PAM, PPM, PWM
Modulation and Demodulation
Trainer ST2110

Learning Material

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PAM-PPM-PWM Modulation & Demodulation Trainer
ST2110
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Safety Instructions

Read the following safety instructions carefully before operating the instrument. To avoid any personal injury or damage to the instrument or any product connected to it.

Do not operate the instrument if suspect any damage to it.

The instrument should be serviced by qualified personnel only.

For your safety:

Use proper Mains cord	:	Use only the mains cord designed for this instrument. Ensure that the mains cord is suitable for your country.
Ground the Instrument	:	This instrument is grounded through the protective earth conductor of the mains cord. To avoid electric shock the grounding conductor must be connected to the earth ground. Before making connections to the input terminals, ensure that the instrument is properly grounded.
Observe Terminal Ratings	:	To avoid fire or shock hazards, observe all ratings and marks on the instrument.
Use only the proper Fuse	:	Use the fuse type and rating specified for this instrument.
Use in proper Atmosphere	:	Please refer to operating conditions given in the manual.

1. **Do not operate in wet / damp conditions.**
2. **Do not operate in an explosive atmosphere.**
3. **Keep the product dust free, clean and dry.**

Features

- Self contained Trainer
- PAM-PPM-PWM Modulation and Demodulation techniques using Natural and Flat-top sampling
- Analog Sample, Sample and Hold and Flat-top outputs.
- Selectable 4 pulse frequencies on board.
- On board Sine and Square wave Generators
- Voice Communication using Dynamic Mic and Speaker
- On Board Filter and AC Amplifier
- Functional Blocks indicated on board mimics
- Input-Output and test points provided on board.
- Built in DC Power Supply.
- Operating manual
- Eight Switched Faults
- Compact Size.

RoHS Compliance

ROHS Compliant

Scientech Products are RoHS Complied.

RoHS Directive concerns with the restrictive use of Hazardous substances (Pb, Cd, Cr, Hg, Br compounds) in electric and electronic equipments.

Scientech products are “Lead Free” and “Environment Friendly”.

It is mandatory that service engineers use lead free solder wire and use the soldering irons upto (25 W) that reach a temperature of 450°C at the tip as the melting temperature of the unleaded solder is higher than the leaded solder.

Technical Specifications

Pulse Modulation	:	1. Pulse Amplitude Modulation 2. Pulse Position Modulation 3. Pulse Width Modulation
On board Sampling Frequency (Pulse)	:	8 KHz, 16 KHz, 32 KHz, 64 KHz
On board Generators	:	1. Sinewave 1 KHz & 2 KHz (Gain Adjustable) 2. Square wave 1 KHz & 2 KHz
Low Pass Filter	:	4^{th} order Butter worth Filter
Voice Communication	:	Voice Link using Dynamic Mic & Speaker
AC Amplifier	:	With adjustable gain control
DC Output	:	0- 4 V
Switched Faults	:	8 in numbers
Test Points	:	29 in numbers
Interconnections	:	2 mm Sockets
Power Supply	:	230 V ± 10%; 50 Hz
Power Consumption	:	3 VA (approximately)
Dimensions (mm)	:	W420 x H100 x D255
Weight	:	3.0 Kgs. (approximately)

Theory

Introduction of Pulse Modulation:

Pulse modulation may be used to transmit information, such as continuous speech or data. It is a system in which continuous waveforms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times, together with any synchronizing pulses that may be required. At the receiving end, the original waveforms may be reconstituted from the information regarding the samples, if these are taken frequently enough. Despite the fact that information about the signal is not supplied continuously, as in AM and FM, the resulting receiver output can have negligible distortion.

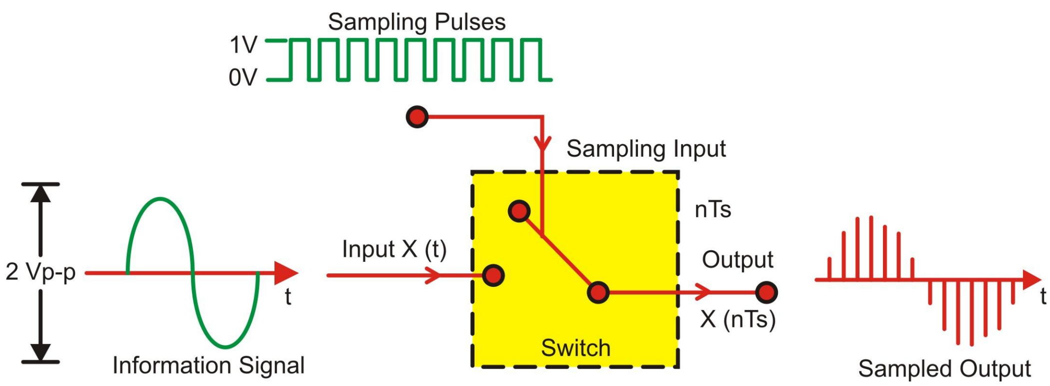
Pulse modulation may be subdivided broadly into two categories, analog and digital. In the former case, the indication of sample amplitude is the nearest variable, while in the latter case, a code, which indicates the sample amplitude to the nearest predetermined level, is sent. Pulse amplitude and pulse time modulation, to be treated next, are both analog.

Theory of sampling:

The signals we use in the real world, such as our voice, are called "analog" signals. To process these signals for digital communication, we need to convert analog signals to "digital" form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. To convert continuous time signal to discrete time signal, a process is used called as sampling. The value of the signal is measured at certain intervals in time. Each measurement is referred to as a sample.

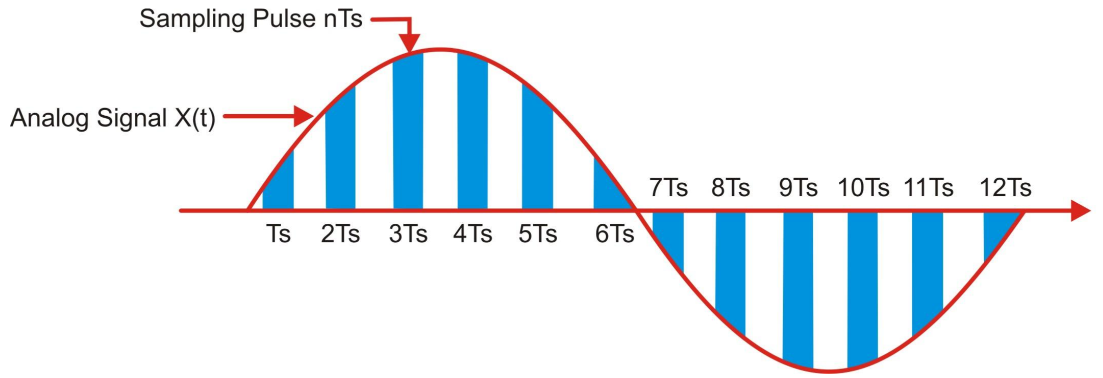
Principle of sampling:

Consider an analogue signal x(t) that can be viewed as a continuous function of time, as shown in figure1. We can represent this signal as a discrete time signal by using values of x(t) at intervals of nTs to form x(nTs) as shown in figure 1. We are "grabbing" points from the function x(t) at regular intervals of time, Ts, called the sampling period.



Basic Sampling Process

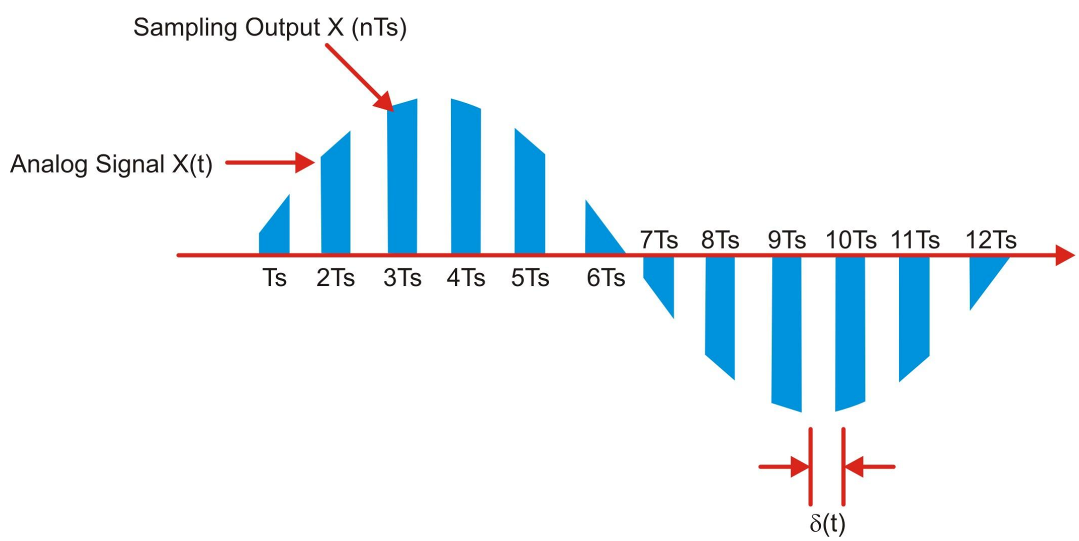
Figure 1



Sampling of signal at sampling interval (period) T_s

Figure 2

Figure 2 depicts the sampling of a signal at regular interval (period) $t=nT_s$ where n is an integer. The sampling signal is a regular sequence of narrow pulses δ (t) of amplitude 1.Figure 3 shows the sampled output of narrow pulses δ (t) at regular interval of time.



Sampled Output of narrow pulses δ (t)

Figure 3

The time distance T_s is called sampling interval or sampling period, $f_s=1/T_s$ is called as sampling frequency (Hz or samples/sec), also called sampling rate.

The Sampling Theorem:

The Sampling Theorem states that a signal can be exactly reproduced if it is sampled at a frequency F_s, where F_s is greater than twice the maximum frequency F_{max} in the signal.

$$F_s > 2 \cdot F_{max}$$

The frequency $2 \cdot F_{max}$ is called the Nyquist sampling rate. Half of this value, F_{max}, is sometimes called the Nyquist frequency.

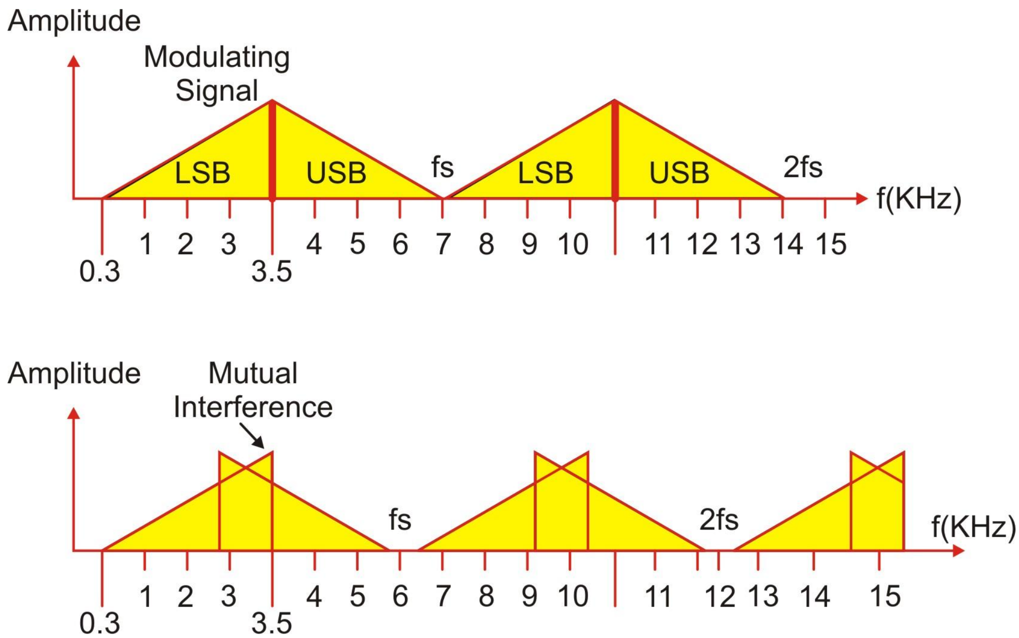
The sampling theorem is considered to have been articulated by Nyquist in 1928 and mathematically proven by Shannon in 1949. Some books use the term "Nyquist Sampling Theorem", and others use "Shannon Sampling Theorem". They are in fact the same sampling theorem.

The sampling theorem clearly states what the sampling rate should be for a given range of frequencies. In practice, however, the range of frequencies needed to faithfully record an analog signal is not always known beforehand. Nevertheless, engineers often can define the frequency range of interest. As a result, analog filters are sometimes used to remove frequency components outside the frequency range of interest before the signal is sampled.

For example, the human ear can detect sound across the frequency range of 20 Hz to 20 kHz. According to the sampling theorem, one should sample sound signals at least at 40 kHz in order for the reconstructed sound signal to be acceptable to the human ear. Components higher than 20 kHz cannot be detected, but they can still pollute the sampled signal through aliasing. Therefore, frequency components above 20 kHz are removed from the sound signal before sampling by a band-pass or low-pass analog filter.

Nyquist Criterion

As shown-in the figure 4 the lowest sampling frequency that can be used without the sidebands overlapping is twice the highest frequency component present in the information signal. If we reduce this sampling frequency even further, the sidebands and the information signal will overlap and we cannot recover the information signal simply by low pass filtering. This phenomenon is known as fold-over distortion or aliasing.



Nyquist Criterion (Sampling Theorem)

Figure 4

The Nyquist criteria states that a continuous signal band limited to F_m Hz can be completely represented by and reconstructed from the samples taken at a rate greater than or equal to $2F_m$ samples/second.

This minimum sampling frequency is called as Nyquist Rate i.e. for faithful reproduction of information signal $f_s > 2\ f_m$.

For audio signals the highest frequency component is 3.4 KHz.

So, *Sampling Frequency* ≥ 2 fm

$\geq 2 \times 3.4$ KHz

≥ 6.8 KHz

Practically, the sampling frequency is kept slightly more than the required rate. In telephony the standard sampling rate is 8 KHz. Sample quantifies the instantaneous value of the analog signal point at sampling point to obtain pulse amplitude output.

Nyquist's Uniform Sampling Theorem for Low pass Signal:

Part - I If a signal x(t) does not contain any frequency component beyond W Hz, then the signal is completely described by its instantaneous uniform samples with sampling interval (or period) of $T_s < 1/(2W)$ sec.

Part – II The signal x(t) can be accurately reconstructed (recovered) from the set of uniform instantaneous samples by passing the samples sequentially through an ideal (brick-wall) low pass filter with bandwidth B, where $W \le B < f_s - W$ and $f_s = 1/(T_s)$.

As the samples are generated at equal (same) interval (T_s) of time, the process of sampling is called uniform sampling. Uniform sampling, as compared to any non-uniform sampling, is more extensively used in time-invariant systems as the theory of uniform sampling (either instantaneous or otherwise) is well developed and the techniques are easier to implement in practical systems.

Sampling Techniques

There are three types of sampling techniques as under:

1. Ideal sampling or Instantaneous sampling or Impulse sampling
2. Natural sampling
3. Flat top sampling

1. Ideal sampling or Instantaneous sampling or Impulse sampling:

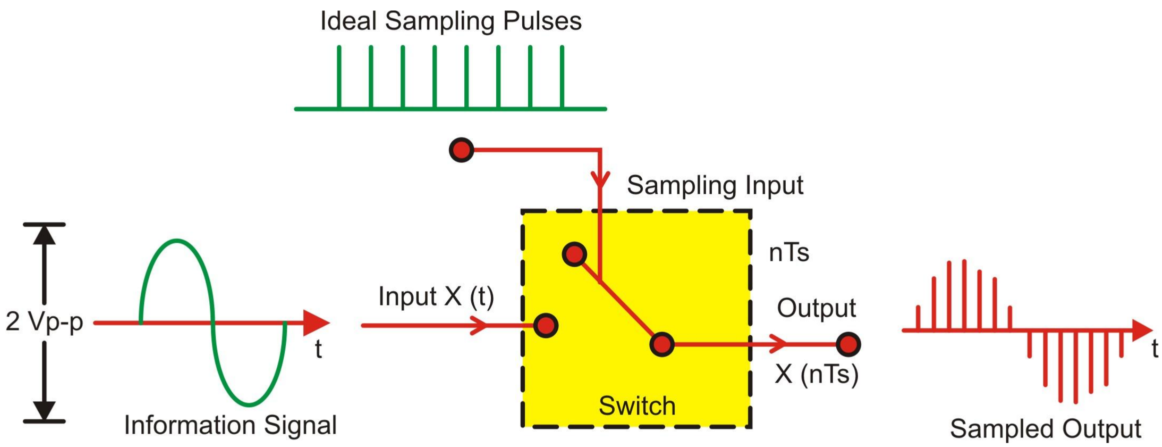
For the proof of sampling theorem we use ideal or impulse sampling.

The concept of 'instantaneous' sampling is more of a mathematical abstraction as no practical sampling device can actually generate truly instantaneous samples (a sampling pulse should have non-zero energy). However, this is not a deterrent in using the theory of instantaneous sampling, as a fairly close approximation of instantaneous sampling is sufficient for most practical systems. To contain our discussion on Nyquist's theorems, we will introduce some mathematical expressions.

If x(t) represents a continuous-time signal, the equivalent set of instantaneous uniform samples {x(nTs)} may be represented as:

$$\{x(nTs)\} = \Sigma\, x(t).\delta(t- nTs)$$

where $x(nTs) = x(t) =nTs$, $\delta(t)$ is a unit pulse singularity function and 'n' is an integer



Ideal sampling process

Figure 5

2. Natural sampling:

In the analogue-to-digital conversion process an analogue waveform is sampled to form a series of pulses whose amplitude is the amplitude of the sampled waveform at the time the sample was taken. In natural sampling the pulse amplitude takes the shape of the analogue waveform for the period of the sampling pulse as shown in figure 6.

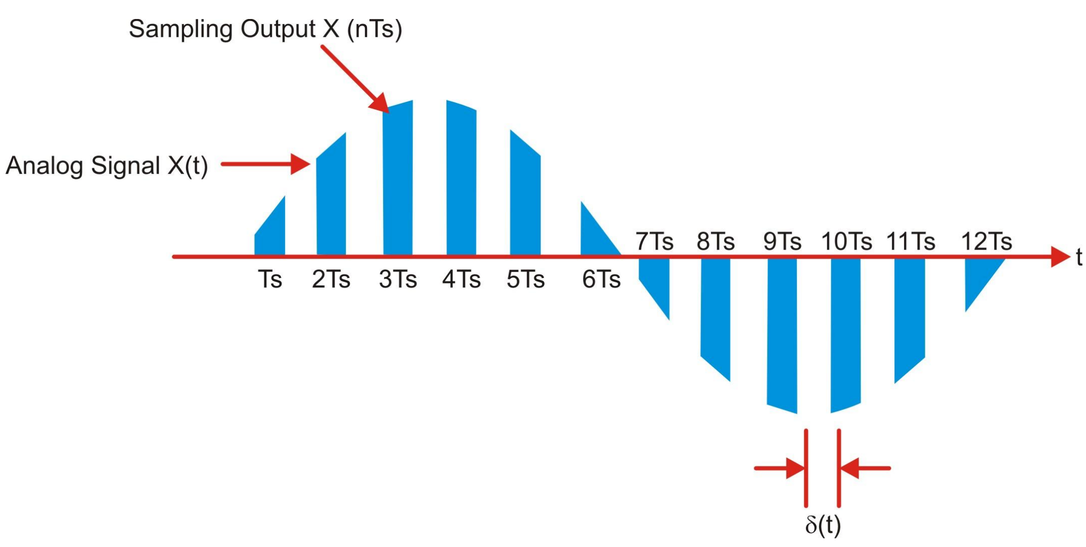


Figure 6

3. Flat Top sampling:

After an analogue waveform is sampled in the analogue-to-digital conversion process, the continuous analogue waveform is converted into a series of pulses whose amplitude is equal to the amplitude of the analogue signal at the start of the sampling process. Since the sampled pulses have uniform amplitude, the process is called flat top sampling as shown in figure 7.

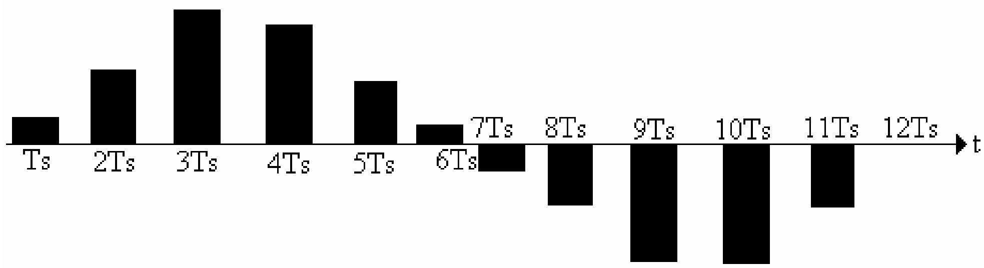


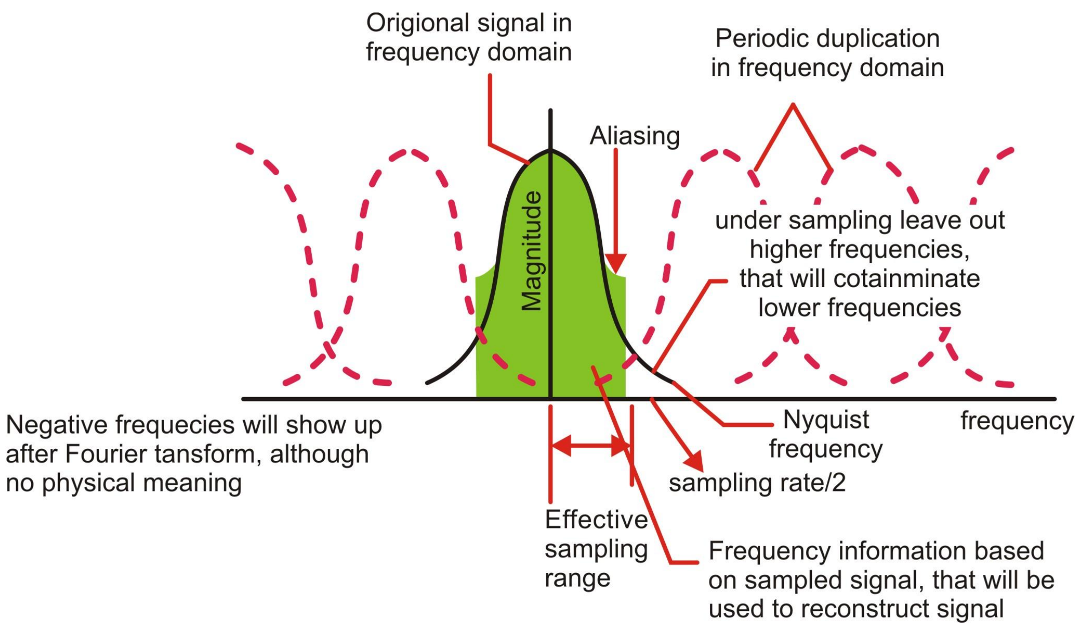
Figure 7

Note that due to the flat-top pulses, the spectrum of the sampled signal is distorted.
The narrower the pulse width, the less distortion.
The original signal may be obtained by using a low-pass filter with a characteristic which inverts the distortion.

Types of sampling

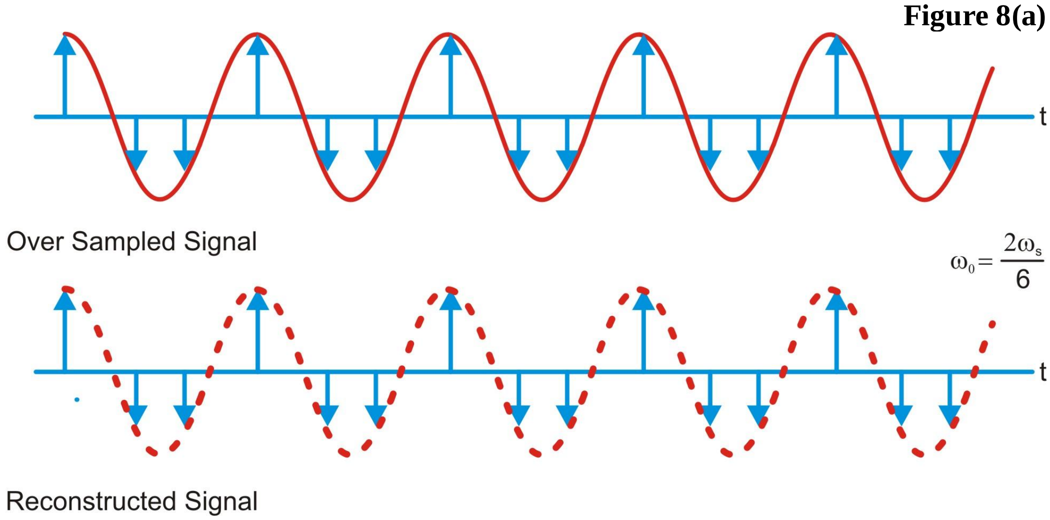
Over Sampling:

Graphically, if the sampling rate is sufficiently high, i.e., greater than the Nyquist rate, there will be no overlapped frequency components in the frequency domain. A "cleaner" signal can be obtained to reconstruct the original signal. This argument is shown graphically in the frequency-domain figure 8(a) and time-domain figure 8(b).



Over sampling in Frequency Domain

Figure 8(a)



Over sampling in Time Domain

Figure 8(b)

Under Sampling:

When the sampling rate is lower than or equal to the Nyquist rate, a condition defined as under sampling, it is impossible to rebuild the original signal according to the sampling theorem.

An example is illustrated below, where the reconstructed signal built from data sampled at the Nyquist rate is way off from the original signal. This argument is shown graphically in the frequency-domain figure 9(a) and time-domain figure 9(b).

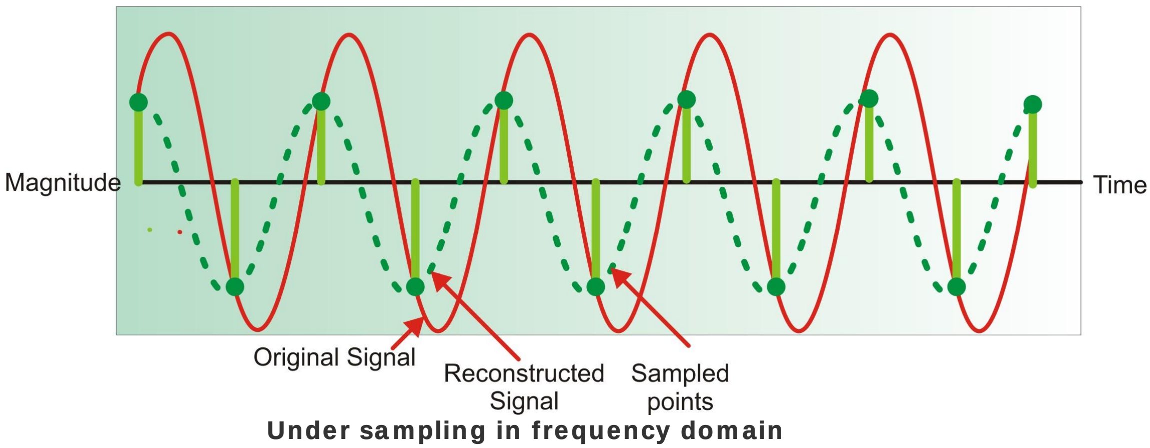


Figure 9(a)

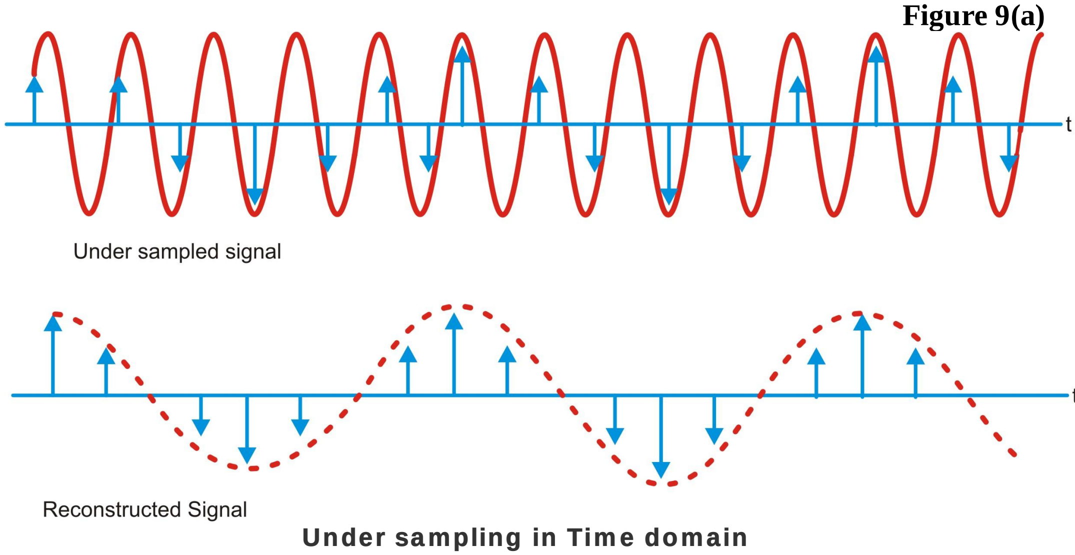


Figure 9(b)

In practice, the continuous signal is sampled using an analog or digital converter (ADC), a non-ideal device with various physical limitations. This result in deviations from the theoretically perfect reconstruction capabilities collectively referred to as distortion.

Effect of Duty Cycle on Information Recovery:

The duty cycle of a signal is defined as, the ratio of pulse duration to the pulse repetition period. This ratio can also be expressed as percentage. e.g. the square wave has equal pulse and no pulse duration; hence its duty cycle is 0.5 or 50%

The duty cycle of the sampling pulses is an important parameter in Pulse Amplitude Modulation system. They govern the following important aspects.

a. The narrower pulses allow us to time division multiplex many such pulse amplitude modulation panels i.e. we can send any no. of pulse amplitude modulated signals over same channel at a time. Hence lower duty cycle beneficial in this respect.

b. The narrower pulses have wider frequency spectrum. Hence the wider bandwidth channel is required.

c. Narrower pulses have less power as the power content of a pulse depends on its amplitude and width. During transmission and demodulation the inherent noise can play a major havoc on the low power signal. Hence a pulse of larger duty-cycle is desirous for this sake.

In practice an engineering compromise is made between narrower and broader pulse width taking into account the efficiency, requirement and inherent noise of the system.

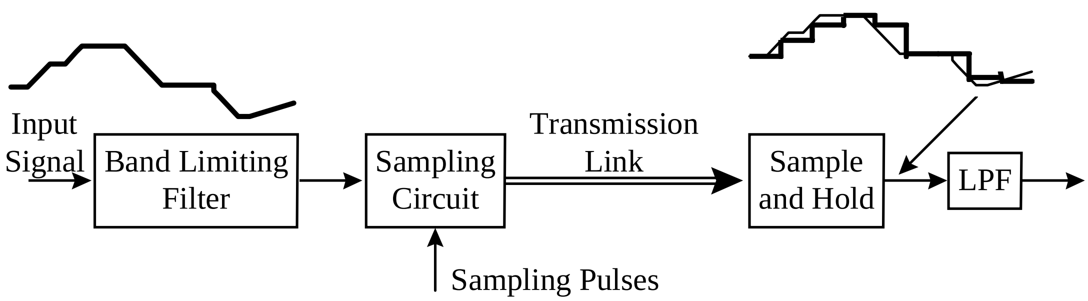
Note: The frequency spectrum of pulse amplitude modulation signal does not contain those harmonics which when multiplied by duty cycle results in an integer. e.g. the square wave with duty cycle 0.5 (50%) does not contain even harmonics as they result in an integer when multiplied with duty cycle. Thus a square wave-sampling signal only contains odd harmonics.

Pulse Amplitude Modulation

Most digital modulation systems are based on pulse modulation. It involves variation of a pulse parameter in accordance with the instantaneous value of the information signal. This parameter can be amplitude, width, repetitive frequency etc.

Depending upon the nature of parameter varied, various modulation systems are used. Pulse amplitude modulation, pulse width modulation, pulse code modulation are few modulation systems cropping up from the pulse modulation technique. In pulse amplitude modulation (PAM) the amplitude of the pulses are varied in accordance with the modulating signal.

In true sense, pulse amplitude modulation is analog in nature but it forms the basis of most digital communication and modulation systems. The pulse modulation systems require analog information to be sampled at predetermined intervals of time. Sampling is a process of taking the instantaneous value of the analog information at a predetermined time interval.
A sampled signal consists of a train of pulses, where each pulse corresponds to the amplitude of the signal at the corresponding sampling time. The signal sent to line is modulated in amplitude and hence the name **Pulse Amplitude Modulation** (PAM).



Block diagram of Pulse Amplitude Modulation

Figure 10

Pulse amplitude modulation, the simplest form of pulse modulation, is illustrated in Figure 11. It forms an excellent introduction to pulse modulation in general. Pulse amplitude modulation is a pulse 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. As shown in Figure 11. The two types are double polarity pulse amplitude modulation, which is self-explanatory and single polarity pulse amplitude modulation, in which a fixed DC level is added to the signal, to ensure that the pulses are always positive. As will be seen shortly, the ability to use constant-amplitude pulses is a major advantage of pulse modulation, and since Pulse Amplitude Modulation does not utilize constant amplitude pulses, it is infrequently used. When it is used, the pulses frequency modulates the carrier. It is very easy to generate and demodulate pulse amplitude modulation. In a generator, the signal to be converted to Pulse Amplitude Modulation is fed to one input of an AND gate. Pulses at the sampling frequency are applied to the other input of the AND gate to open it during the wanted time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant. The pulses are then passed through a pulse shaping network, which gives them flat tops.

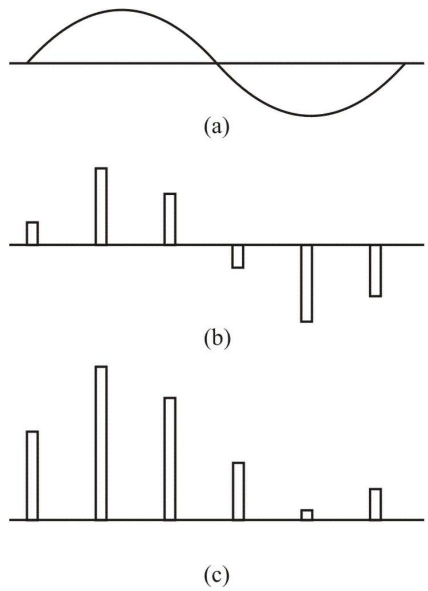


Figure 11

Pulse Modulation System

1. **Pulse Amplitude Modulation (PAM):**

   In pulse amplitude modulation system the amplitude of the pulse is varied in accordance with the instantaneous level of the modulating signal. Now days, the PAM system is not generally used, but it forms the first stage of the other types of pulse modulation.

2. **Pulse Width Modulation (PWM):**

   In PWM system the width of the pulse is varied in accordance with the instantaneous level of the modulating signal.

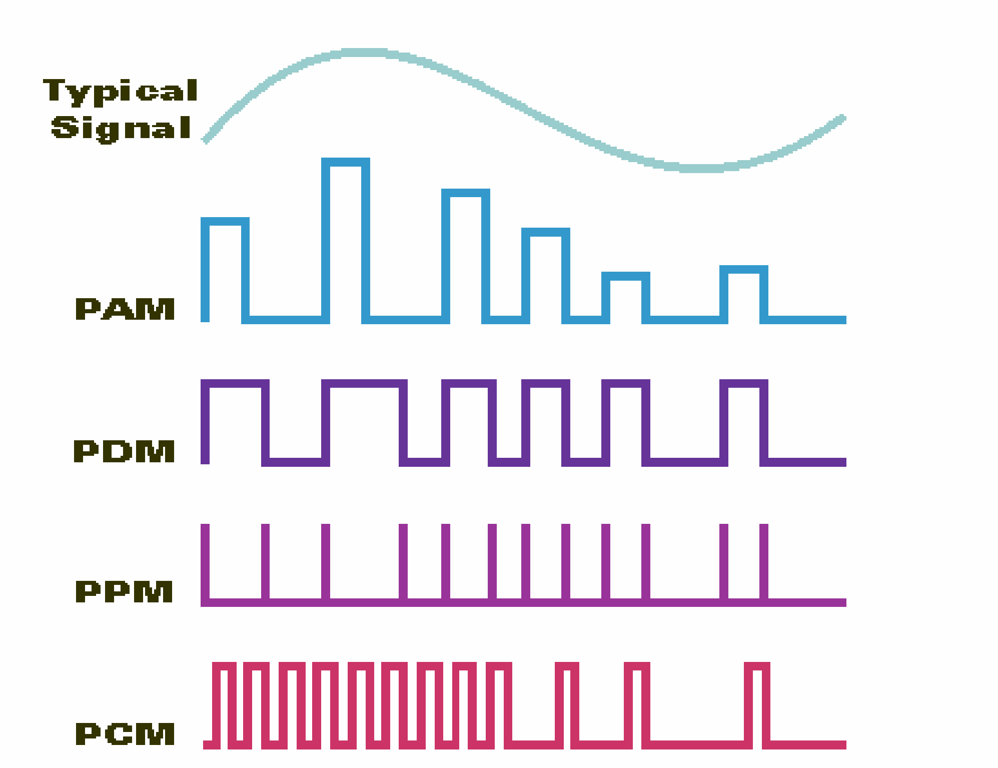
3. **Pulse Position Modulation (PPM):**

   In PPM System, the position of the pulse relative to the zero reference level is varied in accordance with the instantaneous level of the modulating signal.

4. **Pulse Code Modulation (PCM):**

   In PCM System the amplitude of the sampled waveform at definite time intervals is represented as a binary code. The first three techniques of the above described systems are not truly digital but in fact are analog in nature. The very fact that the variation of a particular pulse parameter is continuous rather than being in the discrete steps makes the system analog in nature.

   The Waveforms related to pulse modulation is shown in figure 12.

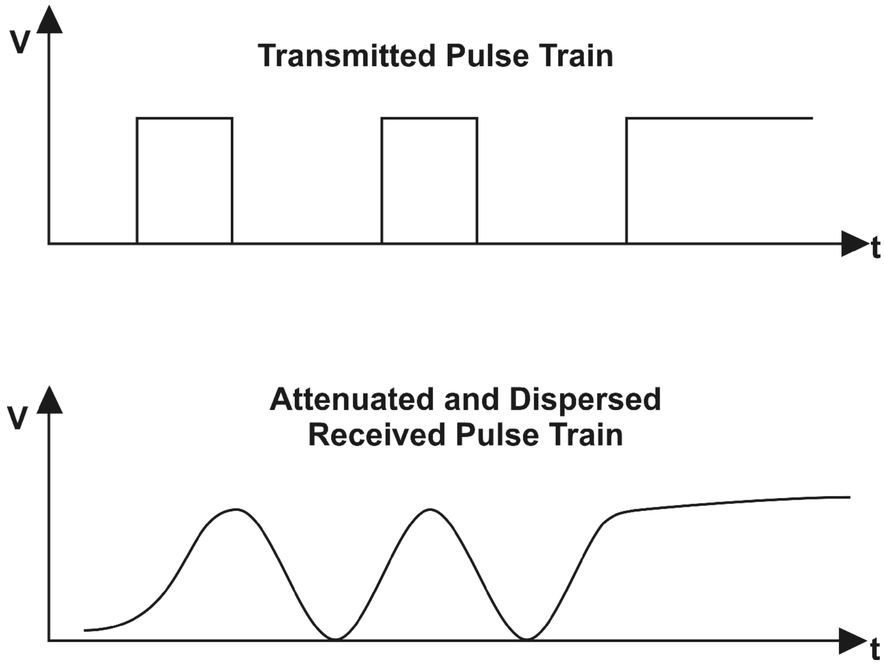


Pulse Modulation Techniques

Figure 12

As a result of this, the PAM signals are vulnerable to noise & dispersion of the pulse. The channel introduces noise on the signal from various sources. Also the receiver is not noise free.

The pulses also suffer attenuation & dispersion as they pass through the channel. The primary line constants (L, C, G, & R) limit the velocity at which a particular frequency can travel. The result is different frequency travel at different velocities in the medium. Therefore some frequency component of the square wave arrives later as compared to the other. This causes widening of the pulse width. The phenomenon is called 'dispersion'. The combined effect of attenuation, dispersion & noise is so large that the pulse is impaired & introduced at the receiver as shown in figure 18.



Pulse Train distortion due to Channel Characteristics

Figure 13

Sample & Hold circuit

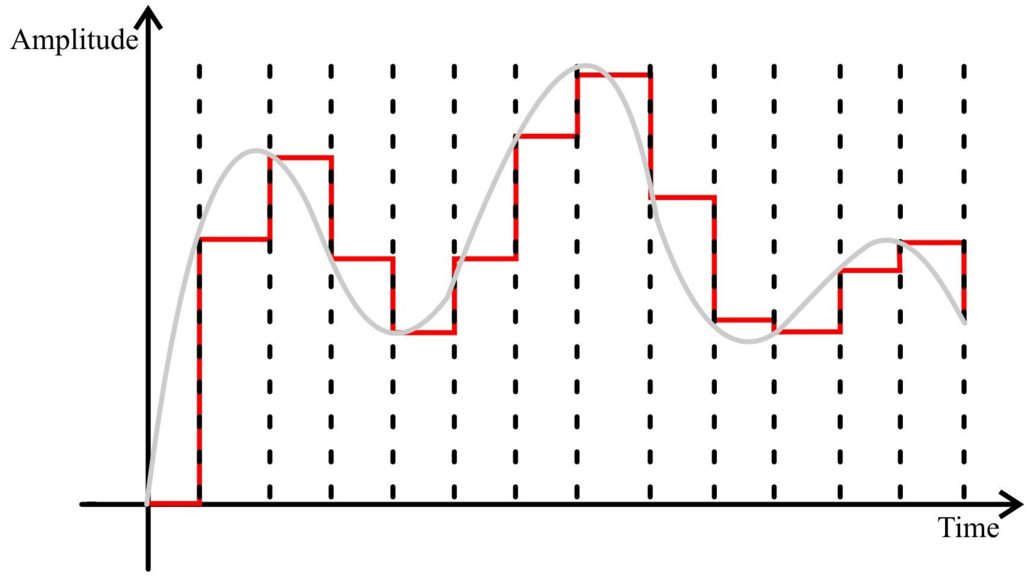
In electronics, a sample and hold circuit is used to interface real-world signals, by changing analogue signals to a subsequent system. The purpose of this circuit is to hold the analogue value steady for a short time while the converter or other following system performs some operation that takes a little time.

Sampling mode:

In this mode, the switch is in the closed position and the capacitor charges to the instantaneous input voltage.

Hold mode:

In this mode, the switch is in the open position. The capacitor is now disconnected from the input. As there is no path for the capacitor to discharge, it will hold the voltage on it just before opening the switch. The capacitor will hold this voltage till the next sampling instant.



Sample and Hold Waveform

Figure 14

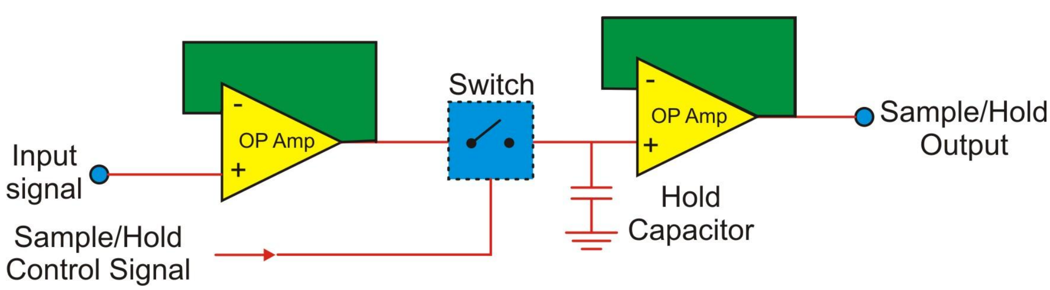
Now, from figure 14 the area under the curve (which is equivalent to the signal power) is greater and so the filter output amplitude and quality of reproduced signal is improved.

In most circuits, a capacitor is used to store the analogue voltage and an electronic switch or gate is used to alternately connect and disconnect the capacitor from the analogue input. The rate at which this switch is operated is the sampling rate of the system.

In a sample and hold circuit the switch opens for a very short duration. The sample and hold circuit integrates for a short duration charge into a capacitor.

The 'hold' facility can be provided by a capacitor, when the switch connects the capacitor to PAM output it charges to the instantaneous value.

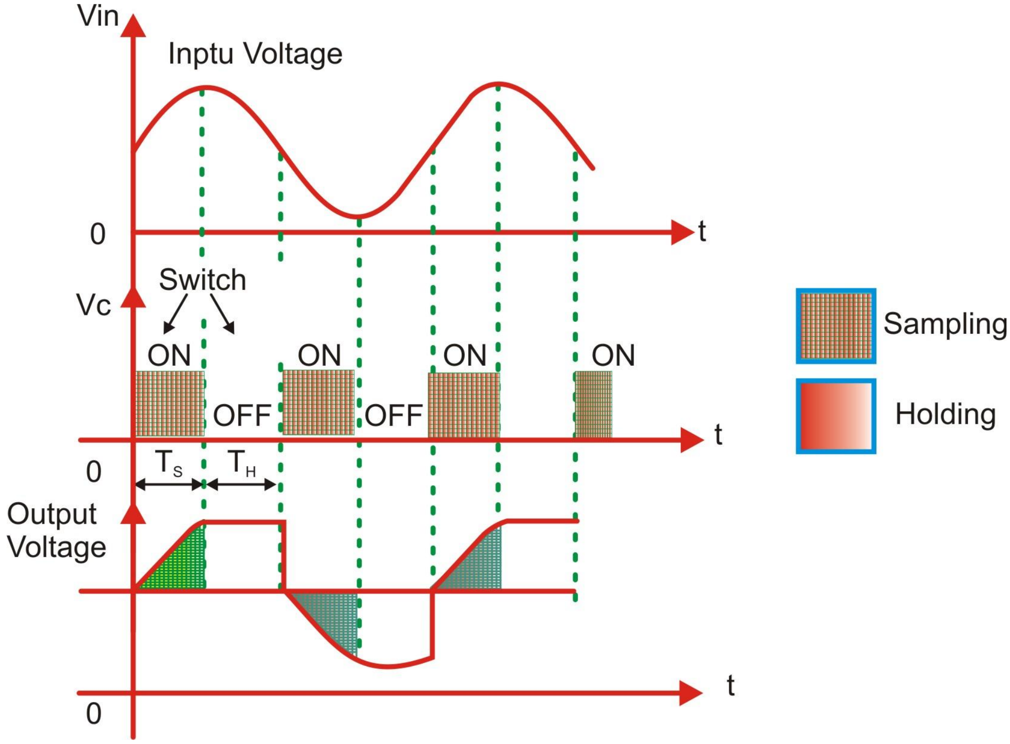
A buffered sample and hold circuit consists of unit gain buffer preceding and succeeding the charging capacitor. The high input impedance of the preceding buffer prevents the loading of the message source and also ensures that the capacitor charges by a constant rate irrespective of the source impedance see figure 15(a).



Sample Hold Circuit

Figure 15(a)

The high input impedance of the succeeding buffer prevents the charging from the capacitor due to loading and hence the capacitor can hold the charge for infinite time, at least theoretically. However, small leakage current through the capacitor dielectric into '+'ve input of second buffer is always present which causes gradual charge loss. The rate of change of voltage with respect to time dv / dt is called as droop rate and is important parameter in sample and Hold circuit design. The sample and hold waveform is shown in figure 15(b).



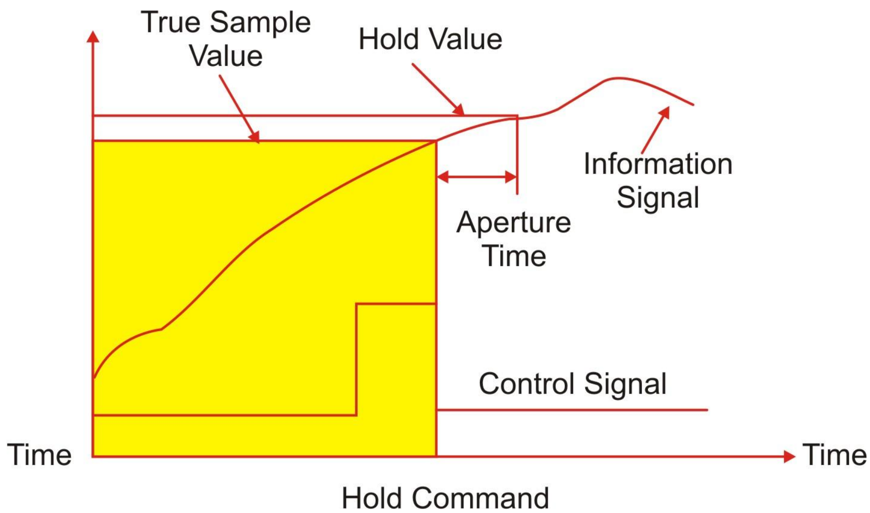
Sample and hold wave form

Figure 15(b)

Important Parameters of Sample & Hold Circuit

1. Aperture time:

The aperture time is defined as the delay time between the beginnings of the hold command to the time the capacitor voltage ceases to follow the information signal. Hence the hold value is different from the true sample value. The aperture time cannot be reducing to zero because on application of finite time taken by a switch to close & open on application of the hold signal. Therefore a small value of aperture time is sought after.



Timing Diagram for Sample and Hold Circuit

Figure 16

2. Acquisition Time:

In sample mode, it takes finite time for the capacitor to charge to the information signal value depending on the RC time constant. This is called as the acquisition time. The acquisition time is dependent on the current flowing from the input buffer through switch and hence on RC time constant. The maximum acquisition time occurs when the capacitor voltage has to change by the full amplitude of the information signal.

3. Droop Rate:

As it has been discussed earlier, the presence of leakage current through capacitor dielectric to +ve input of succeeding buffer causes charge loss of capacitor. Hence the voltage level at the output falls with in time. This rate of change of voltage with respect to time dv/dt is known as droop rate. Over value of droop rate is desirable as the circuit should be able to maintain the sample at a relatively constant level until the next sample.

4. Feed Through:

At high frequencies, the stray capacitance within the switch causes some of the input signal to appear at the output during the hold state (switch open). The fraction of input signal appearing at the output of sample and hold circuit is called feed through.

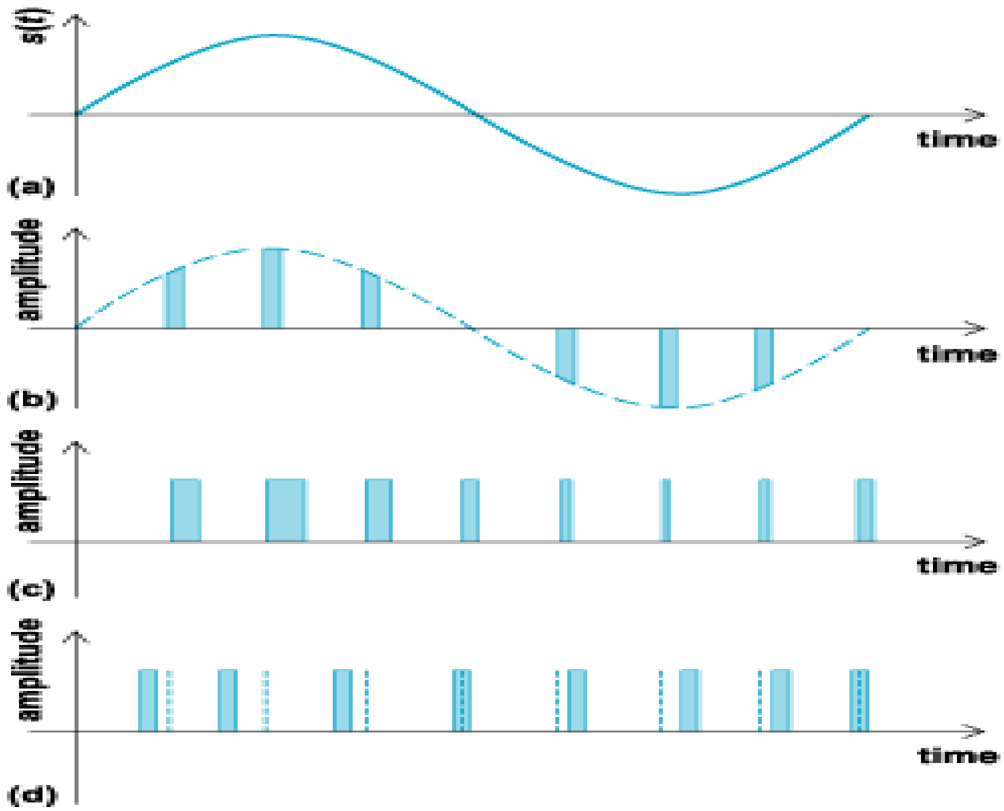
The sample and hold feature provides both problem and benefit will be seen afterwards.
The sample and hold feature problem and benefit as will be seen afterwards.

Pulse Position Modulation

The Amplitude and width of the pulses is kept constant in this system, while the position of each pulse, in relation to the position of a recurrent reference pulse is varied by each instantaneous sampled value of the modulating wave. As mentioned in connection with pulse width modulation, pulse-position modulations has the advantage of requiring constant transmitter power output, but the disadvantages of depending on transmitter receiver is synchronization.

Pulse Width Modulation

In pulse width modulation of pulse amplitude modulation is also often called PDM (pulse duration modulation) and less often, PLM (pulse length modulation). In this system, as shown in Figure 17, we have fixed amplitude and starting time of each pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.



(a) Analog signal, $s(t)$. (b) Pulse-amplitude modulation. (c) Pulse-width modulation. (d) Pulse-position modulation. **Figure 17**

In Figure 17, there may be a sequence of signal sample amplitudes of (say) 0.9, 0.5, 0 and -0.4V. These can be represented by pulse widths of 1.9, 1.5, 1.0 and 0.6µs respectively. The width corresponding to zero amplitude was chosen in this system to be 1.0µs, and it has been assumed that signal amplitude at this point will vary between the limits of + 1 V (width = 2µs) and -1 V (width = 0µs). Zero amplitude is thus the average signal level, and the average pulse width of 1µs has been made to correspond to it. In this context, a negative pulse width is not possible. It would make the pulse end before it began, as it were, and thus throw out the timing in the receiver. If the pulses in a practical system have a recurrence rate of 8000 pulses per second, the time between the commencements of adjoining pulses is 10^6 /8000 = 125µs. This is adequate not only to accommodate the varying widths but also to permit time-division multiplexing. Pulse width modulation has the disadvantage, when compared with pulse position modulation, which will be treated next, that its pulses are of varying width and therefore, of varying power content. This means that the transmitter must be powerful enough to handle the maximum-width pulses, although the average power transmitted is perhaps only half of the peak power. On the other hand, pulse width modulation still works if synchronization between transmitter and receiver fails, whereas pulse-position modulation does not, as will be seen.

Low Pass Filter

The pulse amplitude modulation,-pulse width modulation & pulse position modulation system the message is recovered by a low pass filter. The type of filter used is very important, as the signal above the cut-off frequency would affect the recovered signal if they were not attenuated sufficiently.

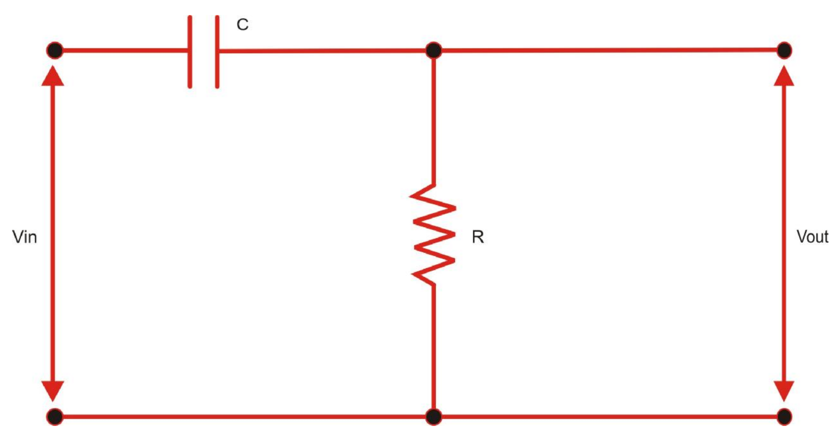
Filter Basic:

The simplest type of filter is a resistance- capacitance (RC) filter. The high pass filter and low pass RC filters are as shown in Figure 18.

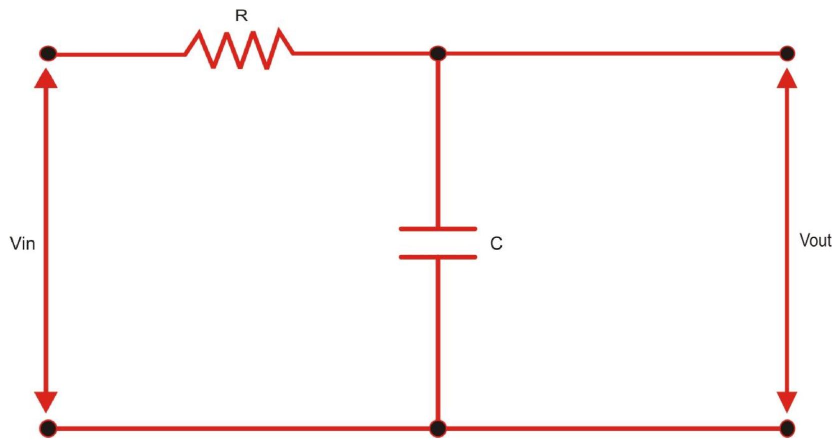
The analysis of these filters becomes easier if we think of them as AC potential dividers. The reactance of the capacitor is frequency dependent with a high value at low frequencies and a low value at high frequencies.

In case of high pass filter, the series capacitance has high reactance at low frequencies and hence results in reduction in output voltage. An increase in frequency causes an increase in output voltage with V_{out} approaching input voltage V_{in}.

The effect of capacitors is just opposite case of low pass filter. Here, the capacitance is in short and hence V_{out} reduces as frequency increases there by decreasing its reactance. The ratio of V_{out} / V_{in} is known as transfer function for the circuit. For RC low pass filter, the transfer function can be derived by using potential divider resistance.



Passive High Pass Filter



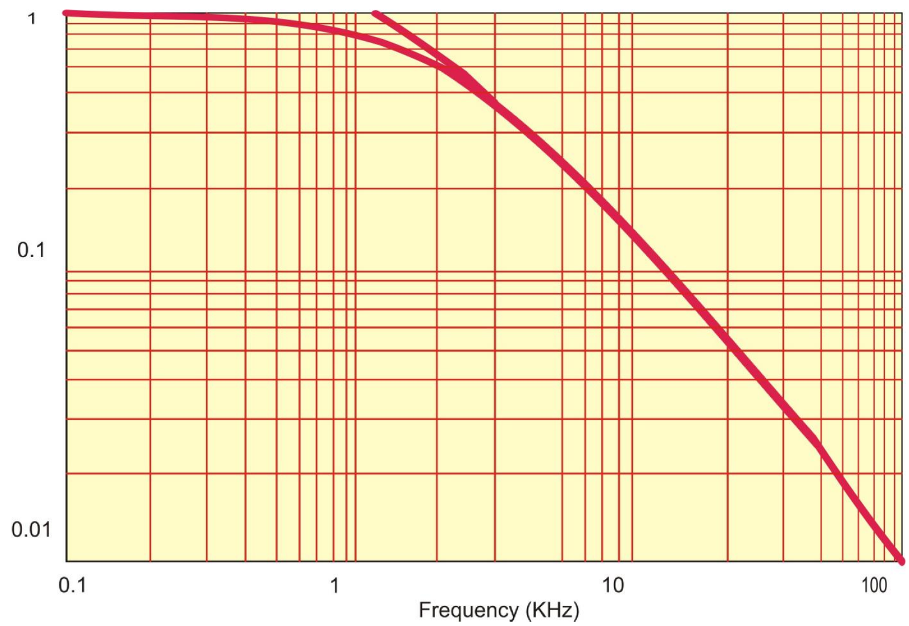
Passive Low Pass Filter

Figure 18

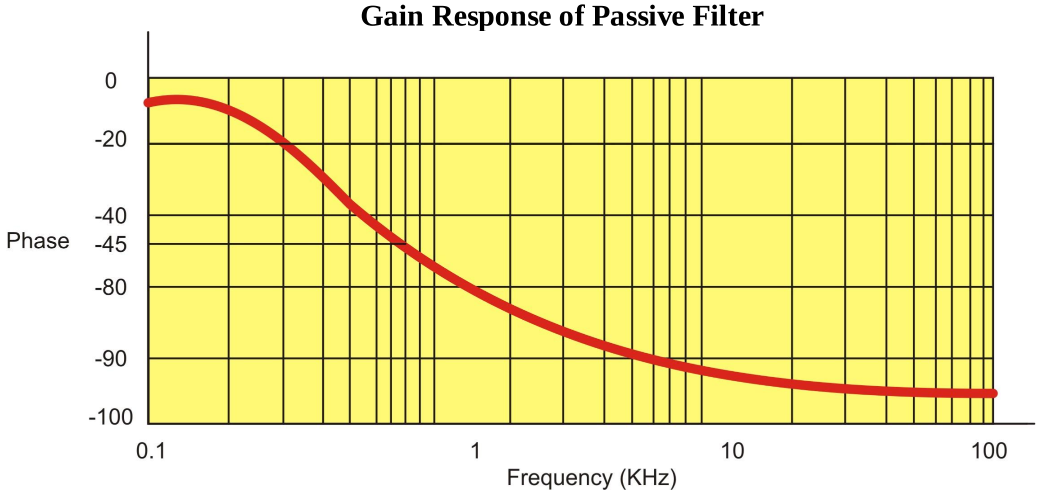
$$\frac{\text{Vout}}{\text{Vin}} = \frac{1}{\sqrt{1+\omega^2 R^2 C^2}}$$

$$\text{If } \omega = \frac{1}{RC} \text{ Then } \frac{\text{Vout}}{\text{Vin}} = \frac{1}{\sqrt{2}} = 0.707 = -3\text{dB}$$

So, This is the half-power point of the filter i.e. at frequency ω=RC, the output power decreases to half of the input power. This is also known as the cutoff frequency (Fc). The filter not only causes amplitude but a change in phase is also experienced. A typical response of a low pass filter is as shown Figure 19.



Gain Response of Passive Filter

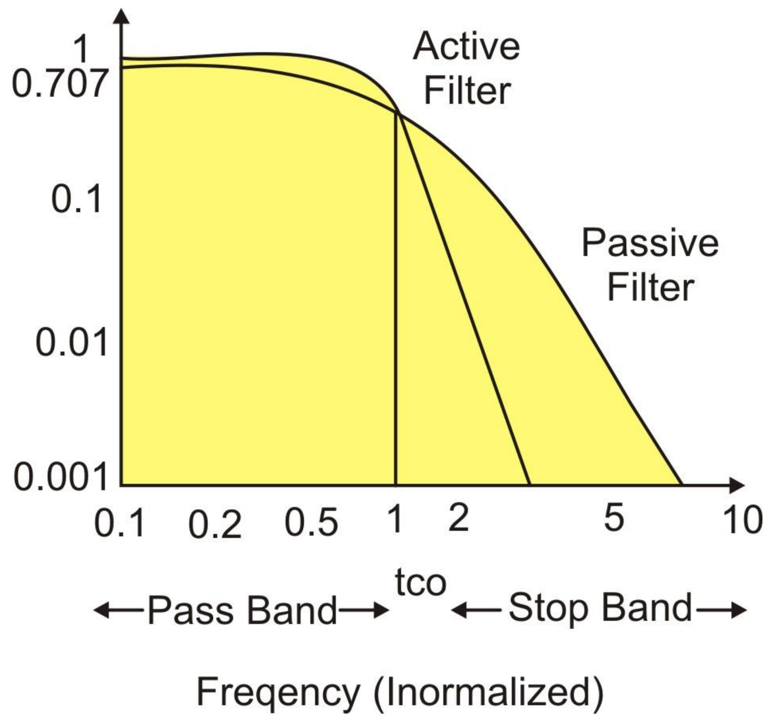


Phase Response of Passive Filter

Figure 19

The RC Filter is a passive filter and does not give a steeper fall-off. Cascading many such RC Filters give a steeper fall-off but a price of successive attenuation of the signal.

Active filters give much flatter response in the pass band and they also have a steeper cut-off gradient. The following Figure 20 shows a comparison between two types of the signal.



Amplitude Response of Active & Passive Filters

Figure 20

Active filters give flat response in the pass band and they also have a steeper cut-of gradient. The following Figure shows a comparison between two types of filter responses.

The other advantages offered by Active filters are:

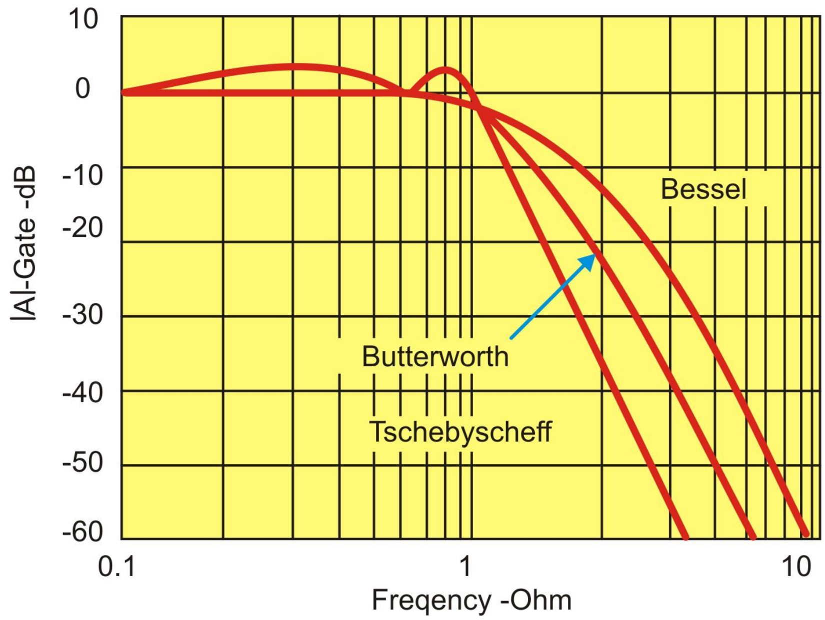
1. Gain frequency adjustment flexibility (i.e. easy tuning)

2. No loading problem between source, load or successive stages

3. They are economical than passive filters

The active filters employ transistors or op-amps in addition to resistor and capacitor. The resistors at the output of the op-amp create a non-inverting voltage amplifier of voltage gain K while other resistor and capacitor sets the frequency response properties of the filter.

An ideal filter should have zero loss in pass band and infinite loss in stop band. In practice no ideal response exists, but there are many responses which approximate the ideal response namely, Butterworth, Chebysehv, Bessel etc. the comparison of these filter responses are shown in Figure 21.



Comparison of Filter Responses

Figure 21

The voltage controlled voltage source (VCVS) can be arranged in the following manner to get the Butter worth response. The same arrangement with different component values gives a different response function / characteristic. See Figure 22.

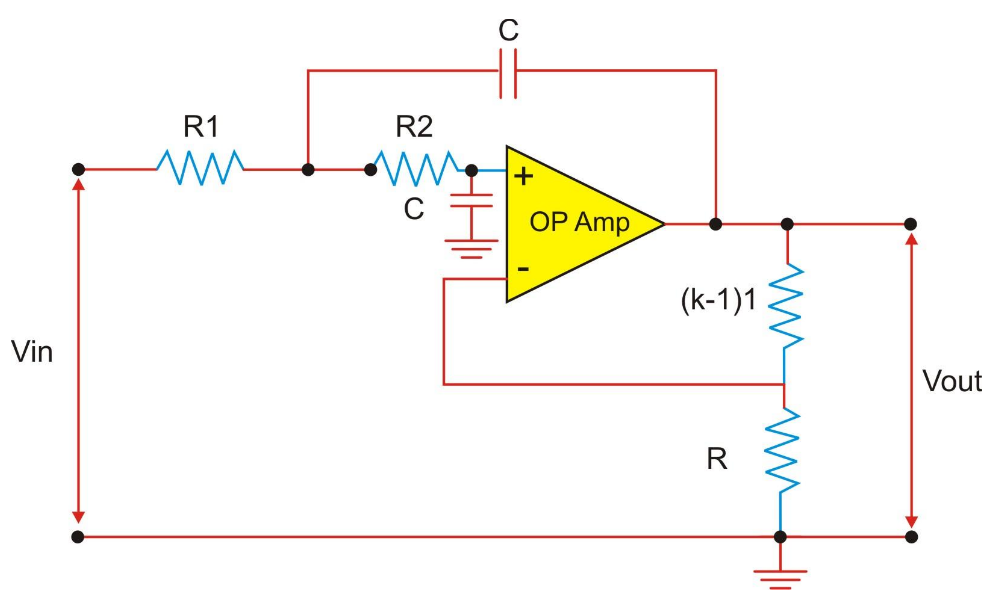


Figure 22

The nth order filter has a rate of fall off 6dB/Octave or 20dB/decade and one capacitor or inductor is required for each pulse (order)

The following table summarizes the effect of fall-off gradient on a signal such as square wave. See Figure 23

Filter Order	**Fall-off/ dB Octave**	**fall-off/ dB decade**	**Phase at cut-off frequency**
First	6	20	-45
Second	12	40	-90
Fourth	14	80	-180

The amplitude response of a Butterworth filter is given by;

$$\frac{Vout}{Vin} = \frac{1}{\sqrt{1+(f/fc)^{2n}}}$$

Where n is the order of the filter.

Frequency Response of a second order Butterworth low pass filter

The arrangement shown in Figure 24 can be used as second order Butterworth filter with cut-off frequency

$$Fc = \frac{1}{2pRC}$$

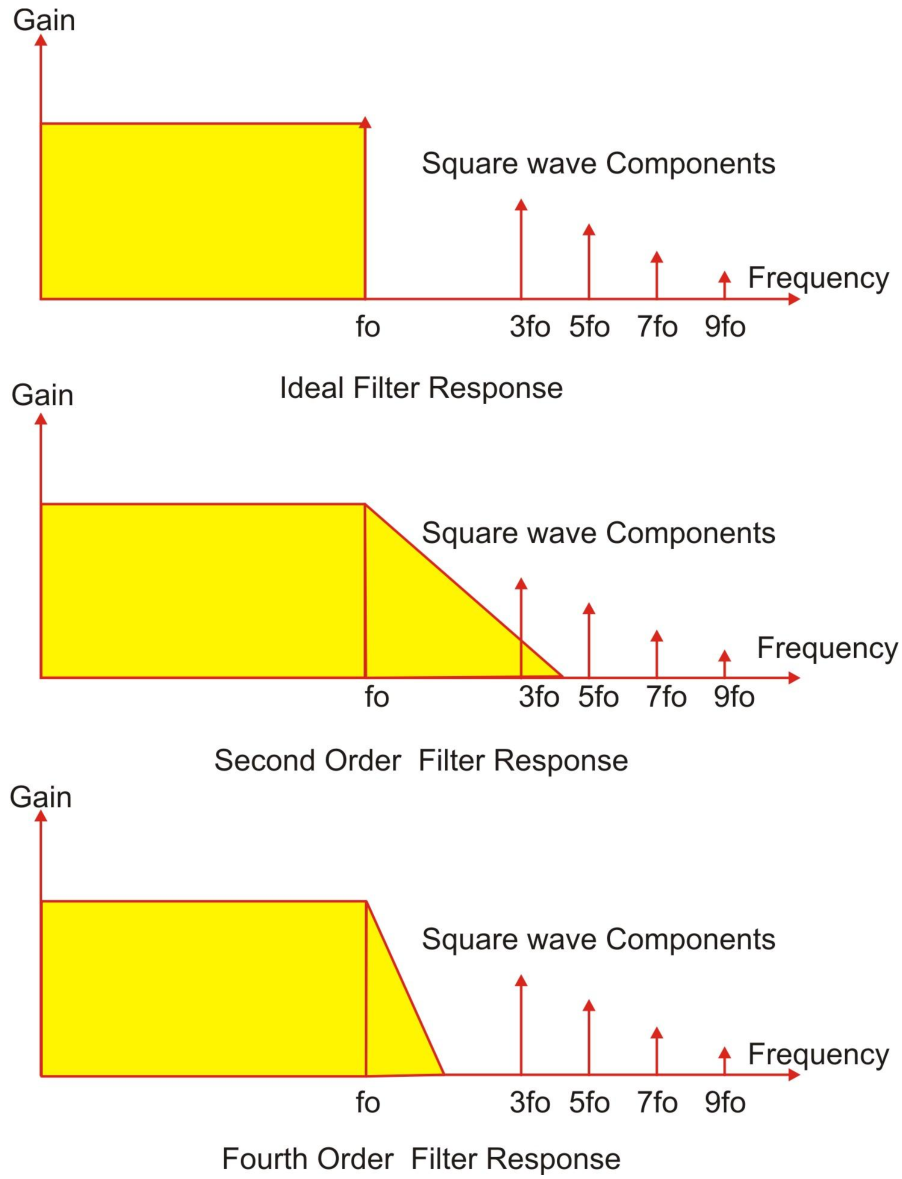
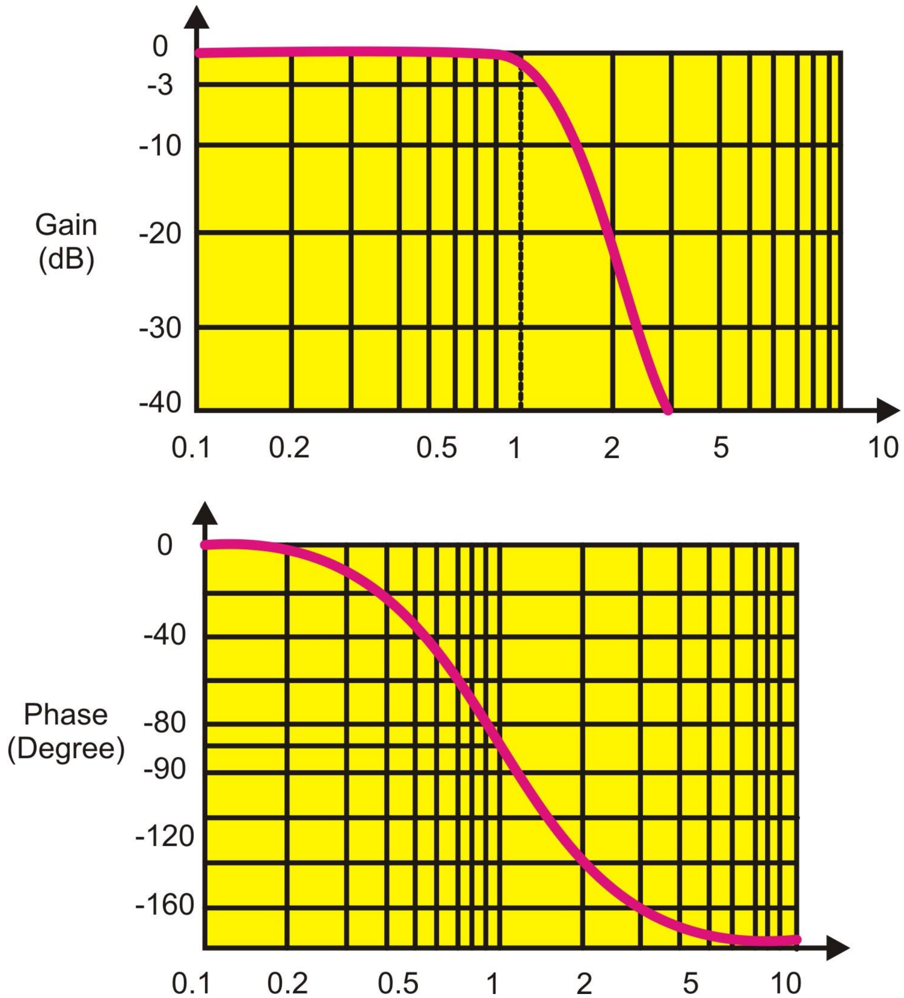


Figure 24

The amplitude and phase responses of second order Butterworth low pass filter with respect to frequency is shown in Figure 25.



Frequency (Normalized)

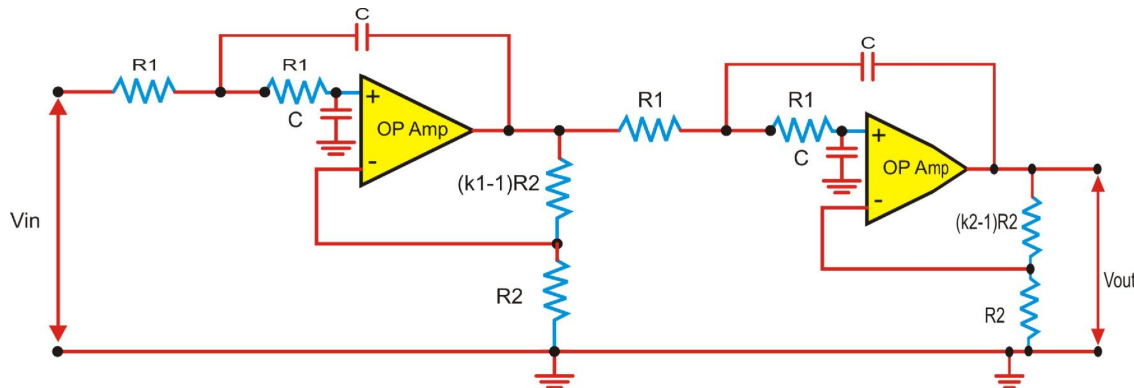
Amplitude Vs Frequency and Phase Vs Frequency
Response of Second Order Butterworth Low Pass Filter

Figure 25

For this circuit the voltage gain has been set equal to 1.586.

Fourth Order Butterworth Low Pass Filter:

The fourth order Butterworth filter can be formed by cascading two-second order Butterworth filters. As can be seen from Figure 26 the components R and C are identical in both filter stages and they determine the cut -off frequency. In our circuit the gain of first stage has been set to 1.15 and that of other is set at 2.235.



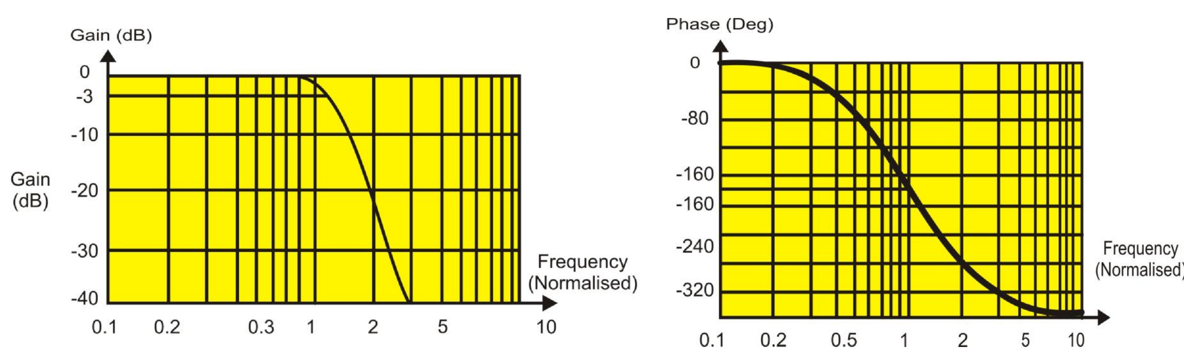
Fourth Order Butterworth Low Pass Filter

Figure 26

The amplitude/frequency and phase / frequency responses of fourth order Butterworth low pass filters are shown in Figure 27.

The filter design should be done critically so that any unwanted frequency component existing close to the desired frequency component attenuate sufficiently to save the output from getting corrupted. Through increasing order filter is desirable, there is a price that we have to pay for steeper fall-off.

1. Additional circuitry increases complexity & cost.
2. Increase in order increases phase lag, though it is not so critical in audio circuits.



(a) **(b)**

(a) Gain Response of Fourth Order Butter Worth Low Pass Filter
(b) Phase Response of Fourth Order Butter Worth Low Pass Filter

Figure 27

Recommended testing instruments needed for experimentation

1. Dual Trace Oscilloscope.
2. Oscilloscope probes X1 – X10 etc.

Experiment 1

Objective: Study of Pulse Amplitude Modulation using Natural & Flat top Sampling.

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

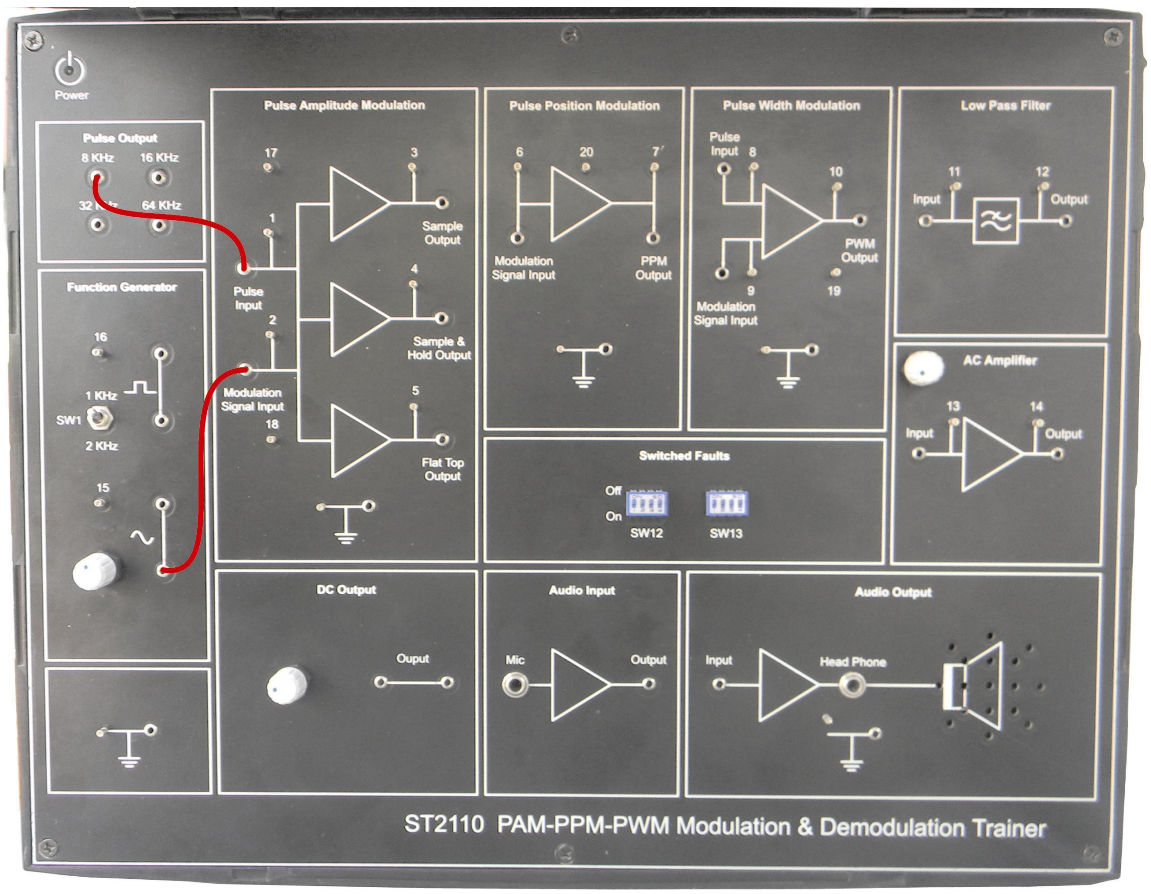


Figure 1.1

Procedure:

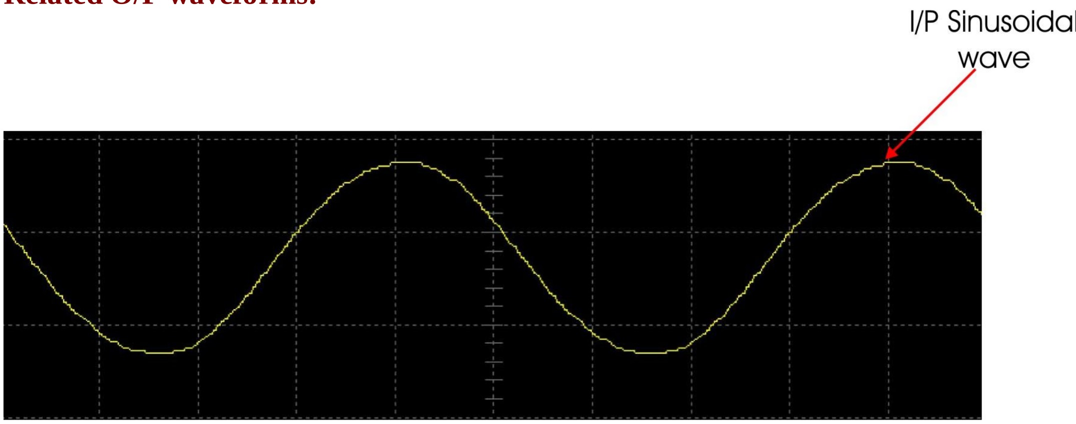
1. Connect the circuit as shown in Figure 1.1.

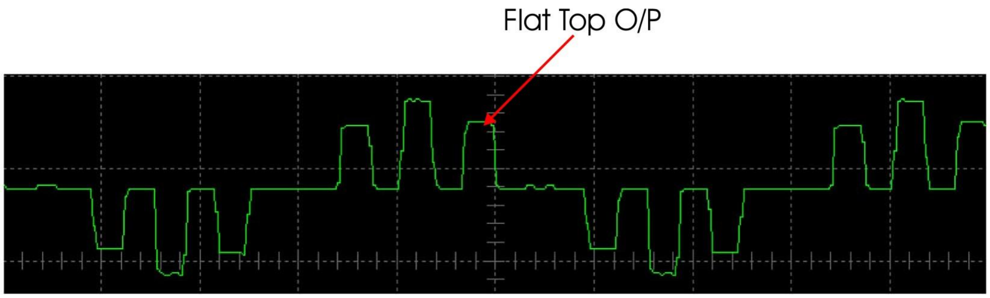
    a. Output of sine wave to modulation signal input in PAM block keeping the switch in 1 KHz position.

    b. 8 KHz pulse output to pulse input.

2. Switch 'On' the power supply & oscilloscope.
3. Observe the outputs at TP(3 & 5) these are natural & flat top outputs respectively.
4. Observe the difference between the two outputs.
5. Vary the amplitude potentiometer and frequency change over switch & observe the effect on the two outputs.
6. Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.
7. Switch 'On' fault No. 1, 2, 3, 4 one by one & observe their effect on Pulse Amplitude Modulation output and try to locate them.
8. Switch 'Off' the power supply.

Related O/P waveforms:





Questions:

1. Define PAM in brief?
2. What are the advantages of using PAM signal?
3. Give the classification of sampling?
4. Why flat top sampling is better than natural sampling?
5. What is the significance of sampling?

Experiment 2

Objective: Study of PAM using Sample & Hold circuit

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

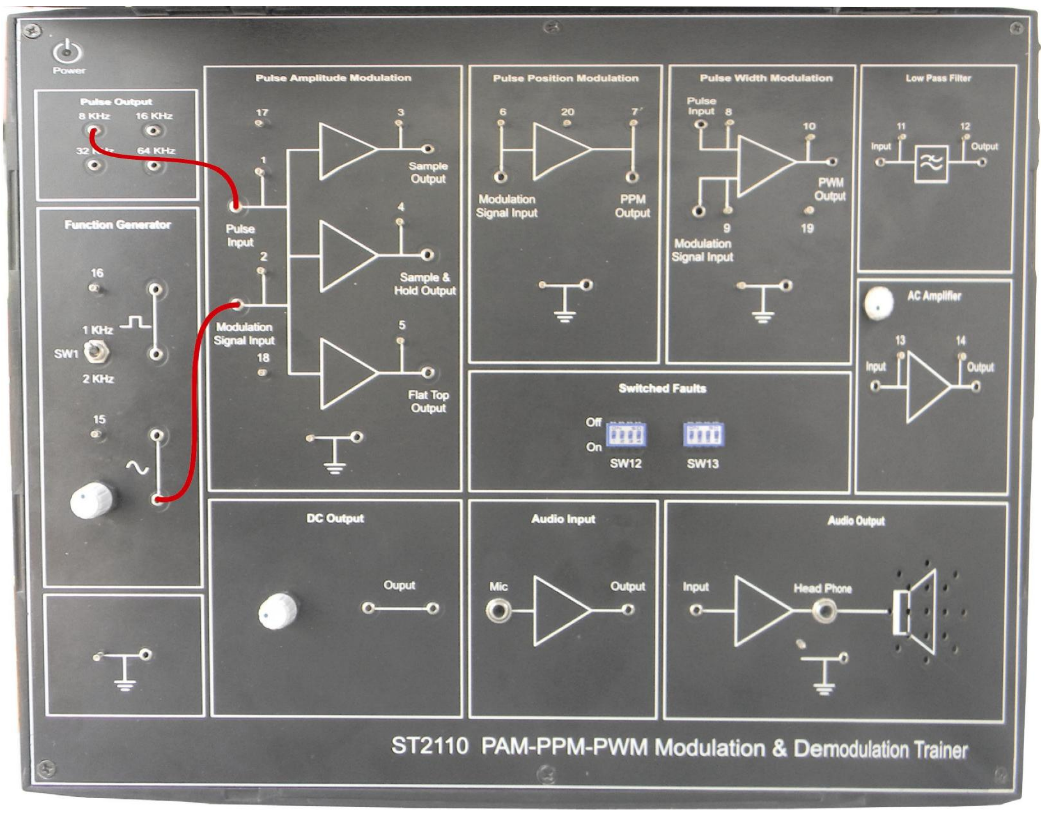
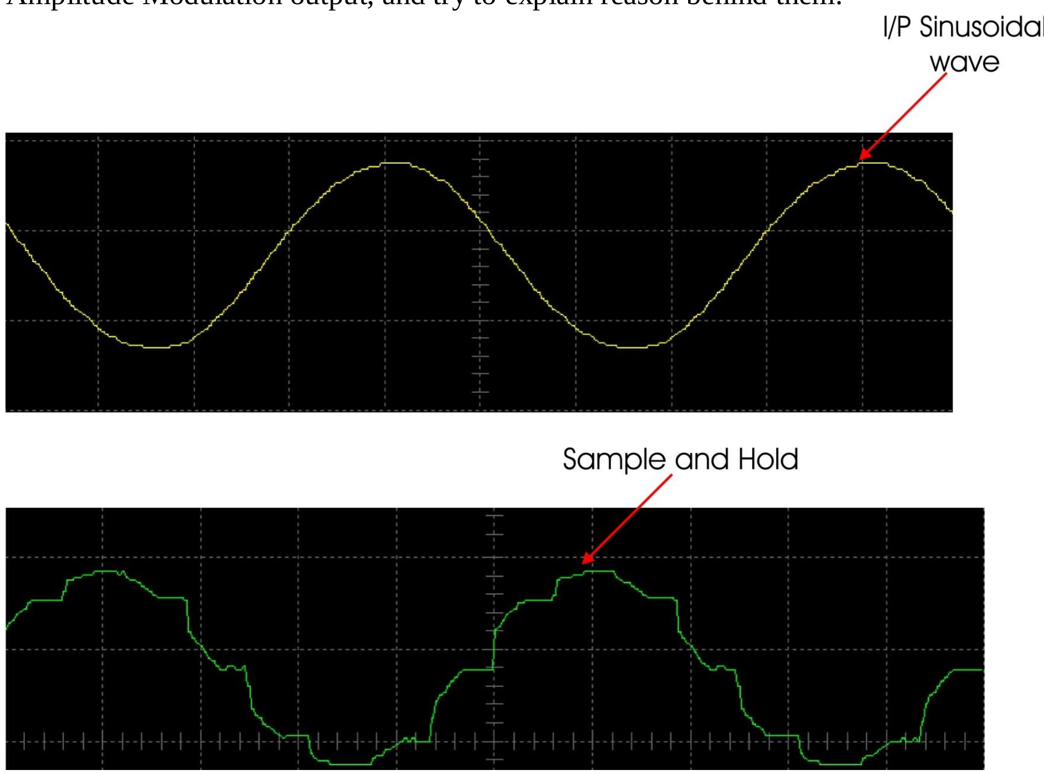


Figure 2.1

Procedure:

1. Connect the circuit as shown in Figure 2.1.
    - **a.** Output of sine wave to modulation signal IN of PAM block.
    - **b.** 8 KHz pulse output to pulse IN of PAM block.
    - **c.** Keep the frequency selector switch in 1 KHz position.
2. Switch 'On' the power supply & oscilloscope.
3. Monitor the output of sample & hold circuit at TP4.
4. Vary the amplitude of input sine wave & its frequency by the frequency change over switch to 2 KHz.
5. Also vary the input pulse's frequency by connecting the pulse input to different Square wave frequencies available on-board viz. 16, 32, & 64 KHz.

    Switch 'On' fault No. 1, 2 & 3, one by one & observe their effect on Pulse Amplitude Modulation output, and try to explain reason behind them.



Sample & Hold circuit Output

Questions:

1. What do you understand by sample and hold circuit?
2. What is the importance of doing sample and hold of signals?
3. Draw the circuit diagram of sample and hold circuit?
4. Write the advantages of sample and hold circuit.

Experiment 3

Objective: Study of Pulse Amplitude Modulation & Demodulation with Sample, Sample & Hold & Flat Top

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

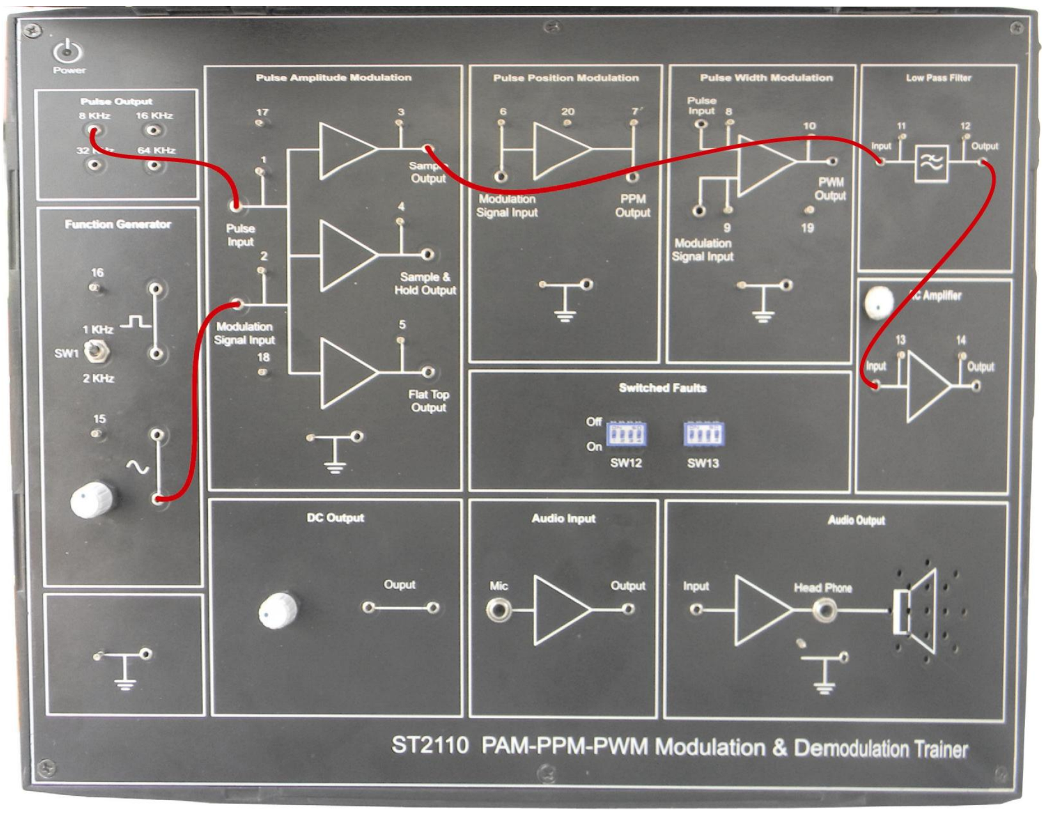


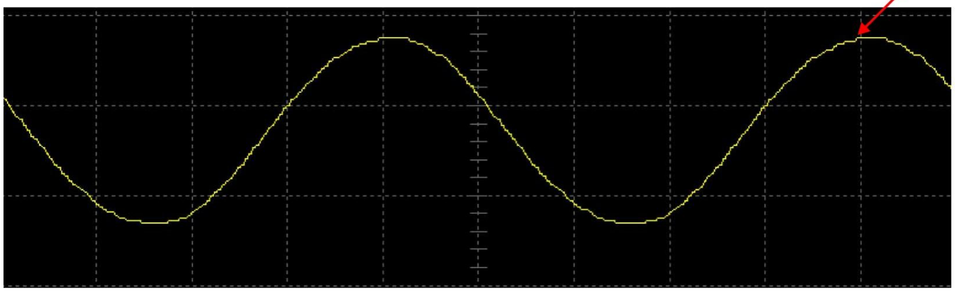
Figure 3.1

Procedure:

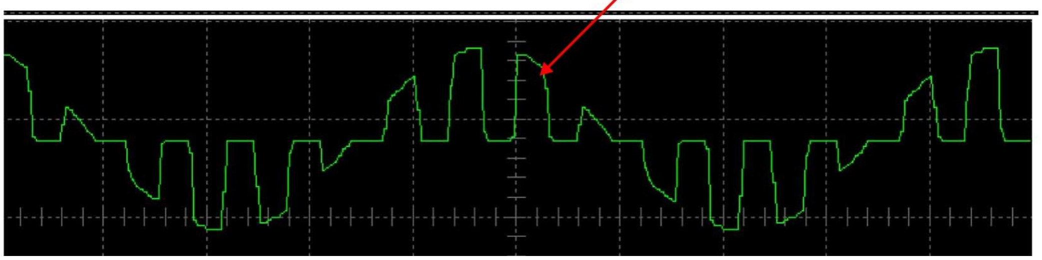
1. Connect the circuit as shown in Figure 3.1.
    a. Output of sine wave to modulation signal IN in PAM block keeping the switch in 1 KHz position.
    b. 8 KHz pulse output to pulse input.
    c. Connect the sample output to low pass filter input.
    d. Output of low pass filter to input of AC amplifier. Keep the gain pot in AC amplifier block in anti clock wise position.
2. Switch ‘On’ the power supply & oscilloscope.
3. Observe the outputs at TP (3 & 5) these are natural & flat top outputs respectively.
4. Observe the difference between the two outputs.
5. Vary the amplitude potentiometer and frequency change over switch & observe the effect on the two outputs.
6. Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.
7. Switch ‘On’ fault No. 1, 2, 3, 4 one by one & observe their effect on Pulse Amplitude Modulation output and try to locate them.
8. Monitor the output of AC amplifier. It should be a pure sine wave similar to input.
9. Vary the amplitude of input, the amplitude of output will vary.
10. Similarly connect the sample & hold & flat top outputs to low pass filter and see the demodulated waveform at the output of AC amplifier.
11. Switch ‘On’ the switched faults No. 1, 2, 3, 4, 5 & 8 one by one and see their effects on output.
12. Switch ‘Off’ the power supply.

Related Wave Forms:

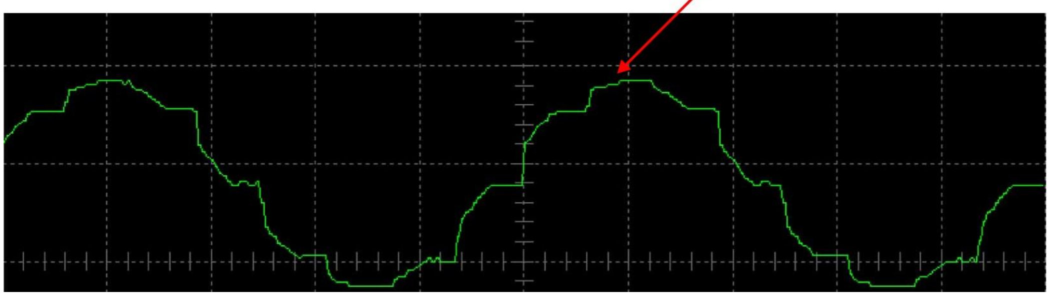
I/P Sinusoidal wave



Sampled O/P



Sample and Hold



Flat Top O/P

Questions:

1. Define PAM in brief?
2. Why flat top sampling is better than natural sampling?
3. What do you understand by sample and hold circuit?
4. What is the importance of doing sample and hold of signals?
5. Write the advantages of sample and hold circuit.

Experiment 4

Objective: Study of PPM using DC Input

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

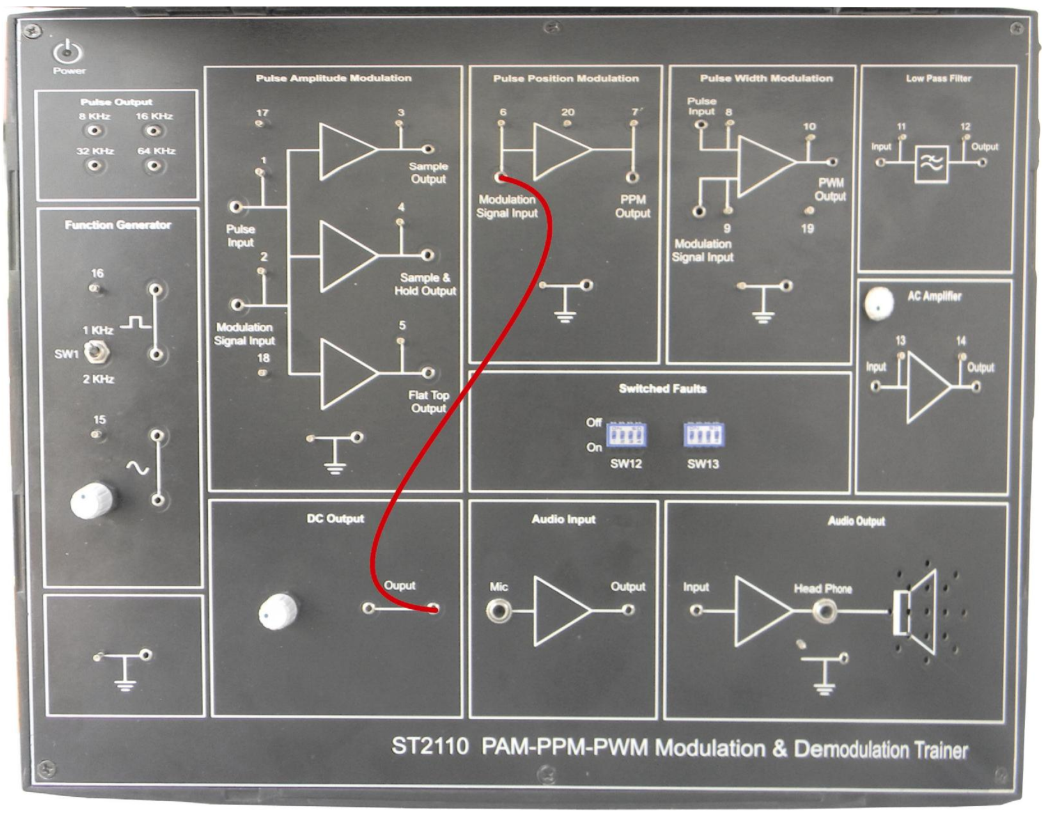


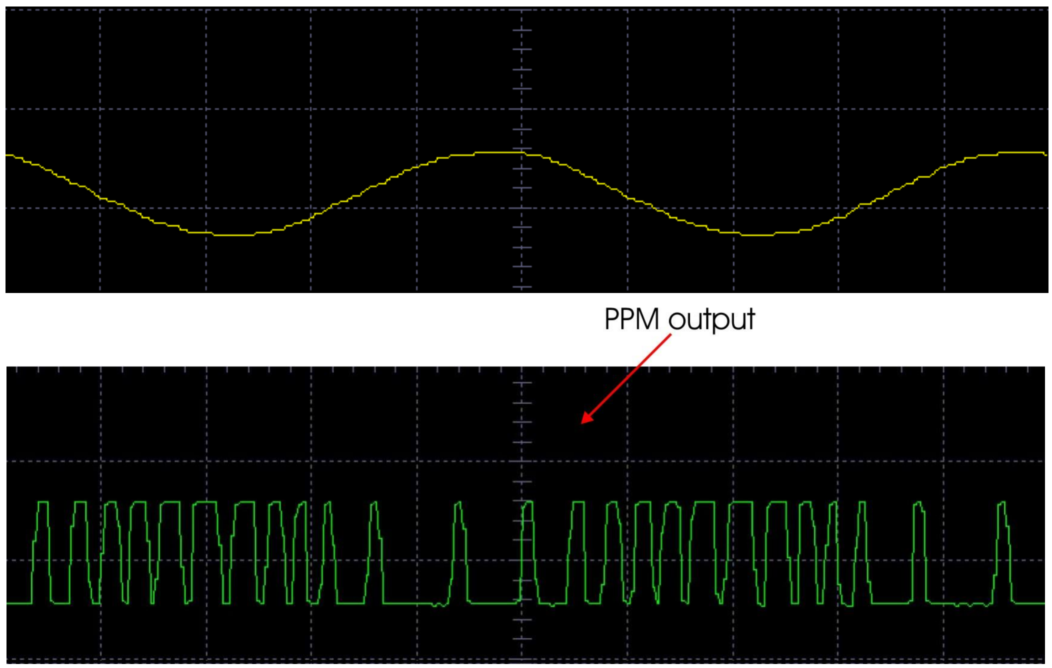
Figure 4.1

Procedure:

1. Connect the circuit as shown in Figure 4.1 and also described below for clarity.
   - **a.** Connect the DC output to input of PPM block.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the output of PPM block at TP7.
4. Vary the DC output while observing the output of PPM block.
5. Switch 'On' the switched faults No. 1, 2, & 6 one by one & observe their effects PPM input and try to locate them.

6. Switch ‘Off’ the power supply.

Related Wave Form:



Questions:

1. What is Pulse position modulation?
2. Draw the output waveform of Pulse position modulation?
3. Write the advantages and disadvantages of Pulse position modulation?
4. Write the applications of Pulse position modulation?

Experiment 5

Objective: Study of PPM using Sine wave Input

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

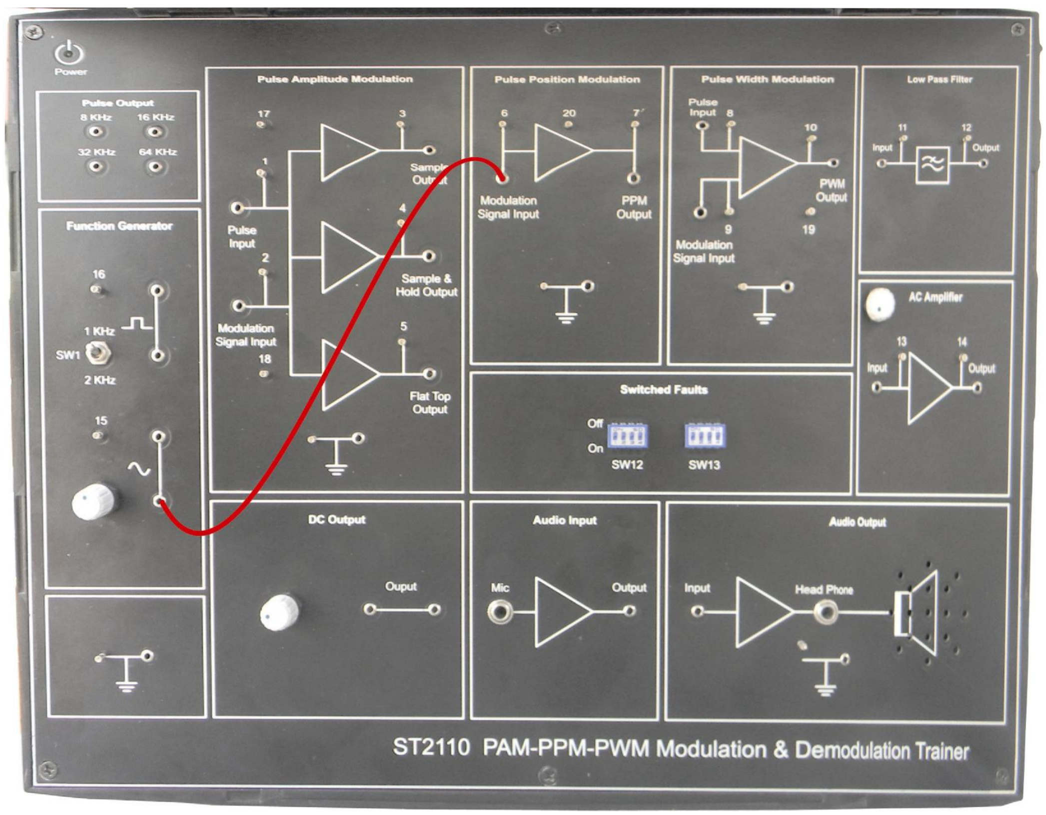
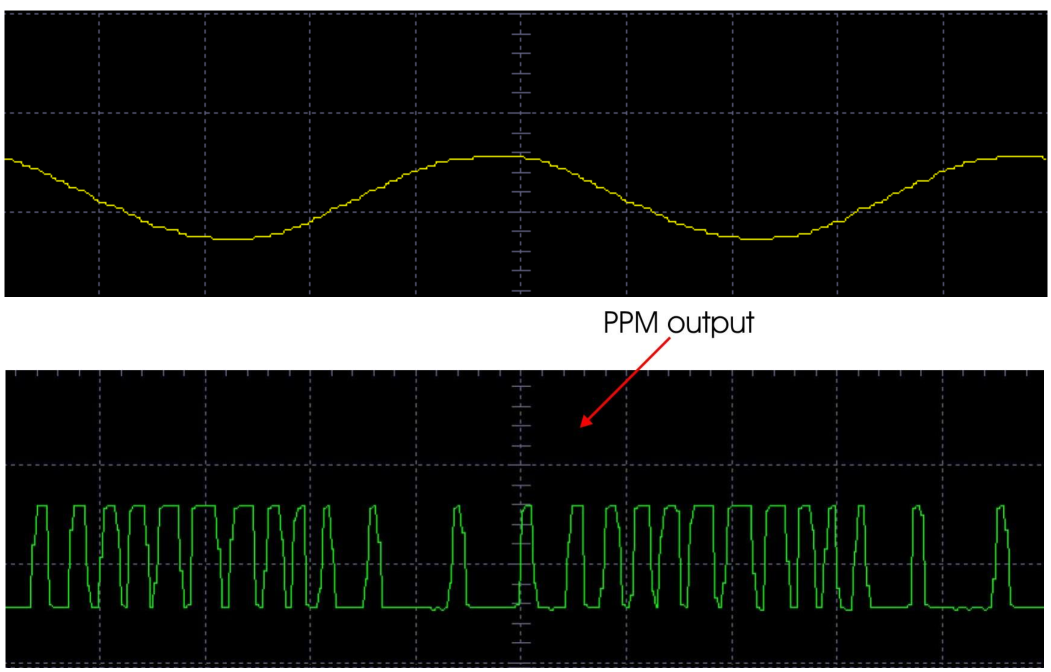


Figure 5.1

Procedure:

1. Connect the circuit as shown in Figure 5.1 and also described below for clarity.
   a. Input of pulse position modulation blocks to sine wave output of FG block.
2. Switch 'On' the power supply & oscilloscope.
3. Keep the oscilloscope at 0.5mS / div, time base speed and in X-5 mode, and observe the pulse position modulated waveform at the pulse position modulation block output.
4. Vary the amplitude of sine wave and observe the pulse position modulation, keep the amplitude preset in center. Here you can best observe the pulse modulation.
5. Switch 'On' fault No. 1, 2, & 6 one by one & observe their effects in pulse position modulation output and try to locate them.
6. Switch 'Off' the power supply.



Questions:

1. What is Pulse position modulation?
2. Draw the output waveform of Pulse position modulation?
3. Write the advantages and disadvantages of Pulse position modulation?
4. Write the applications of Pulse position modulation?
5. What is the difference in applying DC input and Sine wave input?

Experiment 6

Objective: Study of PPM Demodulation

Equipment Required:

1. ST2110 with Power Supply Cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

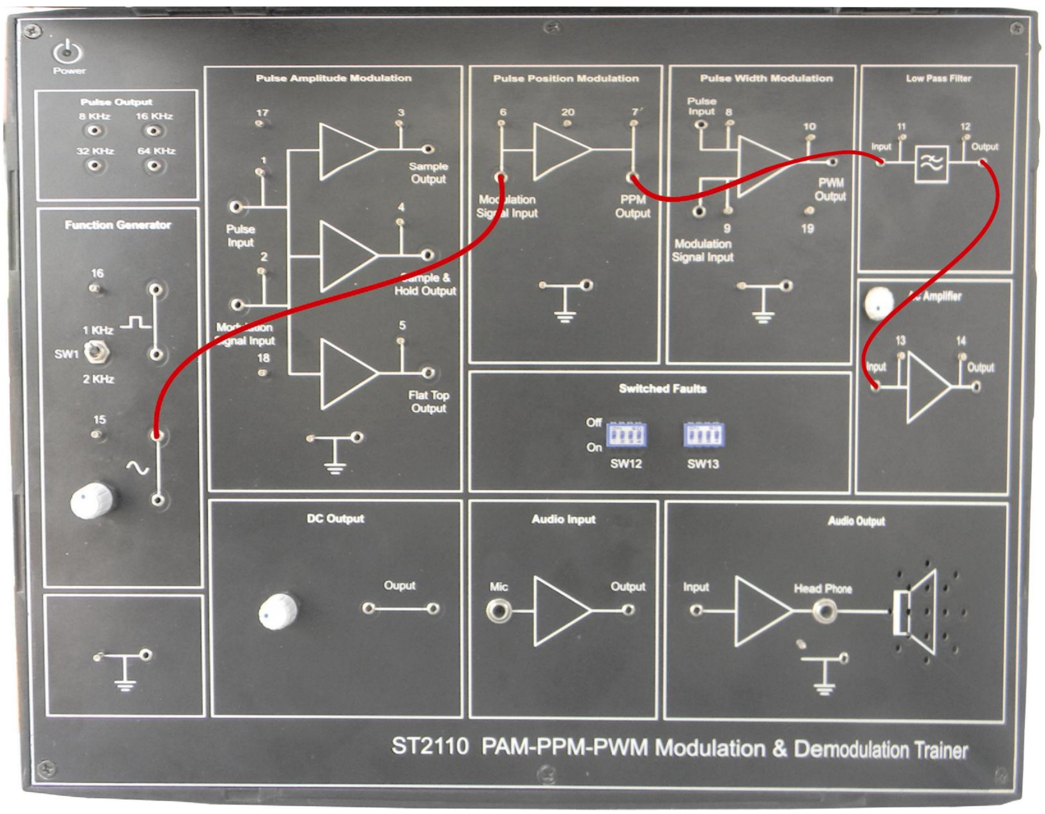


Figure 6.1

Procedure:

1. Connect the circuit as shown in Figure 6.1 and also described below for clarity.
   - **a.** Sine wave of 1 KHz to input of PPM block.
   - **b.** Output PPM block to input of low pass filter.
   - **c.** Output of low pass filter to input of AC amplifier.
   - **d.** Keep the gain potentiometer in amplifier block at maximum position.
2. Switch ‘On’ the power supply & oscilloscope.
3. Observe the waveform at the TP12 output of low pass filter block.
4. Then observe the demodulated output at TP14 output of AC amplifier.
5. Switch ‘On’ fault No. 1, 2, 6 & 8 one by one & observes their effect on demodulated waveform & tries to locate them.
6. Switch ‘Off’ the power supply.

Questions:

1. What do you understand by modulation?
2. What do you understand by demodulation?
3. Why modulation is required in digital communication?
4. Why low pass filters are required in demodulation process?
5. Why PPM is not generally used in digital communication system?

Experiment 7

Objective: Study of PWM using different Sampling Frequency

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

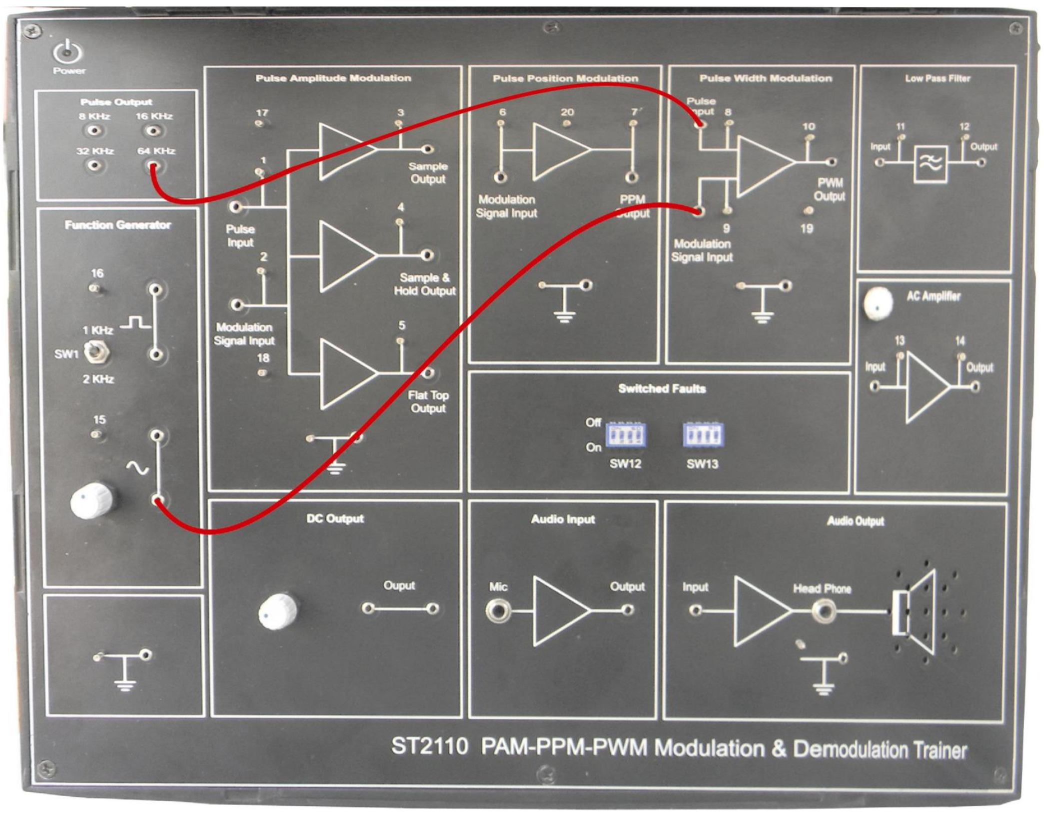


Figure 7.1

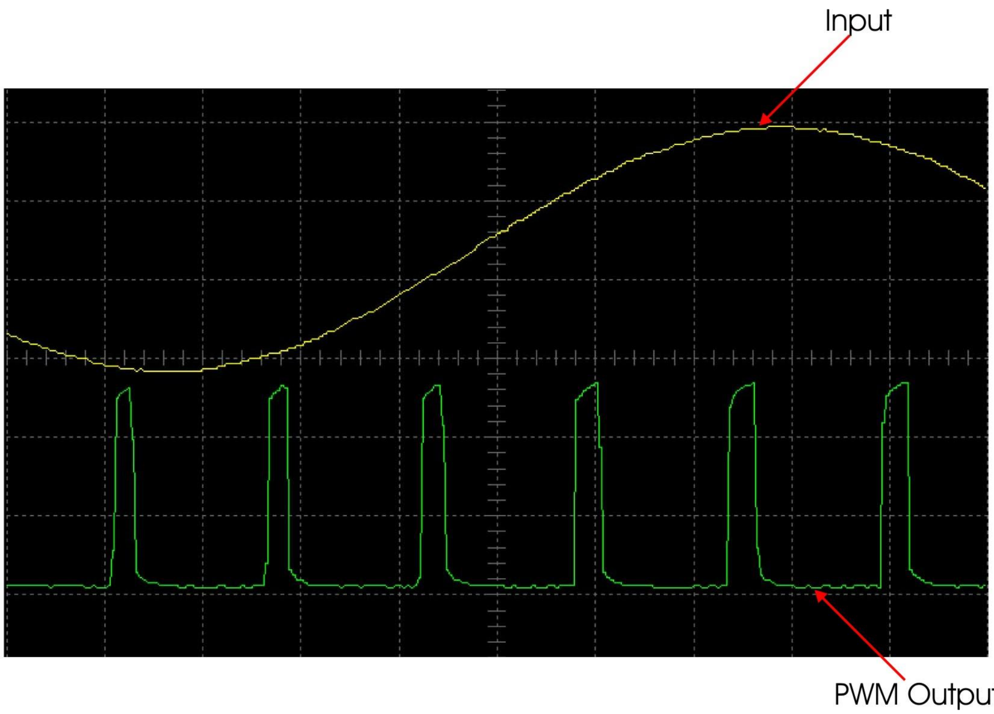
Procedure:

1. Connect the circuit as shown in Figure 7.1 and also described below for clarity.

    a. 1 KHz sine wave output of function generator block to modulation input of PWM block.

    b. 64 KHz square wave output to pulse input of PWM block.

2. Switch ‘On’ the power supply & oscilloscope.
3. Observe the output of PWM block.
4. Vary the amplitude of sine wave and see its effect on pulse output.
5. Vary the sine wave frequency by switching the frequency selector switch to 2 KHz.
6. Also, change the frequency of the pulse by connecting the pulse input to different pulse frequencies viz. 8 KHz, 16 KHz, 32 KHz and see the variations in the PWM output.
7. Switch ‘On’ fault No. 1, 2, & 5 one by one & observes their effect on PWM output and tries to locate them.
8. Switch ‘Off’ the power supply.



Questions:

1. What is meant by sampling?
2. State Sampling theorem.
3. What do you understand by pulse width modulation?
4. What is the effect of varying sampling frequency on pulse width modulation signal?

Experiment 8

Objective: Study of Pulse Width Demodulation

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

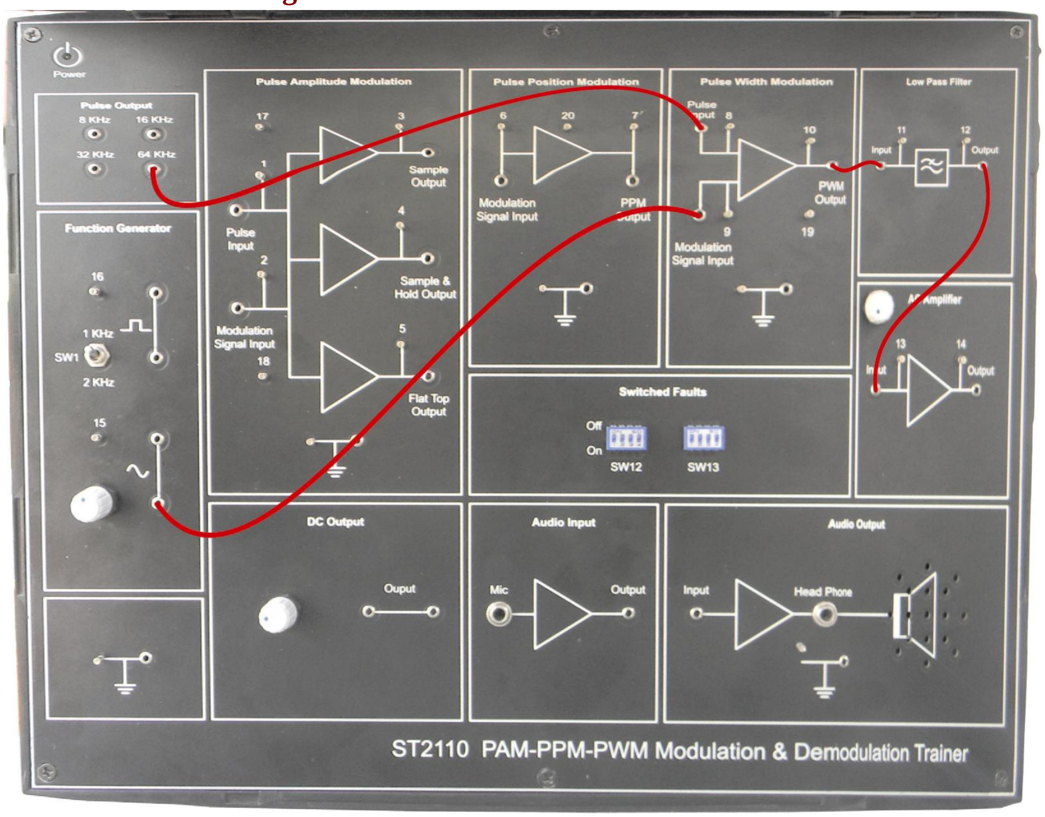


Figure 8.1

Procedure:

1. Connect the circuit as shown in Figure 8.1 and also described below for clarity.
    a. 1 KHz sine wave output of function generator block to modulation input of PWM block.
    b. 64 KHz square wave output to pulse input.
    c. Output of PWM to input of low pass filter.
    d. Output of low pass filter to input of AC Amplifier.
2. Switch ‘On’ the power supply & oscilloscope.
3. Observe the output of low pass filter and AC amplifier respectively to understand the demodulation of pulse width demodulation waveform in detail.
4. Vary the amplitude and frequency of sine wave and observe its effect on the demodulated waveform.
5. Now, connect the pulse input in the pulse width modulation block to the different frequencies available on board viz. 8, 16, 32 KHz and observe their demodulated waveforms.
6. Try varying the amplitude of sine wave signal; you will observe that the output signal varies similarly.
7. Switch ‘On’ fault no, 1, 2, 5 & 8 one by one at a time. Observe their effects on final output and try to locate them.
8. Switch ‘Off’ the power supply.

Questions:

1. What do you understand by modulation?
2. What do you understand by demodulation?
3. Why AC amplifiers are used in demodulation process?
4. Why low pass filters are required in demodulation process?
5. What is the effect of sampling frequency on demodulation output?

Experiment 9

Objective: Study of Voice Link Using Pulse Amplitude Modulation

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Connection Block Diagram:

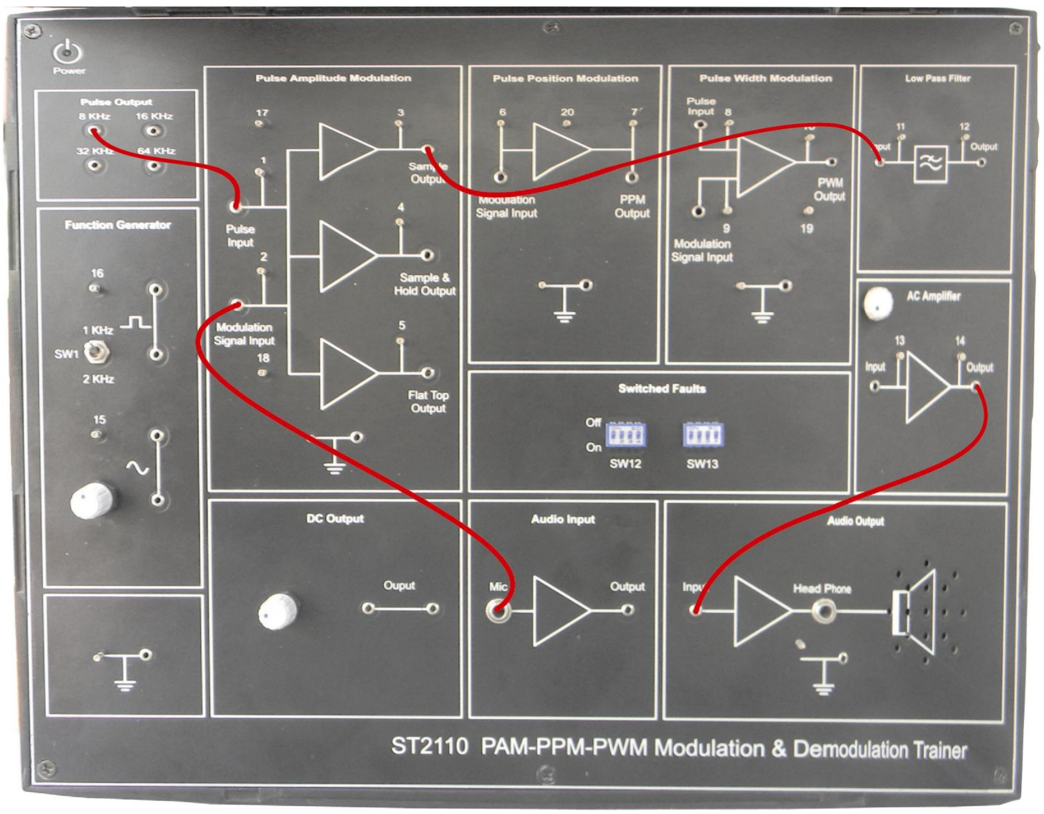


Figure 9.1

Procedure:

1. Connect the circuit as shown in Figure 9.1 and also described below for clarity.
    - **a.** Connect a microphone in the MIC socket in audio input block.
    - **b.** Connect the output of audio input block to MOD input of PAM block.
    - **c.** Connect the 8 KHz pulse output to pulse input of PAM block.
    - **d.** The sample output of PAM block to input of low pass filter.
    - **e.** Output of low pass filter to AC amplifier.
    - **f.** Gain pot of AC amplifier in mid position.
    - **g.** Output of AC amplifier to input of audio output block.
2. Switch ‘On’ the power supply.
3. You can observe the pulse being modulated by audio signal at output of sample output, sample & hold & flat top outputs.
4. Also, you can observe its demodulation and hear the same voice in speaker /headphone which was fed in the microphone in the input.
5. Switch ‘Off’ the power supply.

Questions:

1. What is the difference between analog and digital signals?
2. What is the frequency band for voice signals?
3. What is aliasing?
4. Why sample and hold output is more reliable than sampled output signal.
5. What is the effect of using Pulse Amplitude Modulation for voice communication?

Experiment 10

Objective: Study of Voice Link using Pulse Position Modulation

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords
4. MIC with connecting cord

Connection Block Diagram:

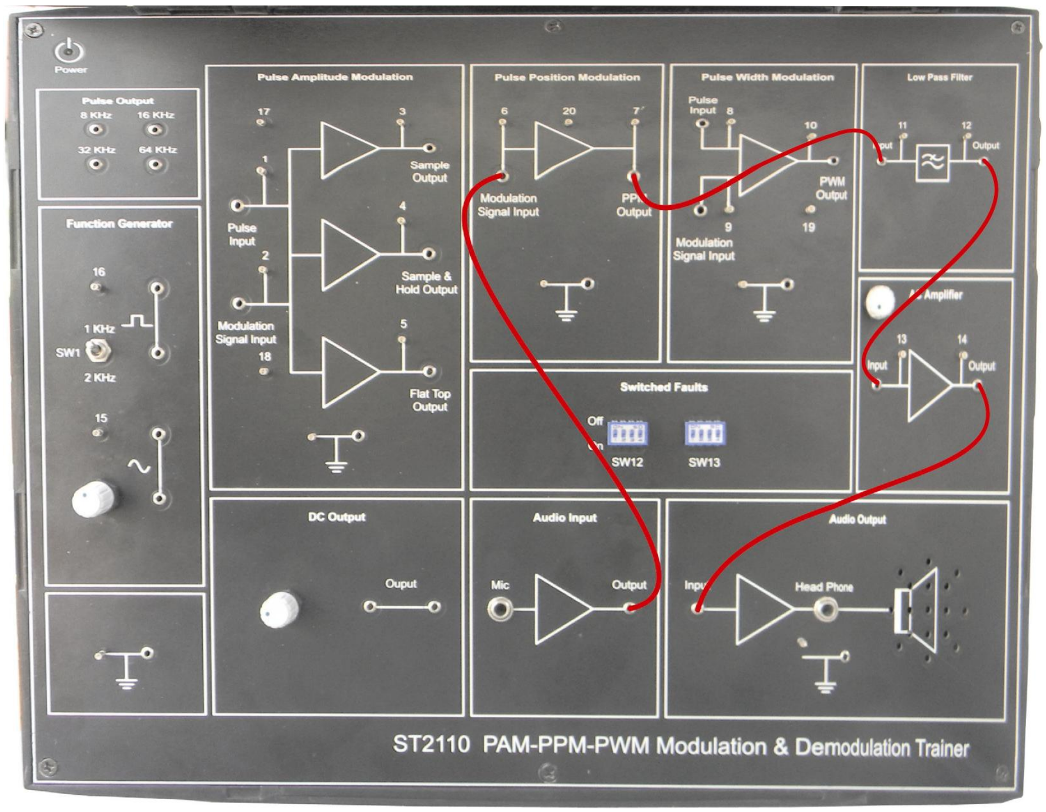


Figure 10.1

Procedure:

1. Connect the circuit as shown in Figure 10.1 and also described below for clarity.

   a. Connect a microphone in MIC socket of audio input block.

   b. Output of audio input block to input of PPM block.

   c. Output of PPM block to input of low pass filter block.

   d. Output of low pass filter block to input of AC amplifier block.

   e. Keep the frequency selector switch in 1 KHz position.

   f. Keep the gain preset of AC amplifier in mid position.

   g. Connect the output of AC amplifier block to input of audio output block.

2. Switch ‘On’ the power supply.

3. You can study the PPM using voice, by observing the waveforms at different stages.

4. The input is heard by means of speaker or headphone.

5. Switch ‘Off’ the power supply.

Questions:

1. What is the difference between analog and digital signals?
2. What is the frequency band for voice signals?
3. What is aliasing?
4. Why sample and hold output is more reliable than sampled output signal.
5. What is the effect of using Pulse position Modulation for voice communication?

Experiment 11

Objective: Study of Voice Communication using Pulse Width Modulation

Equipment Required:

1. ST2110 with power supply cord
2. CRO with connecting probe
3. Connecting cords
4. MIC with connecting cord

Connection Block Diagram:

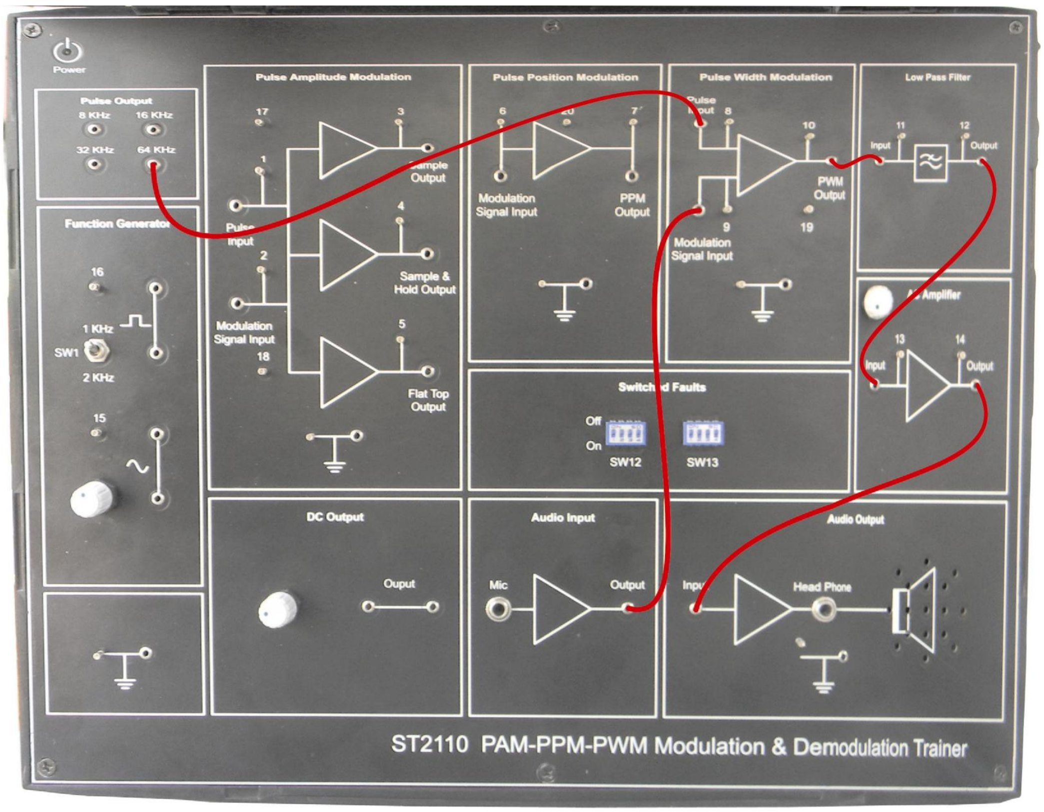


Figure 11.1

Procedure:

1. Connect the circuit as shown in Figure11.1 and also described below for clarity.

   a. Connect a microphone in MIC socket of audio input block.

   b. Output of audio input block to modulation signal in of PWM block.

   c. 64 KHz Square Wave output to pulse input.

   d. Output of PWM to input of low pass filter.

   e. Output of low pass filter to input of AC Amplifier.

   f. Keep the gain preset of AC amplifier in mid position.

   g. Connect the output of AC amplifier block to input of audio output block.

2. Perform the pulse width modulation & demodulation experiment.

3. Switch 'On' the power supply & oscilloscope.

4. You can study the pulse width modulation of a pulse signal by voice signal, by observing the outputs at different stages.

5. The voice output can be heard in speaker or headphone.

6. Switch 'Off' the power supply.

Questions:

1. What is the difference between analog and digital signals?
2. What is the frequency band for voice signals?
3. What is aliasing?
4. Why sample and hold output is more reliable than sampled output signal.
5. What is the effect of using Pulse width Modulation for voice communication?

Switched Faults

This block comprises of eight switched faults. Each of them affects different blocks on ST2110.

1. **Switched Fault 1**

   It disconnects the crystal output frequency from the input pin no. 8 of IC2 (74HC4040).

2. **Switched Fault 2**

   It disconnects the square wave input to the sine wave converter.

3. **Switched Fault 3**

   It disconnects the modulating input to the IC4 (DG 211).

4. **Switched Fault 4**

   It cuts the feedback path, and affects the Flat Top output.

5. **Switched Fault 5**

   It disconnects the link from transistor Ql collector to pin no. 6 & 7 of IC6 (7555).

6. **Switched Fault 6**

   It disconnects the output of IC 7555 from input ofIC8 (b) 74HC00.

7. **Switched Fault 7**

   It disconnects the R27 (l.2K) in PWM block from the Ground link.

8. **Switched Fault 8**

   It disconnects the feedback path of IC9 (d) TL074.

FAQ

1. What do you mean by sampling?

Ans: To convert continuous time signal to discrete time signal, a process is used called as sampling.

2. What is sampling theorem?

Ans: The Sampling Theorem states that a signal can be exactly reproduced if it is sampled at a frequency F_s, where F_s is greater than twice the maximum frequency F_{max} in the signal.

$$F_s > 2 \cdot F_{max}$$

3. What is Nyquist frequency?

Ans: The frequency $2 \cdot F_{max}$ is called the Nyquist sampling rate. Half of this value, F_{max}, is sometimes called the Nyquist frequency.

4. List different sampling techniques?

Ans: There are three types of sampling, which are as follows:

1. Ideal sampling or Instantaneous sampling or Impulse sampling
2. Natural sampling
3. Flat top sampling

5. What is under sampling?

Ans: When the sampling rate is lower than or equal to the Nyquist rate, a condition defined as under sampling, it is impossible to rebuild the original signal according to the sampling theorem.

6. What do you mean by aliasing?

Ans: Aliasing is the presence of unwanted components in the reconstructed signal. These components were not present when the original signal was sampled. In addition, some of the frequencies in the original signal may be lost in the reconstructed signal. Aliasing occurs because signal frequencies can overlap if the sampling frequency is too low. As a result, the higher frequency components roll into the reconstructed signal and cause distortion of the signal Frequencies "fold" around half the sampling frequency. This type of signal distortion is called aliasing.

7. Explain the process of sample and hold?

Ans: In electronics, a sample and hold circuit is used to interface real-world signals, by changing analogue signals to a subsequent system. The purpose of this circuit is to hold the analogue value steady for a short time while the converter or other following system performs some operation that takes a little time.

Sampling mode:
In this mode, the switch is in the closed position and the capacitor charges to the instantaneous input voltage.

Hold mode:
In this mode, the switch is in the open position. The capacitor is now disconnected from the input. As there is no path for the capacitor to discharge, it will hold the voltage on it just before opening the switch. The capacitor will hold this voltage till the next sampling instant.

8. How aliasing is removed?
Ans: Aliasing is removed by simply filtering out all the high frequency components before sampling.

9. List methods to avoid aliasing?
Ans: To avoid the aliasing there are two approaches:

1. To raise the sampling frequency to satisfy the sampling theorem,

2. The other is to filter off the unnecessary high-frequency component from the continuous-time signal. We limit the signal frequency by an effective low pass filter, called anti aliasing pre filter, so that the remained highest frequency is less than half of the intended sampling rate. If the filter is not perfect we must give some allowance.

10. What are active and passive filter?
Ans: Filter is a network designed to pass signals having frequencies within certain bands (called pass bands) with little attenuation, but greatly attenuates signals within other bands (called attenuation bands or stop bands).

A filter network containing no source of power is termed passive, and one containing one or more power sources is known as an active filter network.

11. What do you mean by PAM?
Ans: A sampled signal consists of a train of pulses, where each pulse corresponds to the amplitude of the signal at the corresponding sampling time. The signal sent to line is modulated in amplitude and hence the name Pulse Amplitude Modulation (PAM).

12. What is frequency range for speech information?
Ans: The frequency range for speech information is 300 Hz to 3.4 KHz.

13. Define duty cycle?
Ans: The duty cycle of a signal is defined as, the ratio of pulse duration to the pulse repetition period. This ratio can also be expressed as percentage. e.g. the square wave has equal pulse and no pulse duration; hence its duty cycle is 0.5 or 50%

14. What are the aspects of Pulse Amplitude Modulation system?

Ans: Pulse Amplitude Modulation system governs the following important aspects:

a. The narrower pulses allow us to time division multiplex many such pulse amplitude modulation panels i.e. we can send any no. of pulse amplitude modulated signals over same channel at a time. Hence lower duty cycle beneficial in this respect.

b. The narrower pulses have wider frequency spectrum. Hence the wider bandwidth channel is required.

c. Narrower pulses have less power as the power content of a pulse depends on its amplitude and width. During transmission and demodulation the inherent noise can play a major havoc on the low power signal. Hence a pulse of larger duty-cycle is desirous for this sake.

15. What are the various pulse modulation techniques?

Ans: There are four types of pulse modulation techniques, which are as follows:

a. Pulse Amplitude Modulation (PAM):

In pulse amplitude modulation system the amplitude of the pulse is varied in accordance with the instantaneous level of the modulating signal. Now days, the PAM system is not generally used, but it forms the first stage of the other types of pulse modulation.

b. Pulse Width Modulation (PWM):

In PWM system the width of the pulse is varied in accordance with the instantaneous level of the modulating signal.

c. Pulse Position Modulation (PPM):

In PPM System, the position of the pulse relative to the zero reference level is varied in accordance with the instantaneous level of the modulating signal.

d. Pulse Code Modulation (PCM):

In PCM System the amplitude of the sampled waveform at definite time intervals is represented as a binary code. The first three techniques of the above described systems are not truly digital but in fact are analog in nature. The very fact that the variation of a particular pulse parameter is continuous rather than being in the discrete steps makes the system analog in nature.

16. What is pulse position modulation?

Ans: The Amplitude and width of the pulses is kept constant in this system, while the position of each pulse, in relation to the position of a recurrent reference pulse is varied by each instantaneous sampled value of the modulating wave.

17. What is the advantage and disadvantage of pulse position modulation?

Ans: Pulse-position modulation has the advantage of requiring constant transmitter power output. And the disadvantage of depending on transmitter and receiver is synchronization.

Warranty

1. We guarantee this product against all manufacturing defects for 24 months from the date of sale by us or through our dealers. Consumables like dry cell etc. are not covered under warranty.
2. The guarantee will become void, if
   a) The product is not operated as per the instruction given in the Learning Material
   b) The agreed payment terms and other conditions of sale are not followed.
   c) The customer resells the instrument to another party.
   d) Any attempt is made to service and modify the instrument.
3. The non-working of the product is to be communicated to us immediately giving full details of the complaints and defects noticed specifically mentioning the type, serial number of the product and date of purchase etc.
4. The repair work will be carried out, provided the product is dispatched securely packed and insured. The transportation charges shall be borne by the customer.

List of Accessories

1. Patch Cord 16".. 5 Nos.
2. Mains Cord..1 No
3. Head Phone..1 No.
4. Microphone ....1 No.
5. Learning Material ....1 No.