Chapter 3 Analog Pulse Modulation

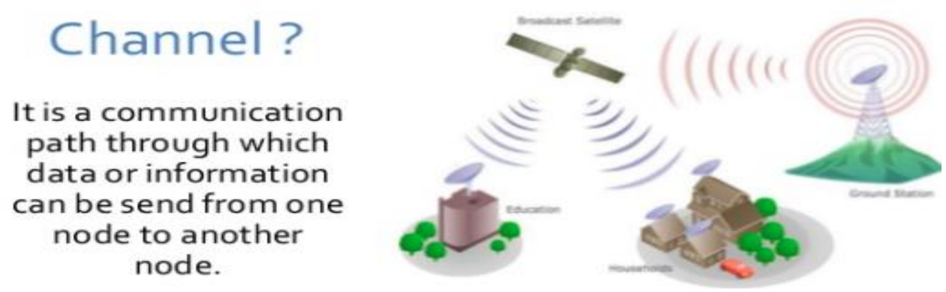
3.1 Introduction

The modulation may be defined *as the process by which some parameter of a high frequency signal termed as carrier, is varied in accordance with the signal to be transmitted.* The analog modulation may be divided into angle modulation and amplitude modulation.

A pulse is an abruptly changing voltage or current wave, which may or may not repeat itself. The simplest non-repetitive pulse is a stepped up voltage or current, a repetitive pulse train and a pulse with its trailing and leading edge.

In pulse modulation system, a train of pulses is employed as the carrier and some parameter of the pulse is varied in accordance with the instantaneous value of the modulating signal. The simplest type of pulse modulation is pulse amplitude modulation (PAM), which is similar to amplitude modulation (AM). The other pulse modulation techniques include pulse width/duration modulation (PWM), pulse position modulation (PPM).

3.2 What is mean by Channel in Electronics Communication?



3.3 Channel capacity:

The maximum rate at which data can be transmitted over a given communication path, or channel, under given conditions, is referred to as the channel capacity. There are four concepts that relate to one another.

- Data rate: The rate, in bits per second (bps), at which data can be communicated.
- Bandwidth: The bandwidth of the transmitted signal as constrained by the transmitter and the nature of the transmission medium, expressed in cycles per second, or Hertz
- Noise: The average level of noise over the communications path
- Error rate: The rate at which errors occur, where an error is the reception of a 1 when a 0 was transmitted or the reception of a 0 when a 1 was transmitted.

Signal to Noise Ratio (SNR)

To measure the quality of a system the SNR is often used. It indicates the strength of the signal with respect to the noise power in the system. It is the ratio between two powers.

SNR=average signal power / average noise power.

It is usually given in dB and referred to as SNR_{dB}.

$$SNR_{dB} = 10 \log_{10} SNR$$
$$= 10 \log_{10} (\text{average signal power / average noise power})$$

A very important consideration in data communications is how fast we can send data, in bits per second. over a channel. Data rate depends on three factors:

1. The bandwidth available

2. The level of the signals we use

3. The quality of the channel (the level of noise) two theoretical formulas was developed to calculate the data rate: one by Nyquist for a noiseless channel and another by Shannon for a noisy channel.

3.4 Nyquist Bandwidth:

In noiseless environment, the limitation on data rate is simply the bandwidth of the signal. A formulation of this limitation, due to Nyquist, states that “if the rate of signal transmission is 2B, then a signal with frequencies no greater than B is sufficient to carry the signal rate”. The converse is also true:” Given a bandwidth of B, the highest signal rate that can be carried is 2B”.

$$C = 2 \times B \times \log_2 M$$

Where C is Bit Rate or capacity, B is Bandwidth, M is number of levels.

Example: Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as

$$\boxed{\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}}$$

3.5 Shannon Capacity Formula

In reality, we cannot have a noiseless channel; the channel is always noisy. In 1944, Claude Shannon introduced a formula, called the Shannon capacity, to determine the theoretical highest data rate for a noisy channel:

$$C = B \log_2 \left(1 + \frac{S}{N}\right) bits/second$$

where

C is channel capacity in bits per second (bps)

W is bandwidth of the channel in Hz

S/N is the signal-to-noise power ratio (SNR). SNR generally is measured in dB using the formula

Shannon Theorem states that if the actual information rate on a channel is less than the error-free capacity, then it is theoretically possible to use a suitable signal code to achieve error-free transmission through the channel.

Example: Let's calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000. The signal-to-noise ratio is usually 3162. What is the channel capacity?

$$C = B \log_2 (1 + \mathrm{SNR}) = 3000 \log_2 (1 + 3162) = 3000 \log_2 3163$$
$$= 3000 \times 11.62 = 34{,}860 \text{ bps}$$

3.6 Nyquist sampling theorem

The Nyquist sampling theorem provides a prescription for the nominal sampling interval required to avoid aliasing. It may be stated simply as follows:

The sampling frequency should be at least twice the highest frequency contained in the signal.

Or in mathematical terms:

$$f_s \geq 2\, f_c$$

where *fs* is the sampling frequency (how often samples are taken per unit of time or space), and fc is the highest frequency contained in the signal.

If this is the case, we have not lost any information during the sampling process and we could theoretically reconstruct the original signal from the sampled signal.

Alternatively we can define a Nyquist frequency based on a certain sampling frequency:

$$f_{\text{Nyquist}} = \frac{1}{2} f_{\text{sample}}.$$

Any signals that contain frequencies higher than this Nyquist frequency cannot be perfectly reconstructed from the sampled signal, and are called undersampled. If our signal only contains frequencies smaller than the Nyquist frequency, we can perfectly reconstruct the original signal given the sampled signal, and we are oversampled. When our signal is band limited to a frequency equal to the Nyquist frequency, we are critically sampled.

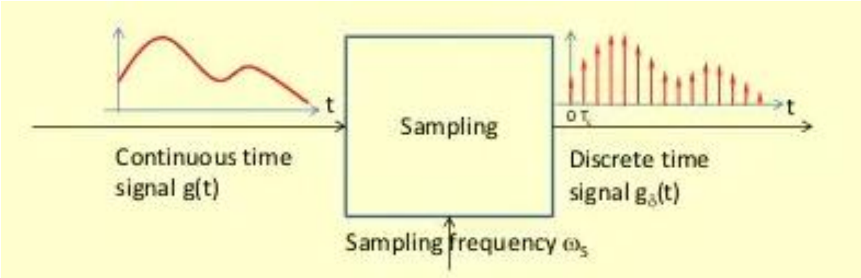


Figure 1.1: Concept of sampling

3.9 Concept of Aliasing

So what happens when a signal is undersampled? Aliasing occurs.

If the sampling frequency is less than twice the maximum analog signal frequency, a phenomenon known as aliasing will occur.

Figure 1.2 we show the process of sampling two different signals (in yellow). Both signals are sampled with the same sampling frequency at points in red. The top signal is oversampled, i.e., its frequency is lower than half the sampling frequency, so we have more than two samples per period of this sinusoid. Here we can perfectly reconstruct the original signal. The bottom signal, however, is undersampled. We have less than two samples per period of this sinusoid and when we try to reconstruct the signal (blue line), we are not reconstructing the original signal, but rather a much lower frequency. This effect is called aliasing. If we are undersampled, the frequencies that are higher than the Nyquist frequency are reconstructed at lower frequencies and will add noise to the actual signal at those lower frequencies.

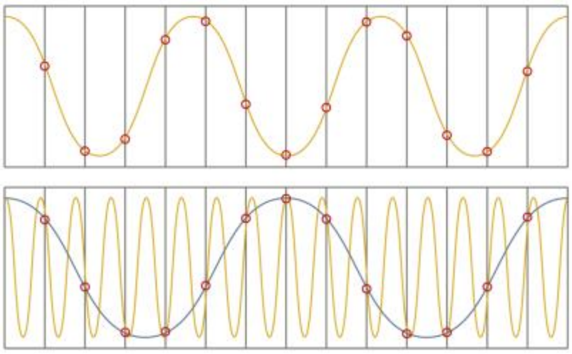
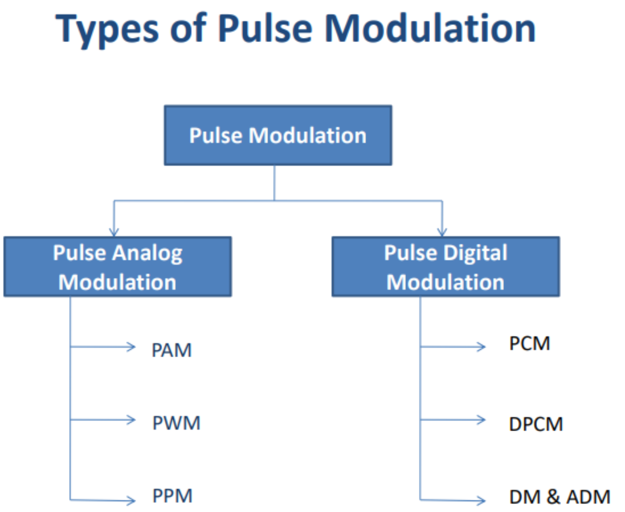


Figure 1.2: Aliasing. The top signal (in yellow) is oversampled (samples in red), while the bottom signal is undersampled. The reconstructed signal (in blue) from the sampled data yields a much lower frequency than the original signal. This is called aliasing

3.8 Pulse Modulation

Pulse modulation is a technique in which the signal is transmitted with the information by pulses. This is divided into Analog Pulse Modulation and Digital Pulse Modulation.

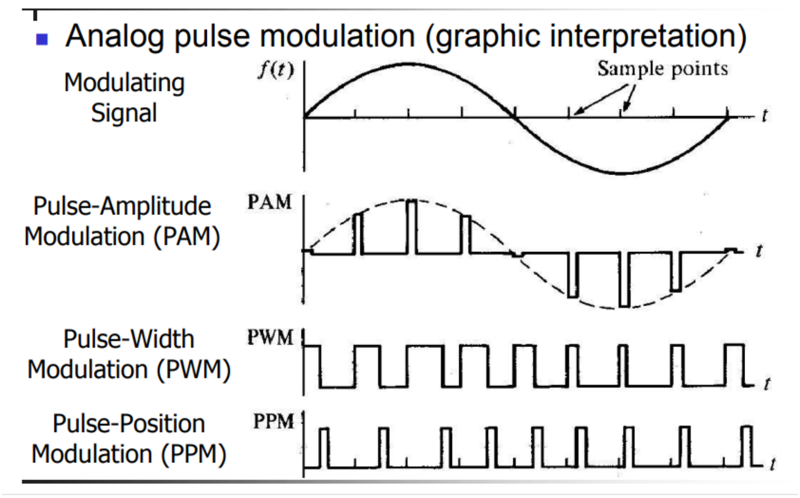


Analog pulse modulation is classified as

- Pulse Amplitude Modulation (PAM)
- Pulse Width Modulation (PWM)
- Pulse Position Modulation (PPM)

Digital pulse modulation is classified as

- Pulse Code Modulation
- Delta Modulation



3.8.1 Pulse Amplitude Modulation

Pulse amplitude modulation is a technique in which the amplitude of each pulse is controlled by the instantaneous amplitude of the modulation signal. It is a modulation system in which the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the signal at the instant of sampling. This technique transmits the data by encoding in the amplitude of a series of signal pulses.

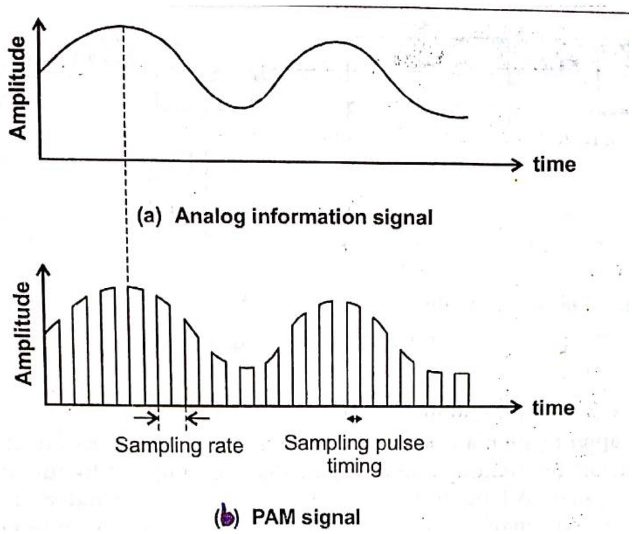


Figure 2.6: PAM Waveforms

As figure 2.6 (b) indicates, the pulse amplitude varies in direct proportion to the sample values of information. It is evident from the PAM waveform that a PAM signal has significant dc content and that the bandwidth required to preserve the shape is much greater than the information bandwidth. Therefore PAM is seldomly used alone as a single channel communication system. But its major role is in time division multiplexing, data telemetry and instrumentation system. It is an intermediate form of modulation with PSK, QAM and PCM. (Described later on)

PAM Modem

PAM wave could be generated using following circuit consisting of monostable multivibrator and gate Analog input passes through a gate circuit which allows the signal to pass when the gate is 'OPEN' and block the signal when it is 'CLOSED'. The gate circuit can be constructed using diodes, BJT's or FET's.

The ON-OFF timing of gate is controlled using a one-shot multivibrator. It generates a narrow fixed width pulse which opens or closes the gate. In order to perform this operation at fixed intervals, the multivibrator is triggered by a clock signal. The multivibrator triggers once at each clock pulse. The clock frequency is decide by the sampling rate while quasistate state timing decides the sampling pulse timing.

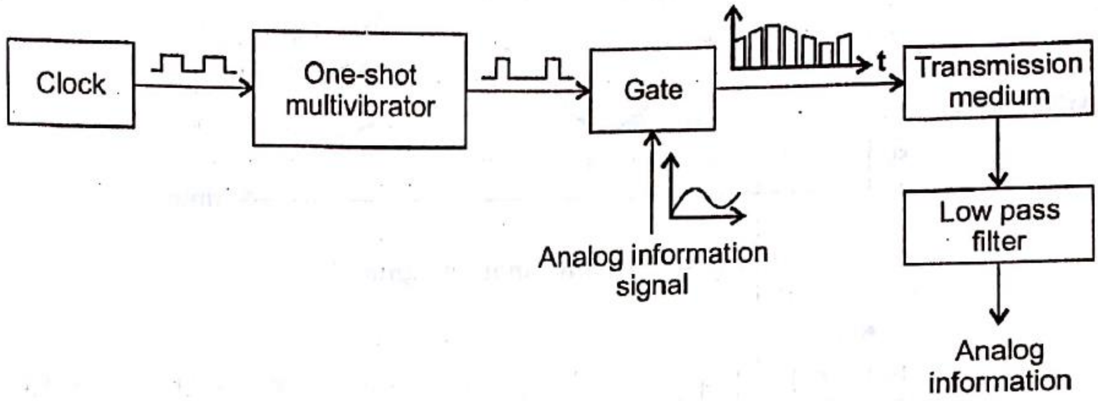


Figure 2.7: PAM Modulation

To recover the original information, the PAM signal is simply passed through a low pass filter. This filter often, referred to as a reconstruction filter, have a frequency response shaped in such a way as to correct for distortion in received signal. Usually the filter cut-off frequency is kept higher than the highest frequency component in the analog signal. All other higher frequencies are eliminated. Since the pulses themselves contain many high frequency harmonics, they are filtered out. Thus, the pulses are smoothed into a continuous analog signal closely resembling the original information signal.

Drawback of PAM is that it is very susceptible to noise. The precise value of the pulse height is significant in PAM system and any variation due to noise will distort the signal. As compared to it, PPM and PWM have better noise performance.

Demodulation of PAM Signals

For pulse amplitude modulated (PAM) signals, the demodulation is done using a Holding circuit. Figure.1.3 shows the block diagram of a PAM demodulator.

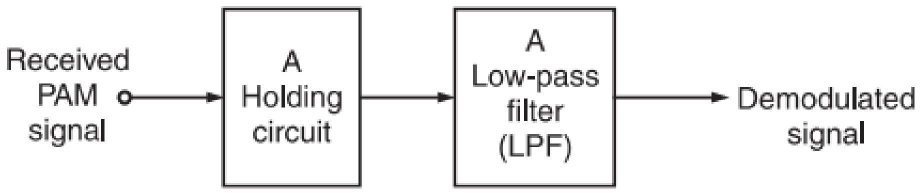


Figure. 1.3: Block diagram of PAM demodulator

In this method, the received PAM signal is allowed to pass through a Holding circuit and a low pass filter (LPF) as shown in figure.1.3.

Now, figure.1.4 illustrates a very simple holding circuit.

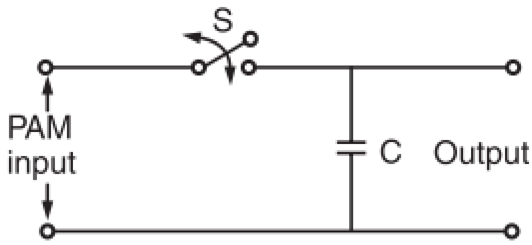


Figure 1.4: A zero-order holding circuit

Here the switch 'S' is closed after the arrival of the pulse and it is opened at the end of the pulse. In this way, the capacitor C is charged to the pulse amplitude value and it holds this value during the interval between the two pulses.

Hence, the sampled values are held as shown in figure 1.5.

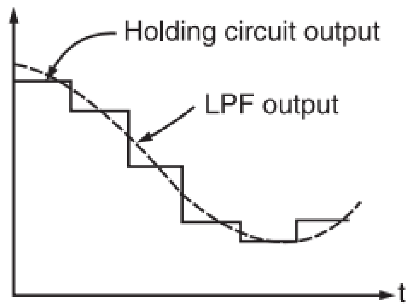


Figure 1.5: the output of a Low Pass filter (LPF)

After this the holding circuit output is smoothened in Low Pass filter as shown in figure 1.5.

Advantages of PAM :

- It is the simple and simple process for modulation and demodulation
- Transmitter and receiver circuits are simple and easy to construct.

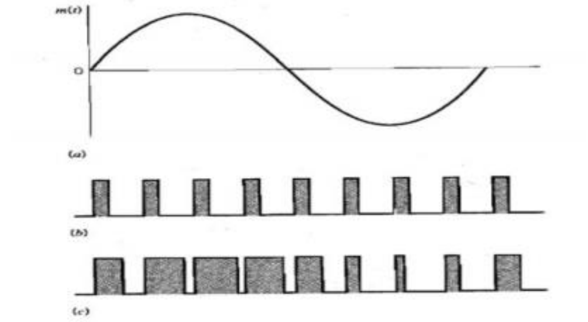
Drawbacks of PAM signal

- The bandwidth required for the transmission of a PAM signal is very large in comparison to the maximum frequency present in the modulating signal.
- Since the amplitude of the PAM pulses varies in accordance with the modulating signal therefore the interference of noise is maximum in a PAM signal. This noise cannot be removed easily.
- Since the amplitude of the PAM pulses varies, therefore, this also varies the peak power required by the transmitter with modulating signal.

3.8.2 Pulse Width Modulation (PWM)

Definition:

In PWM, Width of the pulses of the carrier pulse train is varied in accordance with the modulating signal.



6/5/2020 **Figure:** Illustration of PWM (a) Modulating signal (b) Pulse Carrier (c) PWM signal Prof.Ch.Srinivasa Rao - JNTUK - UCEV 29

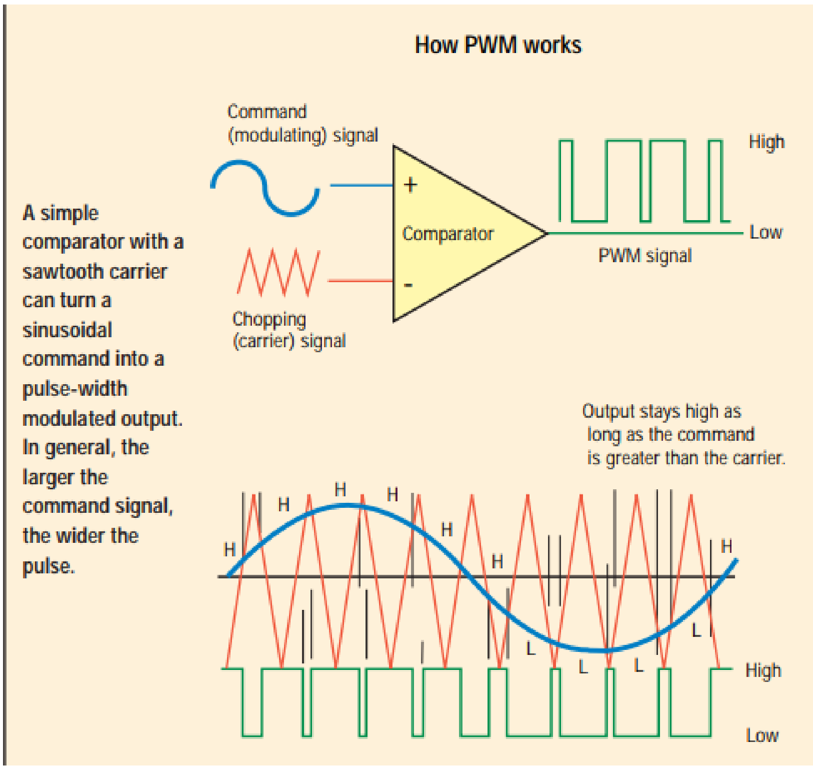


Figure 1.6 (B): Working of Pulse width modulation

Generation of Pulse Width Modulation (PWM) Signal:

Pulse Width Modulating signal can be generated using a Comparator as shown in the figure1.6 (B). Modulating signal forms one of the input to the Comparator and the other input is fed with a non-sinusoidal wave or sawtooth wave. It operates at carrier frequency. The Comparator compares the two signals and generates a PWM signal as its output waveform.

If the value of the Sawtooth triangle signal is more than the modulation signal then the PWM output signal is at "Low" else it's in "High" state. Thus, the value of the input signal magnitude determines the comparator output which defines the width of the pulse generated at the output.

Advantages of PWM:

- Noise is less, since in PWM, amplitude is held constant.
- Signal and noise separation is very easy
- PWM communication does not required synchronization between transmitter and receiver.

Disadvantages of PWM:

- Power will be variable because of varying in width of pulse.
- Bandwidth required for PWM communication is large as compared to the pulse amplitude modulation.

Applications of PWM

- It is used in telecommunications, brightness controlling of light or speed controlling of fans etc.
- Embedded application

3.8.3 Pulse Position Modulation

- In PPM amplitude and width of the pulses are kept constant but the position of each pulse is varied in accordance to the amplitudes of the sampled values of the modulating signal.
- The PPM pulses can be derived from the PWM pulses as shown in the fig. Note that with increase in the modulating voltage the PPM pulses shift further with respect to reference.
- The vertical dotted lines drawn in fig. are treated as reference lines to measure the shift in the position of PPM pulses. The PPM pulses marked 1,2 and 3 in fig. go away from their respective reference lines. This corresponds to increase in the modulating signal voltage.
- Then as the modulating voltage decreases the PPM pulses 4,5,6,7 come progressively closer to their respective reference lines.

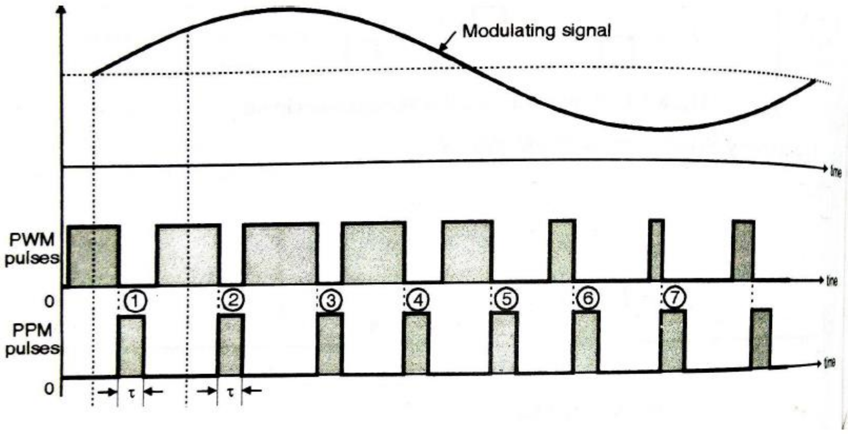


Fig: Generation of PPM pulses with the help of PWM pulses

Generation of PPM Signal

- PPM signal can be generated from PWM signal.
- PWM pulses obtained at the comparator output are applied to a negative edge triggered monostable multivibrator.
- Therefore corresponding to each trailing edge of PWM, the monostable output goes high. It remains high for a fixed time decided by its own RC component.
- Thus as the trailing edge of the PWM signal keep shifting in proportion with the modulating signal x(t),the PPM pulses also keep shifting.

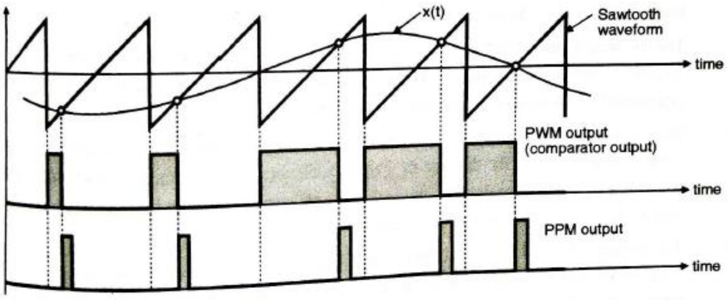


Fig: Waveforms

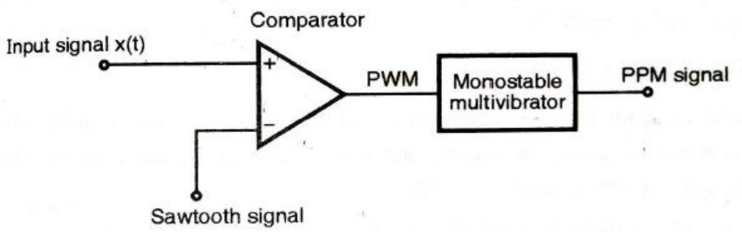


Fig: PPM Generation Circuit

Advantages of PPM

- Since information is not contained in the amplitude of PPM pulses therefore noise doesn't affect the information. Thus it has good noise immunity.
- PPM signal can be reconstructed from the noise contaminated PPM signal. This is possible in PWM but not in PAM.
- Due to constant amplitude of pulses, the transmitted power always remains constant.

Disadvantages of PPM

- As the positions of the PPM pulses are varied with respect to a reference pulses, transmitter has to send synchronizing pulses to operate the timing circuits in the receiver. Without synchronizing pulses, demodulation wouldn't be possible.
- Large BW is required to obtain undistorted pulses at receiver.

Comparison between PAM, PWM, and PPM.

The comparison between the above modulation processes is presented in a single table.

PAM	PWM	PPM
Amplitude is varied	Width is varied	Position is varied
Bandwidth depends on the width of the pulse.	Bandwidth depends on the rise timeof the pulse.	Bandwidth depends on the rise time of the pulse.
Instantaneous transmitter power varies with the amplitude of the pulses	Instantaneous transmitter power varies with the amplitude and width of the pulses.	Instantaneous transmitter power remains constant with the width of the pulses.
System complexity is high	System complexity is low	System complexity is low.
Noise interference is high	Noise interference is low	Noise interference is low.

Multiple choice questions

1. According to Nyquist theorem, sampling frequency should be ___________.

A]	equal to input signal frequency	B]	twice the input signal frequency
C]	less than input signal frequency	D]	none of these

2. The output of sampling process are called as ----------

A]	Pulse code modulation	B]	Pulse amplitude modulation
C]	Frequency modulation	D]	Amplitude modulation

3. Type of analog pulse modulation systems are -----

A]	FM	B]	PCM
C]	PAM	D]	All of these

4. How many voltage levels are present in a PWM signal?

A]	0	B]	1
C]	2	D]	3

5. In pulse amplitude modulation,

A]	Amplitude of the pulse train is varied	B]	Width of the pulse train is varied
C]	Frequency of the pulse train is varied	D]	None of the above

6. Which of the following requires a synchronizing signal?

A]	PAM	B]	PWM
C]	PPM	D]	All of these

7. When the amplitude of pulses is varied to represent analog information, the method is called -----.

A]	PAM	B]	PWM
C]	PPM	D]	All of these

8. In ------, the amplitude and width of pulse are constant.

A]	PAM	B]	PWM
C]	PPM	D]	All of these

13. Shannon's channel capacity is ---------

A]	C=B x log 2(1+S/N)	B]	C=B x 2(1+S/N)
C]	C=B x log 2(1+N/S)	D]	none of the above

Short Answer Questions

1. What is meant by channel capacity? Give the concept of Shannon's channel capacity.
2. Which are the different factors that decide the channel capacity?
3. State the Nyquist sampling theorem. Determine the minimum Nyquist sampling frequency for a maximum analog input frequency of 2KHz.
4. Explain the concept of PPM.
5. Explain the concept of PWM.
6. Explain the concept of PAM.

Long Answer Questions

1. Explain the basic principle of PAM with its block diagram and its waveform.
2. What is the basic idea if pulse width modulation? Draw it's block diagram and waveform.