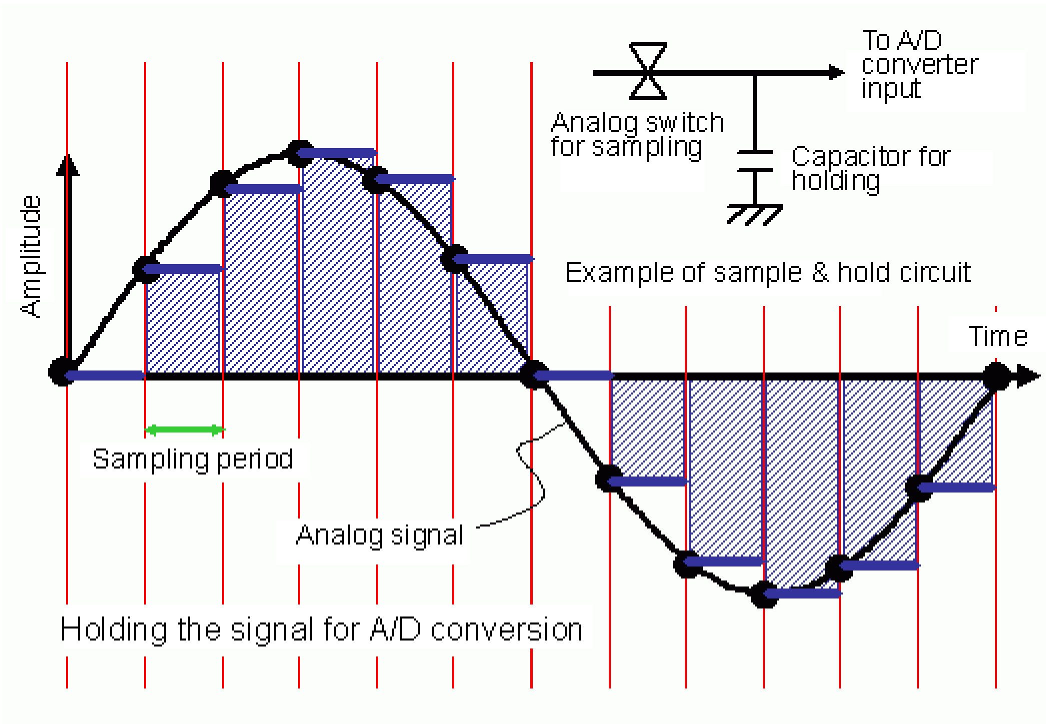
TO STUDY SAMPLE AND HOLD CIRCUIT



TO STUDY PULSE AMPLITUDE MODULATION & DEMODULATION

AIM:

To study the sample and hold circuit.

APPARATUS USED:

(i) Sample and hold circuit Trainer Kit (ii) CRO with connecting probes
(iii) Connecting cords.

THEORY:

In analog communication systems like AM, FM the instantaneous value of the Information signal is used to change certain parameter of the carrier signal.

Pulse modulation systems differ from these systems in a way that they transmit a limited number of discrete states of a signal at a predetermined time. Sampling can be defined as measuring the value of an information signal at predetermined time intervals. The rate at which the signal is sampled is known as sampling rate or sampling frequency. It is the major parameter, which decides the quality of the reproduced signal. If the signal is sampled quite frequently (whose limit is specified by Nyquist Criterion) then it can be reproduced exactly at the receiver with no distortion.

Nyquist Criterion:

As shown in fig. 1, the lowest sampling frequency that can be used without the side bands overlapping is twice the highest frequency component present in the information signal. If we reduce this sampling frequency even further, the side bands 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.

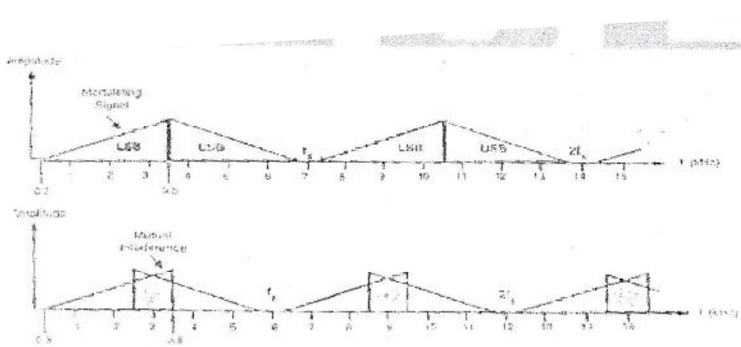


Fig.1

The Nyquist Criterion 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 known as NYQUIST RATE i.e. for faithful reproduction of information signal $f_s > 2f_m$.

Pulse amplitude modulation, the simplest form of pulse modulation. It forms an excellent introduction to pulse modulation in general. Pulse amplitude modulation is 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 fig 2. 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.

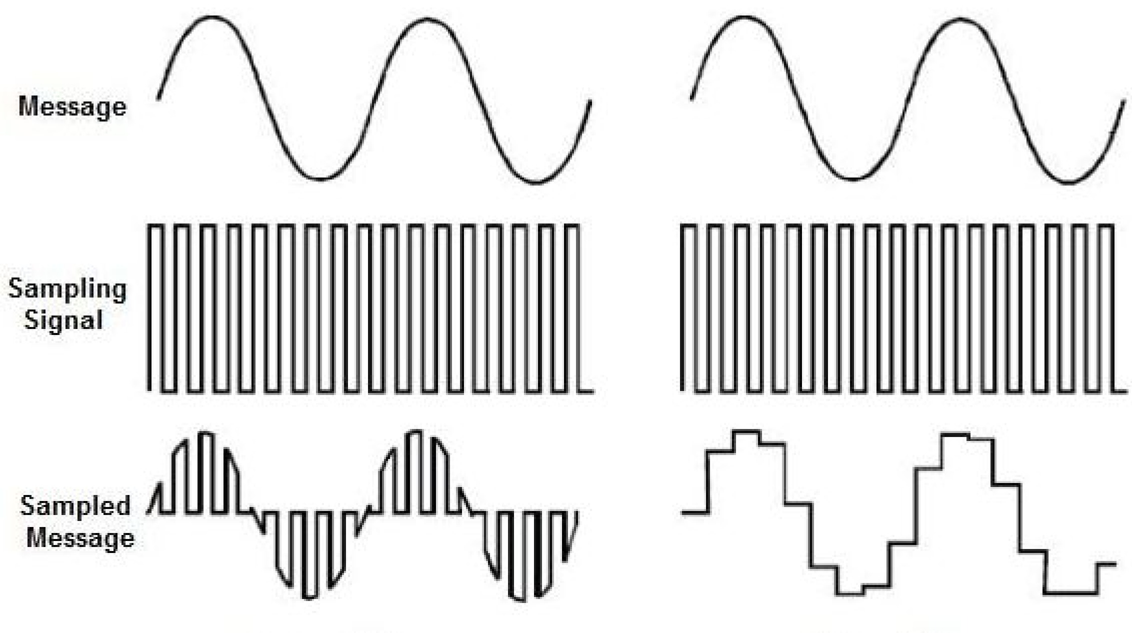


Fig.2

If the pulse width of the carrier pulse train used in natural sampling is made very short compared to the pulse period, the natural pulse amplitude modulation is referred to as instantaneous pulse amplitude modulation. As it has been discussed, shorter pulse is desirous for allowing many signals to be included in TDM format but the pulse can be highly corrupted by noise due to lesser signal power.

One way to maintain reasonable pulse energy is to hold the sample value until the next sample is taken. This technique is formed as Sample-and-Hold techniques. The Sample-and-Hold waveform looks as shown under fig 3.

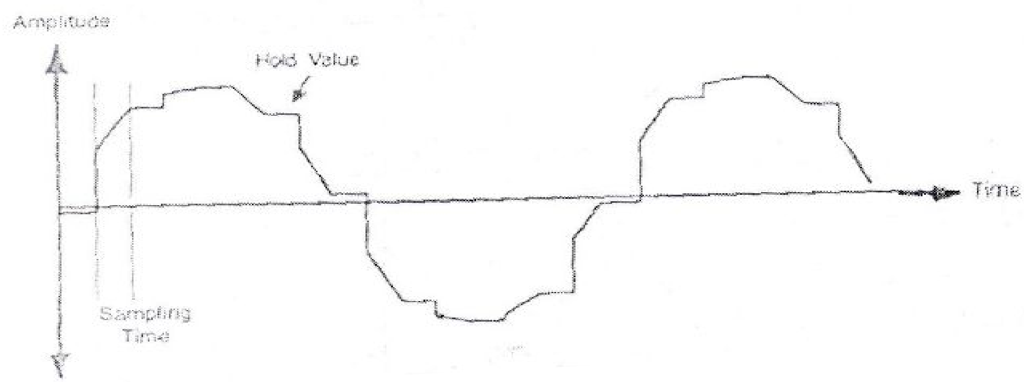
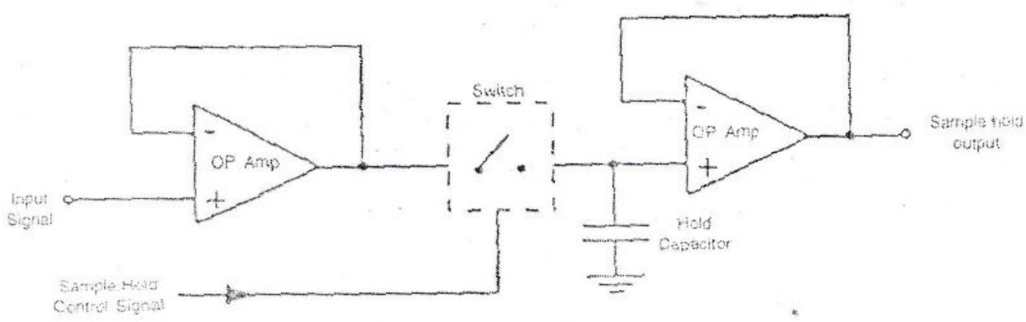


Fig. 3

Now 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. The hold facility can be provided by a capacitor. When the switch connects the capacitor to the pulse amplitude modulation output it changes to the instantaneous value. A buffered sample and hold circuit consists of unity gain buffers preceding and succeeding the charging capacitor. The high input impedance of the proceeding 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 fig 4.



Sample/Hold Circuit

Fig. 4

The high input impedance of the succeeding buffers prevents the charge 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 positive 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 is important parameter in sample and hold circuit design.

Important Parameter of Sample and 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 reduced to zero because on application of finite time taken by a switch to close/open on application of hold signal. Therefore a small value aperture time is sought after. See fig 5.

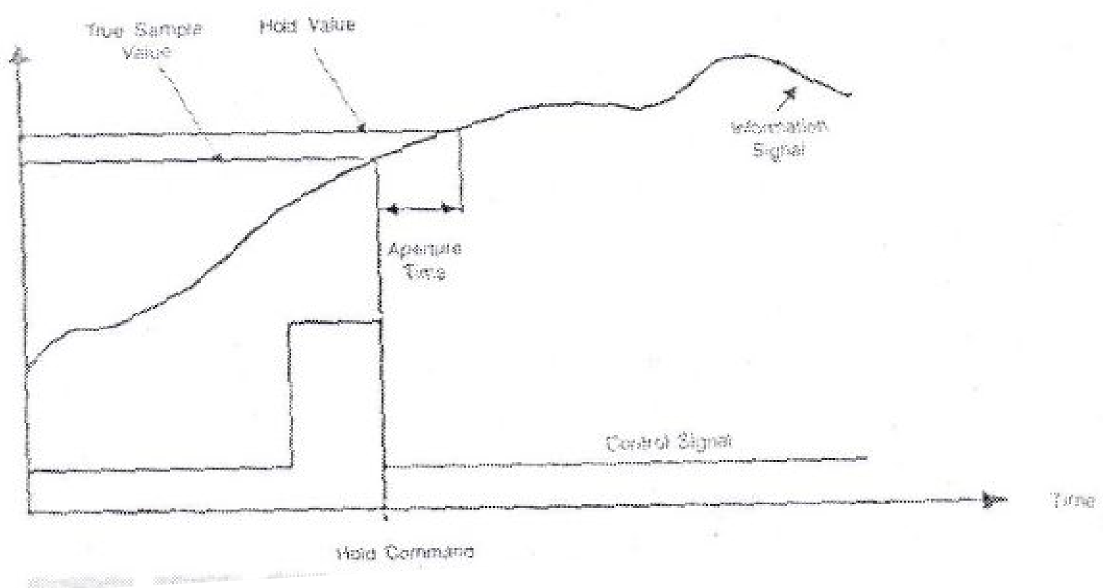


Fig. 5

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.

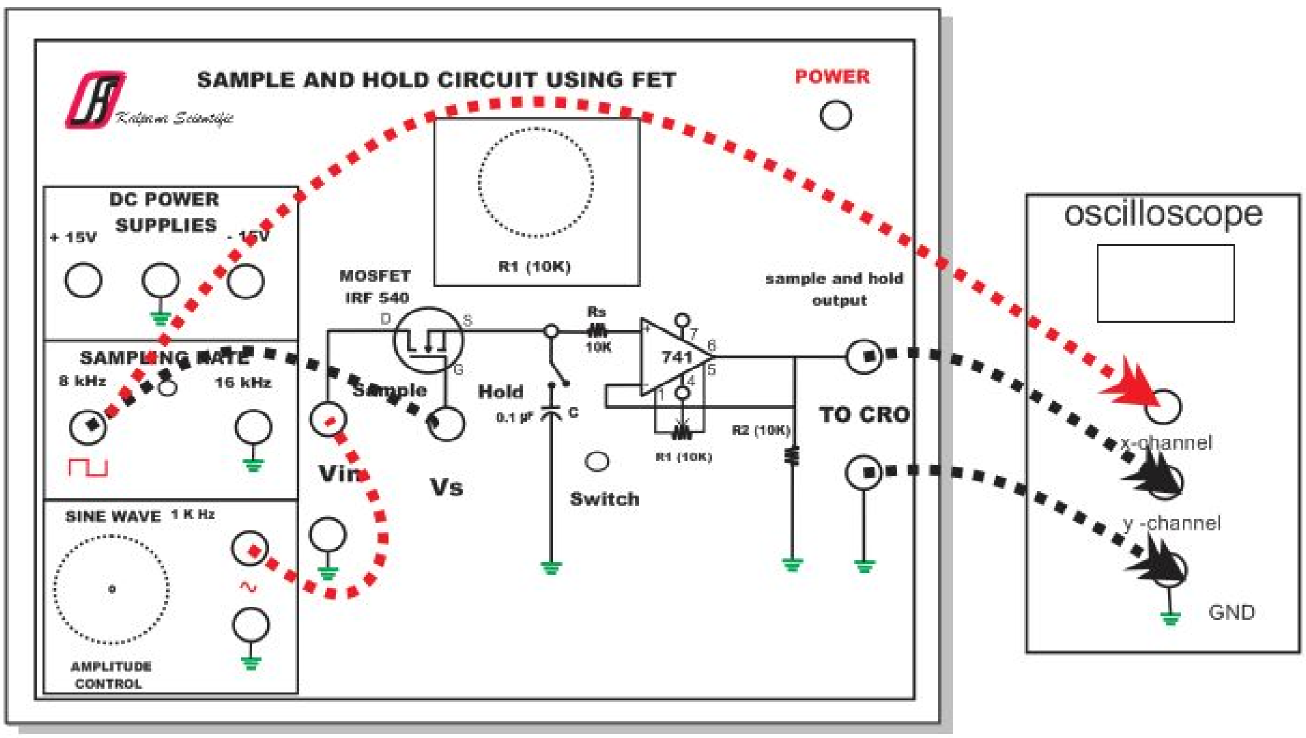
3. Drop Rate: As it has been discussed earlier, the presence of leakage current through capacitor dielectric to +ve input of succeeding buffer causes charge loss of capacitor.

Hence the voltage level at the output falls with 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 sample and hold circuit is called feed through.

PROCEDURE:

Connection Diagram:

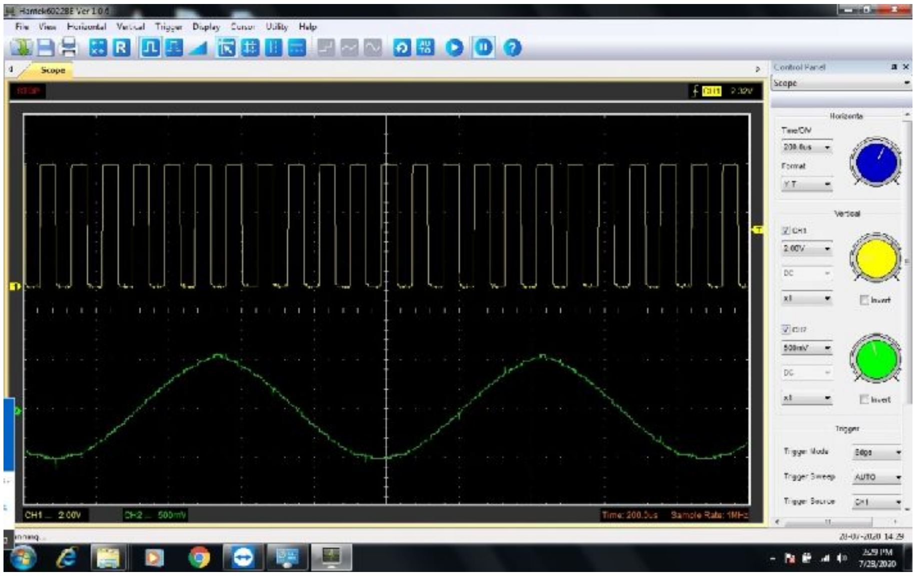


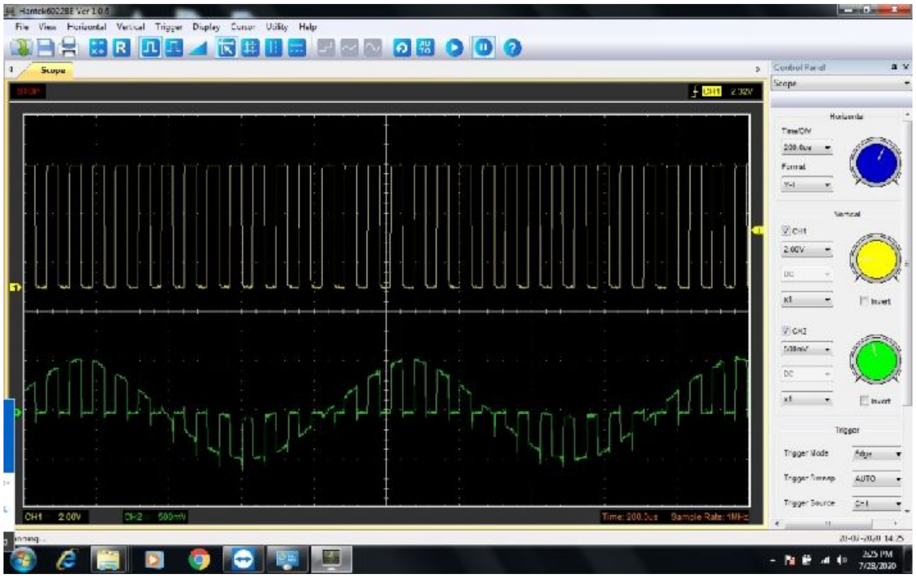
- Switch on the sample and hold circuit trainer kit and CRO.
- Connect the circuit as shown in connection diagram,
- Give sine signal and sampling signal.
- Observe sample and hold circuit output at sample and hold output.

OBSERVATION TABLE:

Sl. No.	Modulating Signal		Gating Pulse	
	Amplitude	Frequency	Amplitude	Frequency

RESULTS AND GRAPHS (SAMPLE):





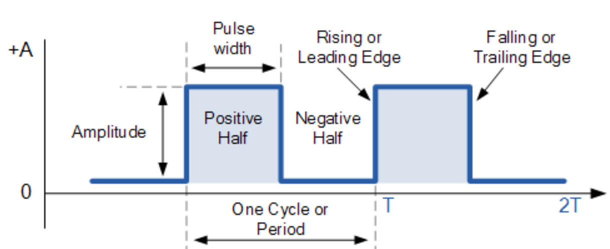
PRECAUTIONS:

- Switch off the experimental kit during making connections.
- Use the CRO carefully.

PROCEDURE TO CALCULATE FREQUENCY:

Frequency: – This is the number of times the **waveform** repeats itself within a one second time period. **Frequency** is the reciprocal of the time period, (f = 1/T) with the standard unit of **frequency** being the Hertz, (Hz).

A Square Wave Waveform



$$\text{Frequency} = \frac{1}{\text{"ON" time} + \text{"OFF" time}}$$