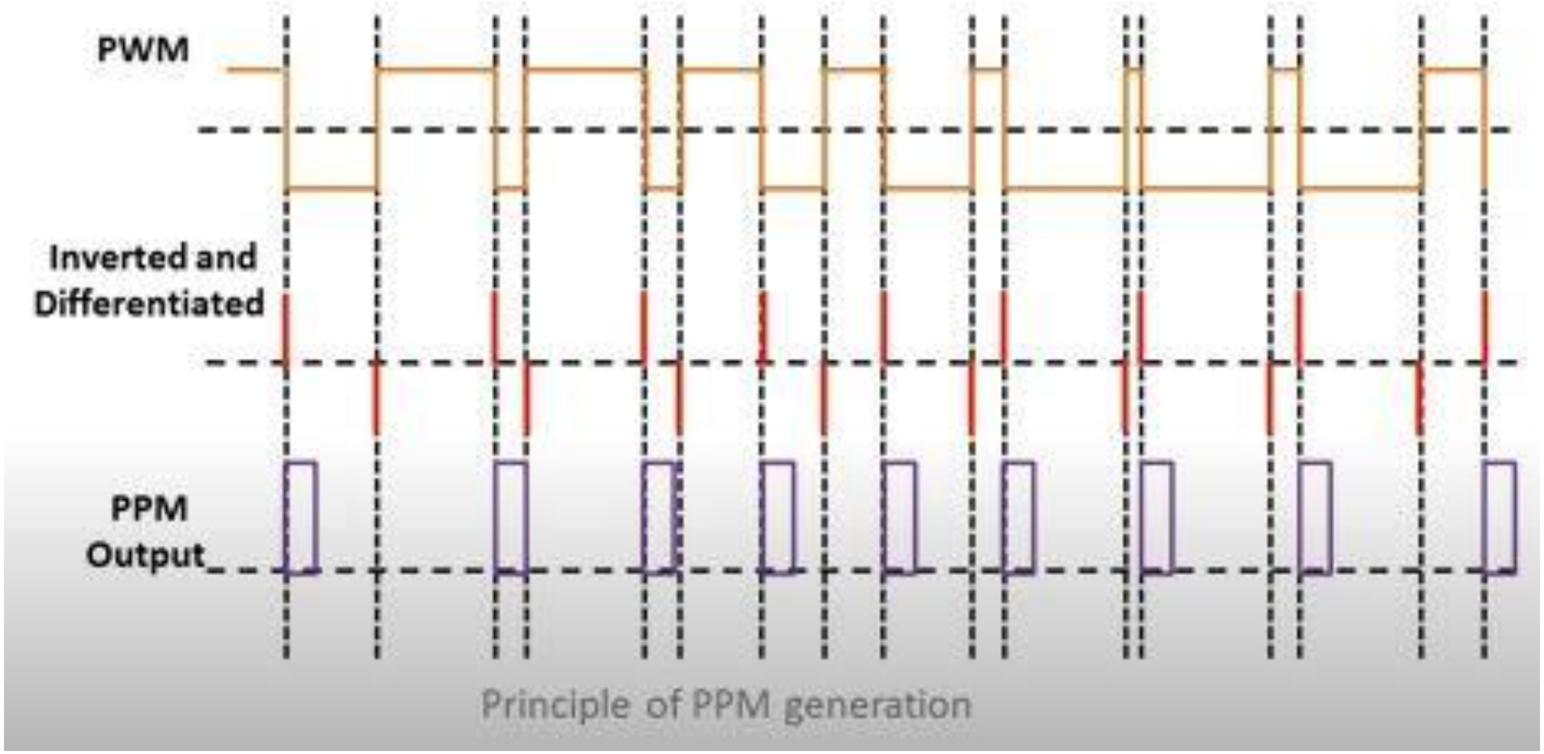
PPM generation from PWM

The input to the PPM Generator is a PWM Signal.

The inverter reverse the polarity of the pulses.

As it passes through differentiator we have;

- **+ve Spikes where original PWM pulse is going from HIGH to LAW**
- **-Ve spikes where original PWM Pulse is going from law to High**



Spikes are fed to a +ve edge triggered fixed width pulse generator.

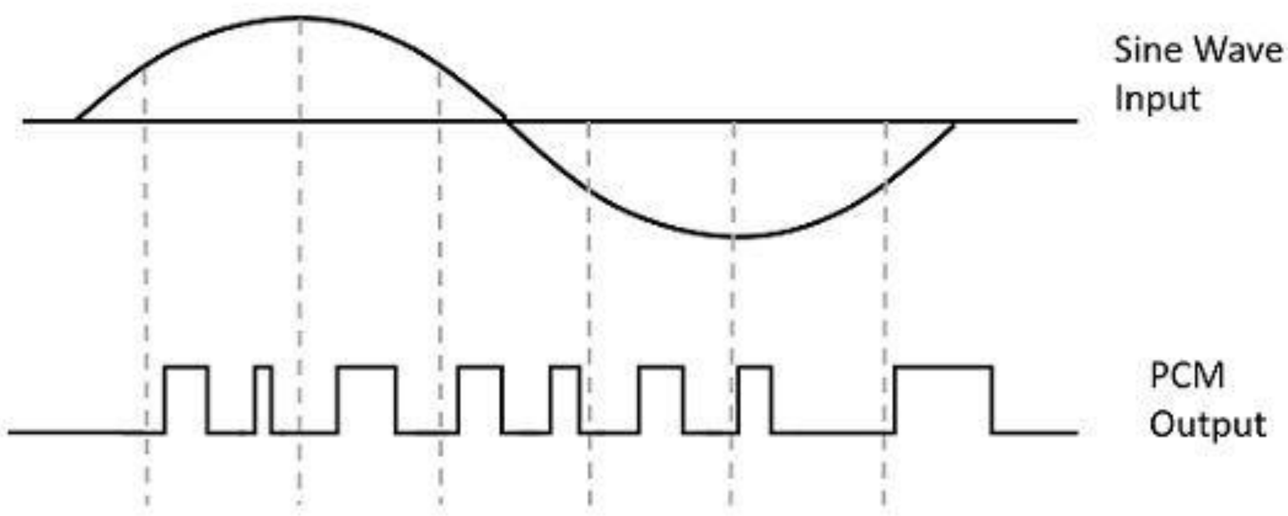
Generator generates the pulse of fixed width when only a +ve spike appears.

The occurrence of these pulses are dependent on the input message signal, hence the pulses produced is PPM.

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 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



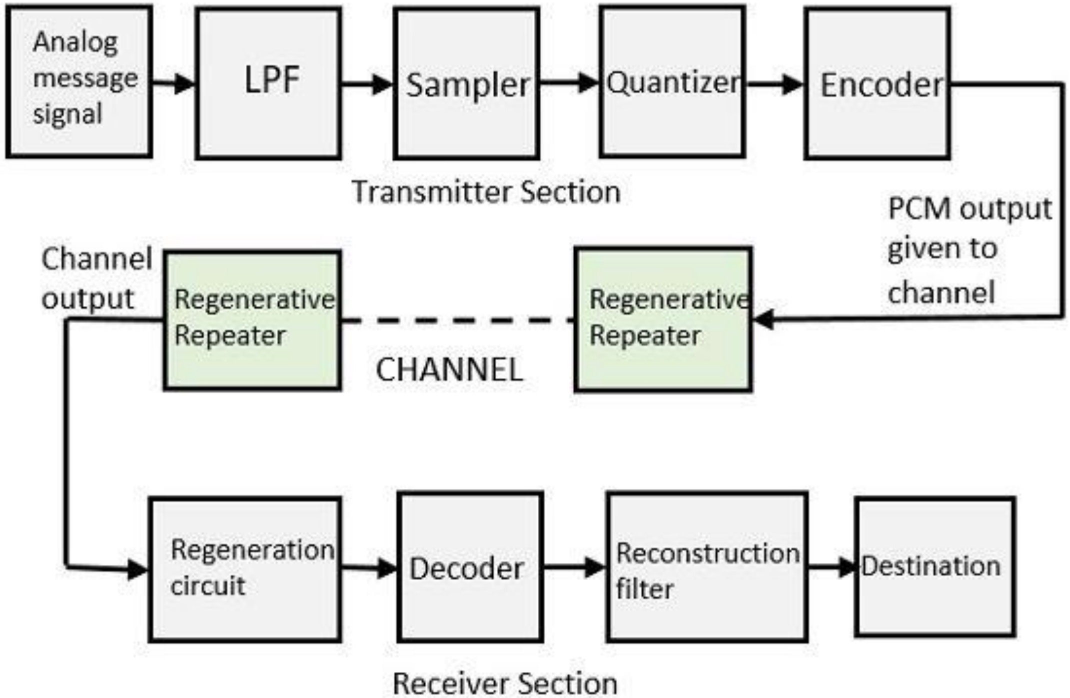
Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.

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.

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. Following is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.



Low Pass Filter

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 technique which helps to collect the sample data at instantaneous values of 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 LPF, Sampler, and Quantizer will act as an analog to digital converter. Encoding minimizes the bandwidth used.

Regenerative Repeater

This section increases the signal strength. The output of the channel also has one regenerative repeater circuit, to compensate the signal loss and reconstruct the signal, and also to increase its strength.

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 given analog signal, codes it and samples it, and then transmits it in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

QUANTIZATION

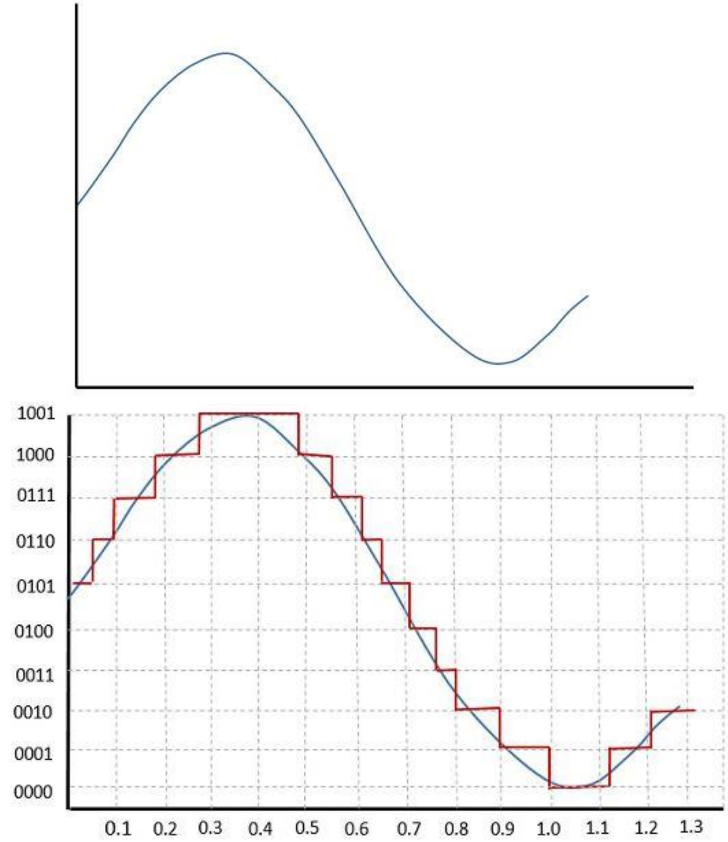
The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called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

Quantizing an Analog Signal

The analog-to-digital converters perform this type of function to create a series of digital values out of the given analog signal. The following figure represents an analog signal. This signal to get converted into digital, has to undergo sampling and quantizing.

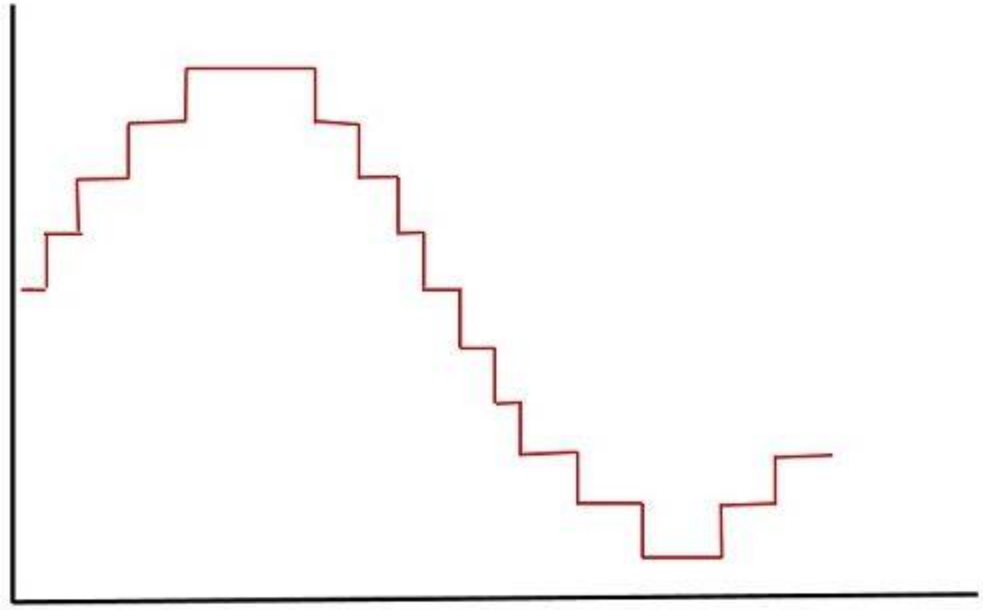
The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a continuous-amplitude sample into a discrete-time signal.

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

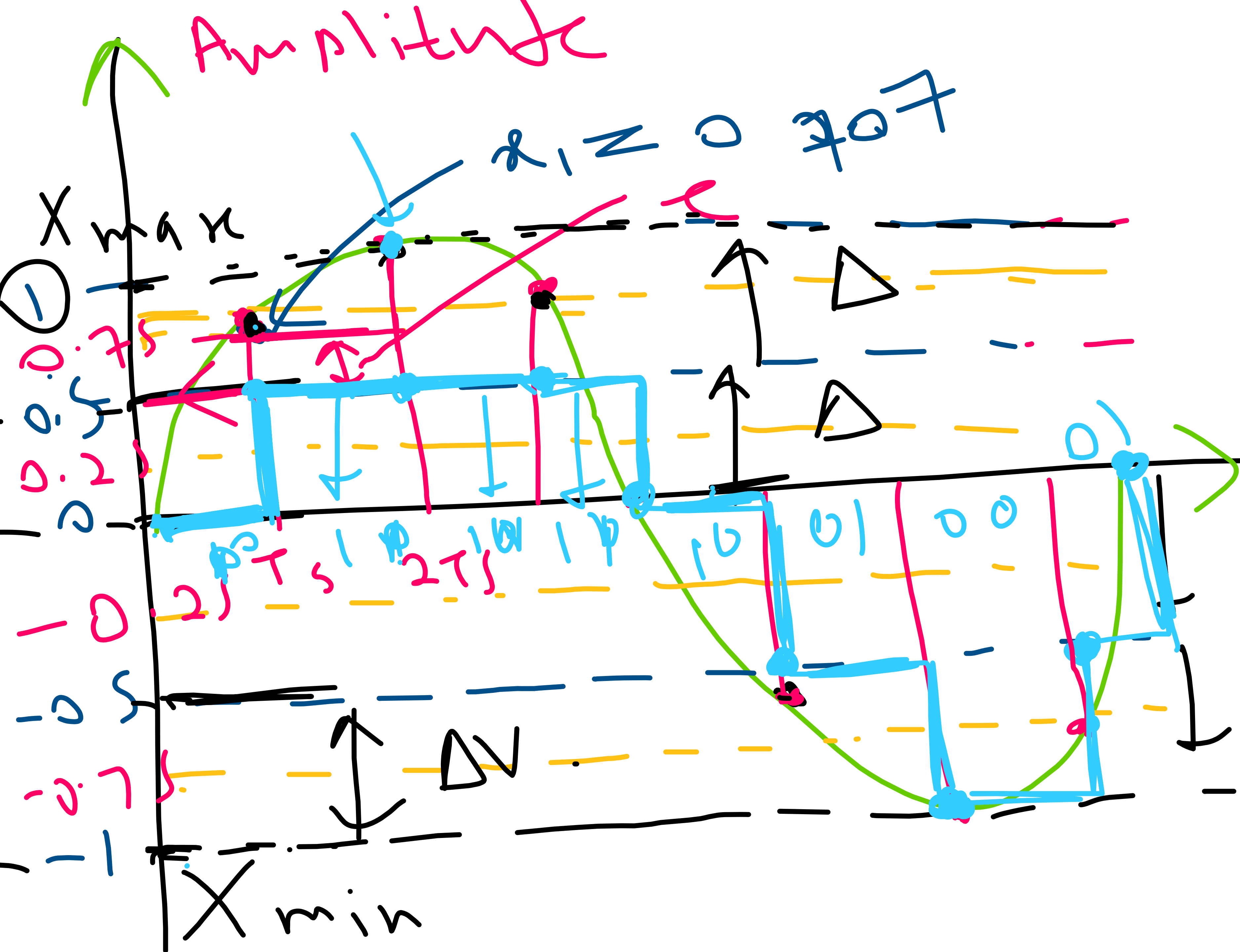


Both sampling and quantization result in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels. The spacing between the two adjacent representation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

The following figure shows the resultant quantized signal which is the digital form for the given analog signal.



This is also called as Stair-case waveform, in accordance with its shape



Amplitude
x1 = 0.707
e
Xmax
1
0.75
0.5
0.25
0
Ts
2Ts
-0.25
-0.5
-0.75
-1
Xmin
Δ
Δ
ΔV
00
01
10
11
10
01
00
00
01

① rounding off the samples to nearest quntized levels.

⇒ we need to define nu of quntized levels L

$L = 2^n$, $n = 1, 2, 3 -$

$= 2, 4, 8, 16$

$n = 2$; $L = 2^n$
$= 4$,

Let, Bit depth $= n$
$=$ no of bits

No of gun. ^{n}

* Levels $= L = 2$ *

1

* step size $= \Delta$

$$\Delta = \frac{X_{max} - X_{min}}{L}$$

2

ex:

$$\Delta = \frac{1-(-1)}{4} = \frac{2}{4}$$

$$= 0.5$$

Index value (I)

$$I = \text{Round}\left(\frac{X - X_{min}}{\Delta}\right)$$

Sampled value

③

quantized magni (X_q) ④

$$X_q = X_{min} + I\Delta r$$

ex, assume,

$X = 0.707$

$=$

$X_g = X_{min} + I\Delta$ ✓ ? ✓

$I = \text{round}\left(\dfrac{X - X_{min}}{\Delta}\right)$

$= \text{round}\left(\dfrac{0.707 - (-1)}{0.5}\right)$

$I = \text{roun}$

$$I = \text{round}(3.\underbrace{414})$$

$$\underline{\underline{I = 3}}$$

$$x_g = x_{min} + I\Delta$$

$$= -1 + 3 \times (0.5)$$

$$x_g = \underline{\underline{0.5}}$$

quntized (e)rror / quntized noise.

$e = X - X_q$

Sample value.

quntized magn

S

$X = 0.707$

$X_q = 0.50$, $e = 0.207$

X							
Sample val	0	0.707	1	0.707	0	-0.707	-1
Index (I)	2	3	4	3	2	1	0
Quant (X) val	0	0.5	1	0.5	0	-0.5	-1
X_g	10	11	11	11	10	01	00

$$I_0 = \text{round}\left(\frac{X - X_{min}}{\Delta}\right)$$

$$= r\left(\frac{0-(-1)}{0.5}\right) = 2$$

Dynamic Range

ratio of Largest to Smallest amplitude

ex, $DR = \frac{2\cancel{D}}{\left(\frac{\cancel{D}}{2}\right)} = 4$

$DR = 20 \cdot \log\left(\frac{L \cdot A}{S \cdot A}\right)$ (dB)

$$D.R = 20\log\left(\frac{L.A}{S.A}\right)$$

$$= 20\log\left(\frac{2^{n-1}}{2^{-1}}\right)$$

$$= 20\log 2^{n}$$

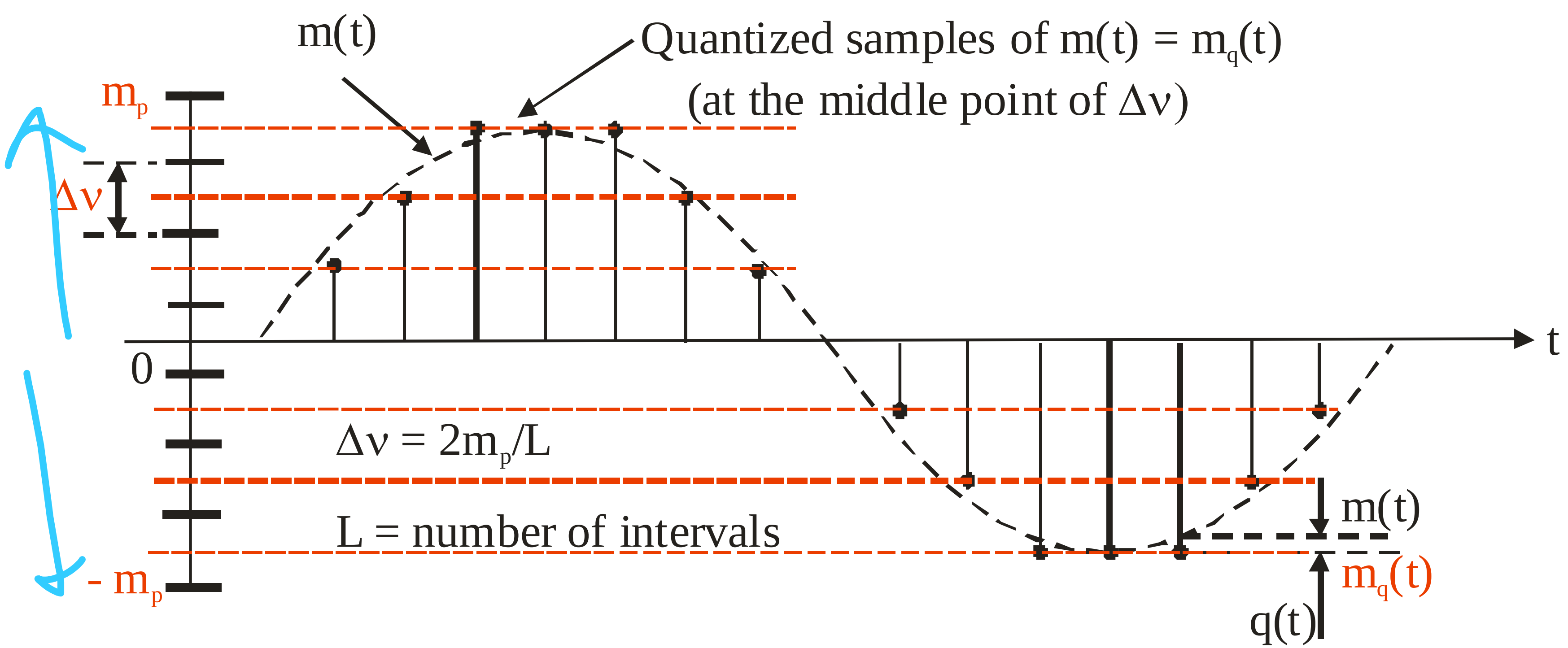
$$= 20n\log(2)$$

$$\approx 20n\,dB$$

How to perform the quantization?

1. **The amplitudes of signal m(t) lie in the *range* ($-m_p, m_p$), which is partitioned into *L intervals*, each of magnitude Step Size : $\Delta v = 2m_p/L$.**

2. **Each sample amplitude is approximated by the *midpoint value* of the interval in which the sample falls.**



Quantization of sampled analog signal

***The amplitude range*: (- m_p, m_p),**

***Interval(Step Size)*: $\Delta\nu = 2m_p/L$**

In quantization process, a sampling value is approximated by the midpoint of the interval. This introduces an *error q(t)*, defined as the *difference between the message signal m(t) and the corresponding quantized sample $m_q(t)$*,

$$q(t) = m(t) - m_q(t)$$

This error is called the *quantization noise* or quantization error.