Experiment 5

Objective:

1. **Study of Signal Sampling and Reconstruction Techniques**
2. Study the effect of Sample /Sample and Hold output on reconstructed waveform
3. Comparison of frequency response of 2^{nd} order and 4^{th} order low pass filter.

Equipment Required:

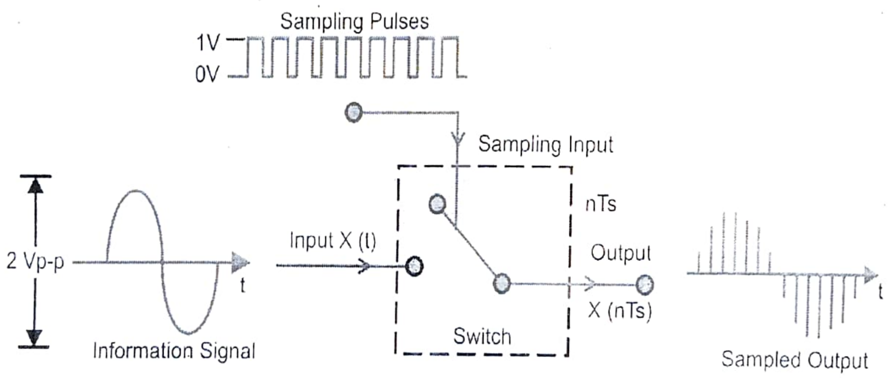
1. **2101 with power supply cord**
2. **CRO with connecting probe**
3. **Connecting cords**

Theory

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.

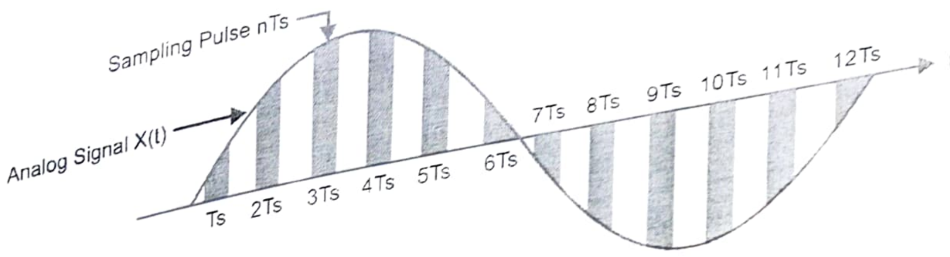
Principle of sampling

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

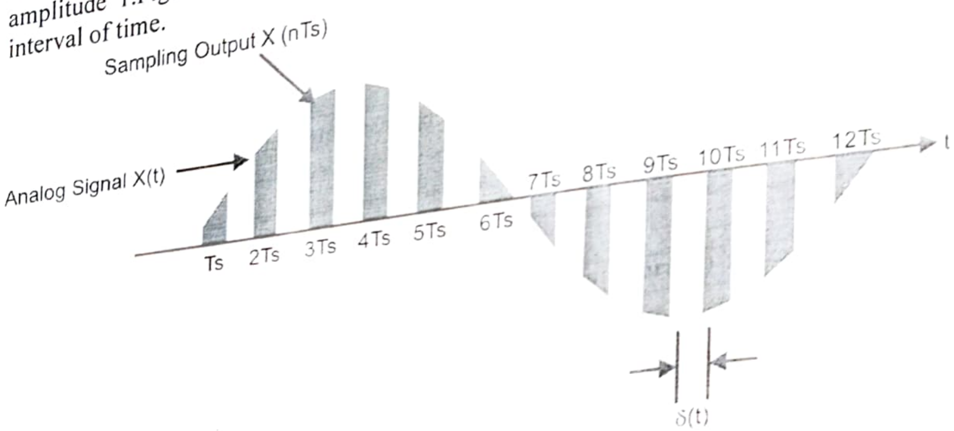
Figure 1



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.

Procedure:

A. Set up for Sampling and reconstruction of signal.

Initial set up of trainer:

Duty cycle selector switch position: Position 5 (to set 50% duty cycle)

Sampling selector switch : Internal position

1. Connect the power cord to the trainer. Keep the power switch in 'Off' position.
2. Connect 1 KHz Sine wave to signal Input.
3. Switch 'On' the trainer's power supply & Oscilloscope.
4. Connect BNC connector to the CRO and to the trainer's output port.
5. Select 320 KHz (Sampling frequency is 1/10th of the frequency indicated by the illuminated LED) sampling rate with the help of sampling frequency selector switch.
6. Observe 1 KHz sine wave (TP12) and Sample Output (TP37) on Oscilloscope. The display shows 1 KHz Sine wave being sampled at 32 KHz, so there are 32 samples for every cycle of the sine wave. (figure 1.1)
7. Connect the Sample output to Input of Fourth Order low pass Filter & observe reconstructed output on (TP46) with help of oscilloscope. The display shows the reconstructed original 1 KHz sine wave. (figure 1.2)
8. By successive presses of sampling Frequency Selector switch, change the sampling frequency to 2KHz, 4KHz, 8KHz, 16KHz and back to 32KHz (Sampling frequency is 1/10th of the frequency indicated by the illuminated LED). Observe how SAMPLE output changes in each cases and how the lower sampling frequencies introduce distortion into the filter's output waveform. This is due to the fact that the filter does not attenuate the unwanted frequency component significantly. Use of higher order filter would improve the output waveform.
9. So far, we have used sampling frequencies greater than twice the maximum input frequency.

Conclusion:

As the sampling frequency increases the output of sample port has more number of samples of applied input signal.

B. Setup for sample and hold output & reconstructed signal output.

1. Repeat the first five steps as above.
2. Connect Sample & Hold output to Fourth Order low pass Filter's Input. Set the Duty Cycle Selector switch to position 5. (figure 2.1)
3. Observe the waveform at Sample & Hold output (TP39) on oscilloscope. Vary the sampling frequency selector from 32 KHz to 2 KHz to illustrate how each sample is held at the sample/hold output. Also observe the filter output at TP46. (figure 2.2)
4. Vary the position of Duty Cycle Selector switch from 0 to 9 and note that

incontrast to step 7, the filter's output amplitude is now independent of the sampling duty cycle and is equal to the amplitude of the original input signal. This is an important result - with Sample and Hold Output, the proportion of sampling time to holding time has no effect on reconstructed waveform provided that Nyquist criteria has been followed. If sample/hold feature is utilized in digital communication system many channels can be multiplexed with maximum amplitude of reconstructed signal.

Conclusion:

For transmitting the signal if a sample and hold amplifier is used just before the transmission channel, the signal will be less suffered from distortion as compared to when only sample amplifier is used.

C. Comparison of frequency response of 2nd order and 4th order low pass filter.

1. Repeat the first five steps above.

2. Connect sample output to input of the Second Order Low Pass Filter and to the input of Fourth Order Low Pass Filter respectively. Observe the outputs of two filters (TP42 and 46 respectively) on the oscilloscope. Vary the sampling frequency with duty cycle set at 5 positions. Compare the output of filter in each case. Note, that the output of fourth order filter always exhibits less distortion than second order filter. This is because fourth order filter has a sharper roll-off and thus rejects (attenuates) more unwanted frequency components caused by sampling.

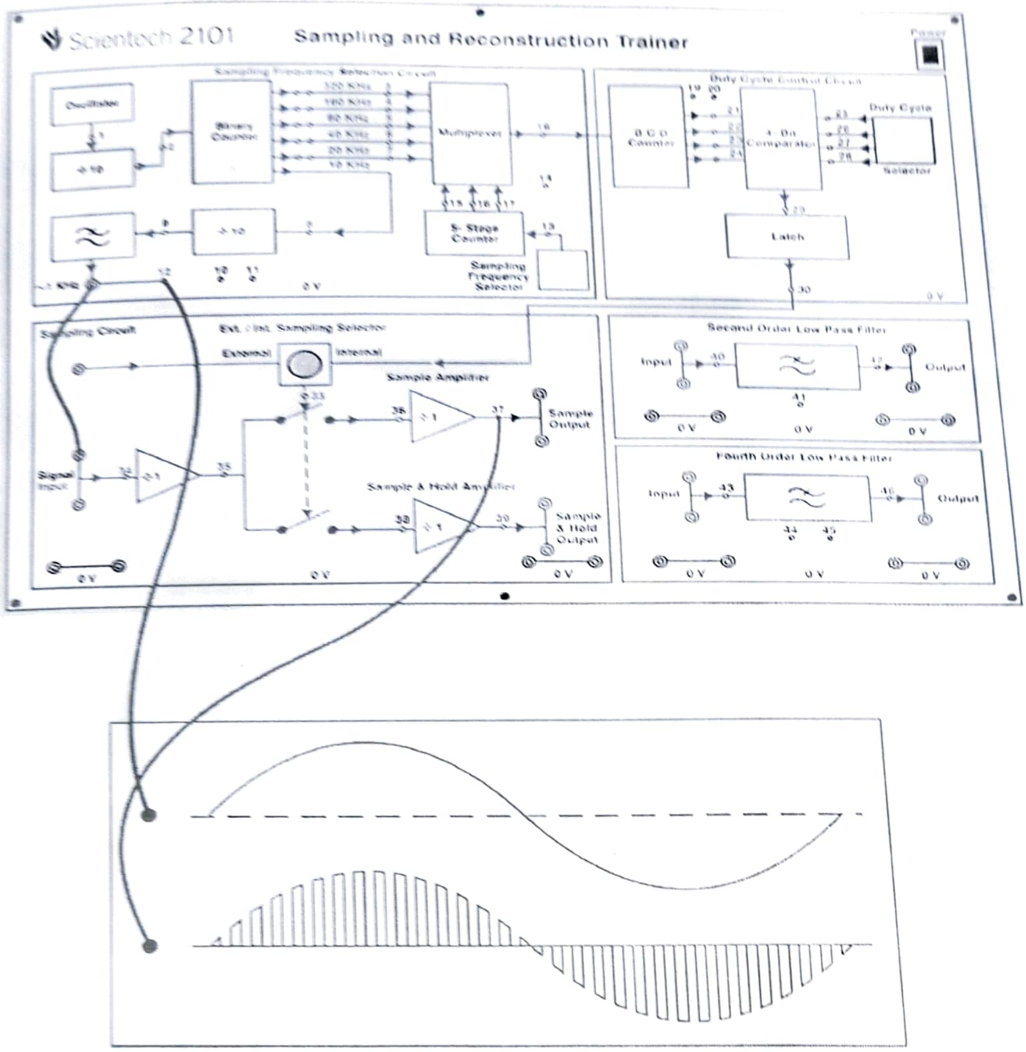
3. Repeat the above procedure with sample and hold circuitry. Observe that the output exhibits less distortion as compared to output in case of sample circuitry.

Conclusion:

As the order of LPF is increased at output, the recovered signal will be reconstructed more like the transmitted signal. To further verify this you can connect output of either sample amplifier or sample and hold amplifier to II order LPF Input and the output of II order LPF to the input of IV order LPF. The order of LPF is now VI.

Observe the output of IV order LPF and compare it with previously obtained waveforms.

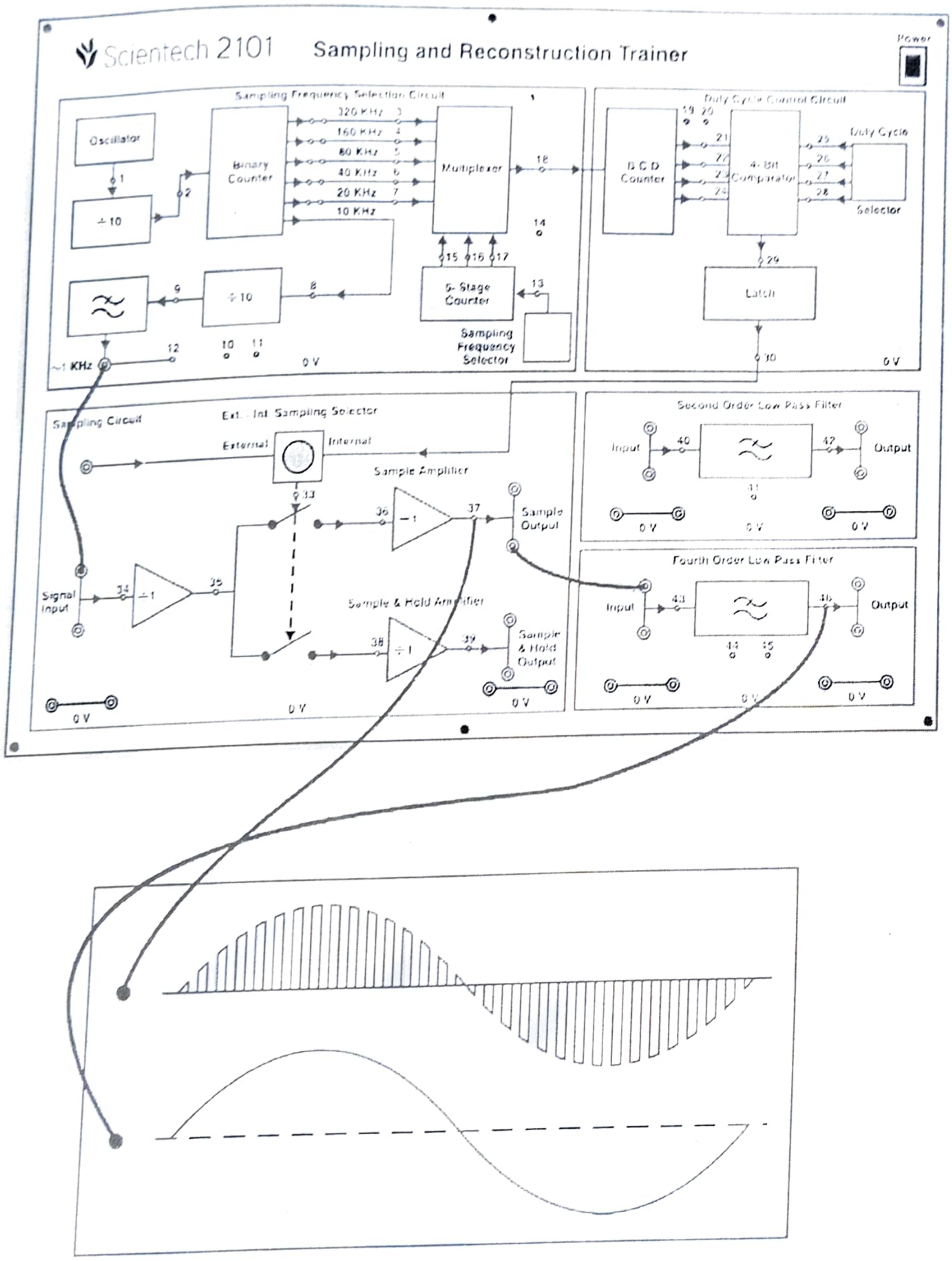
Signal Sampling Connection Diagram:



Signal Sampling

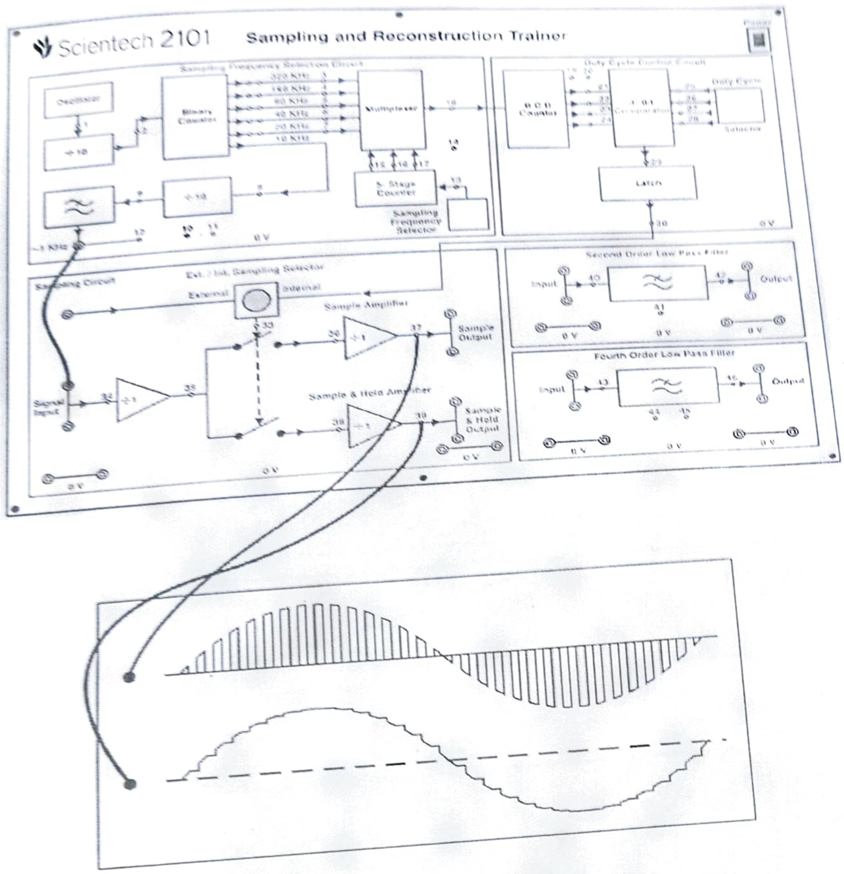
Figure 4

Signal Reconstruction Connection Diagram:



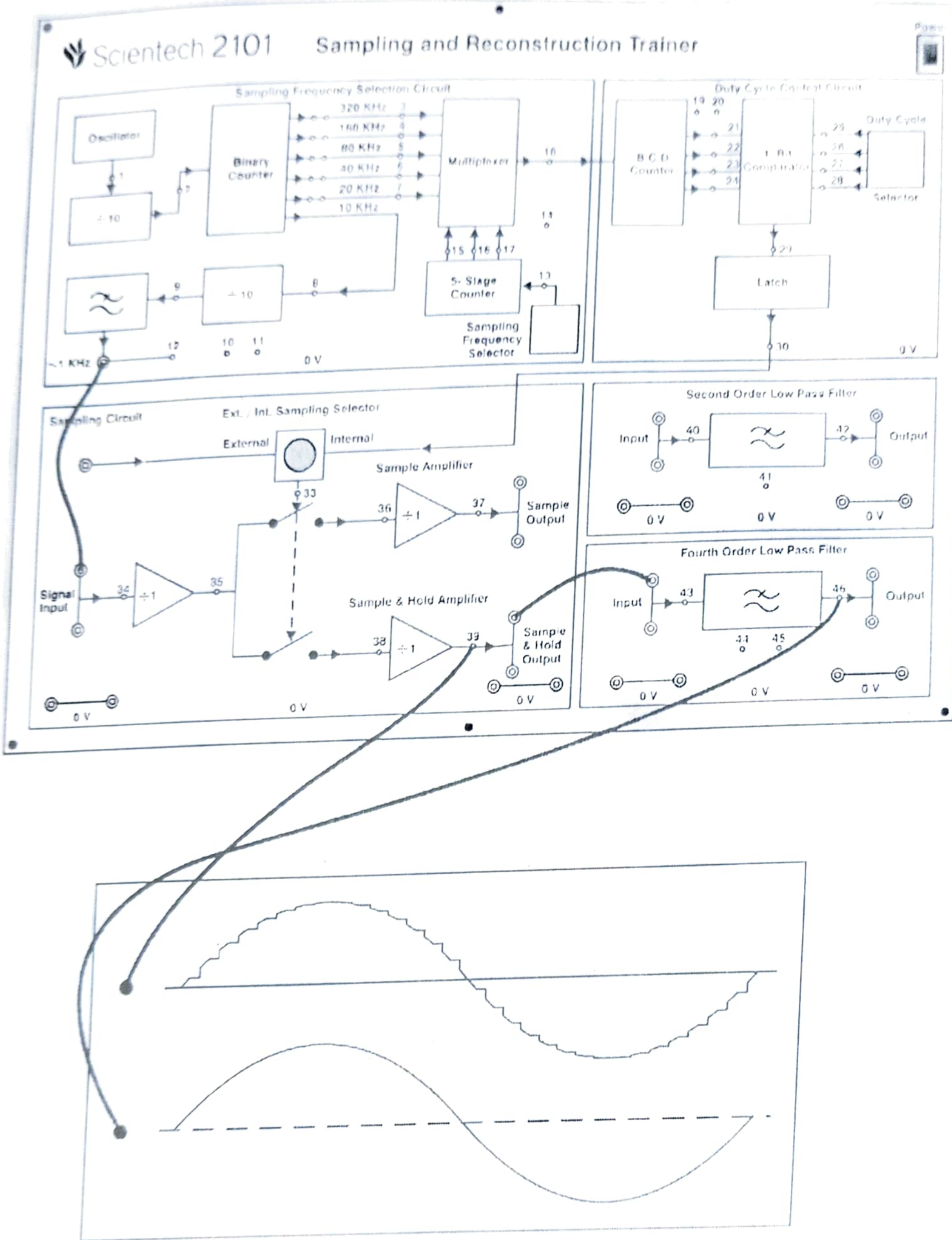
Signal Reconstruction
Figure 5

Connection Block Diagram: for Sample /Sample and Hold output



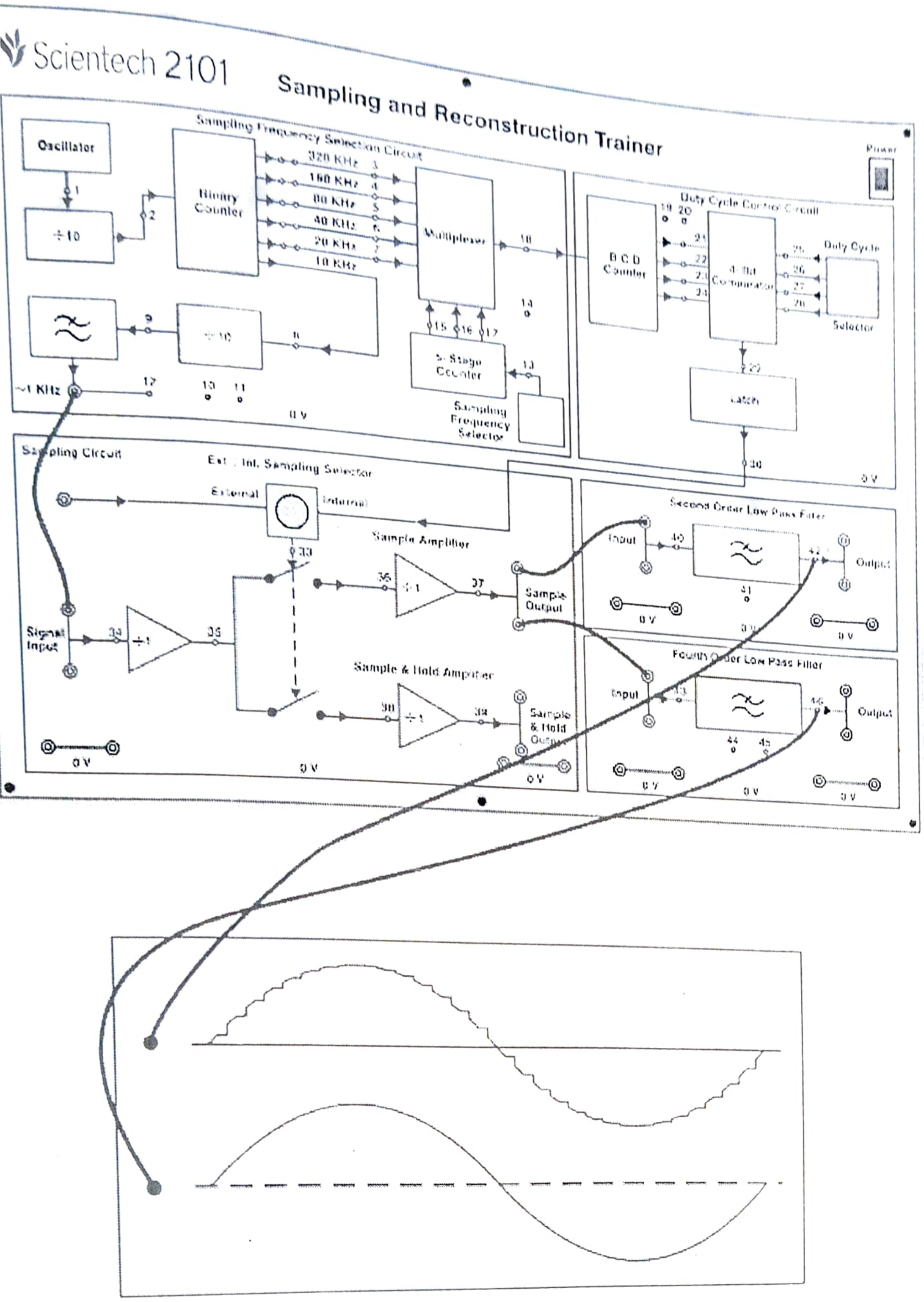
Sampled O/P and Sample & Hold O/P
Figure 6

Connection Block Diagram



Sample & Hold O/P
& LPF O/P
Figure 7.

...nection Block Diagram:



Low Pass Filter O/P
Figure 8

Frequently Asked Questions

1. **What do you mean by sampling?**

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

2. **What is sampling theorem?**

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

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

4. **What is Nyquist frequency?**

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

5. **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

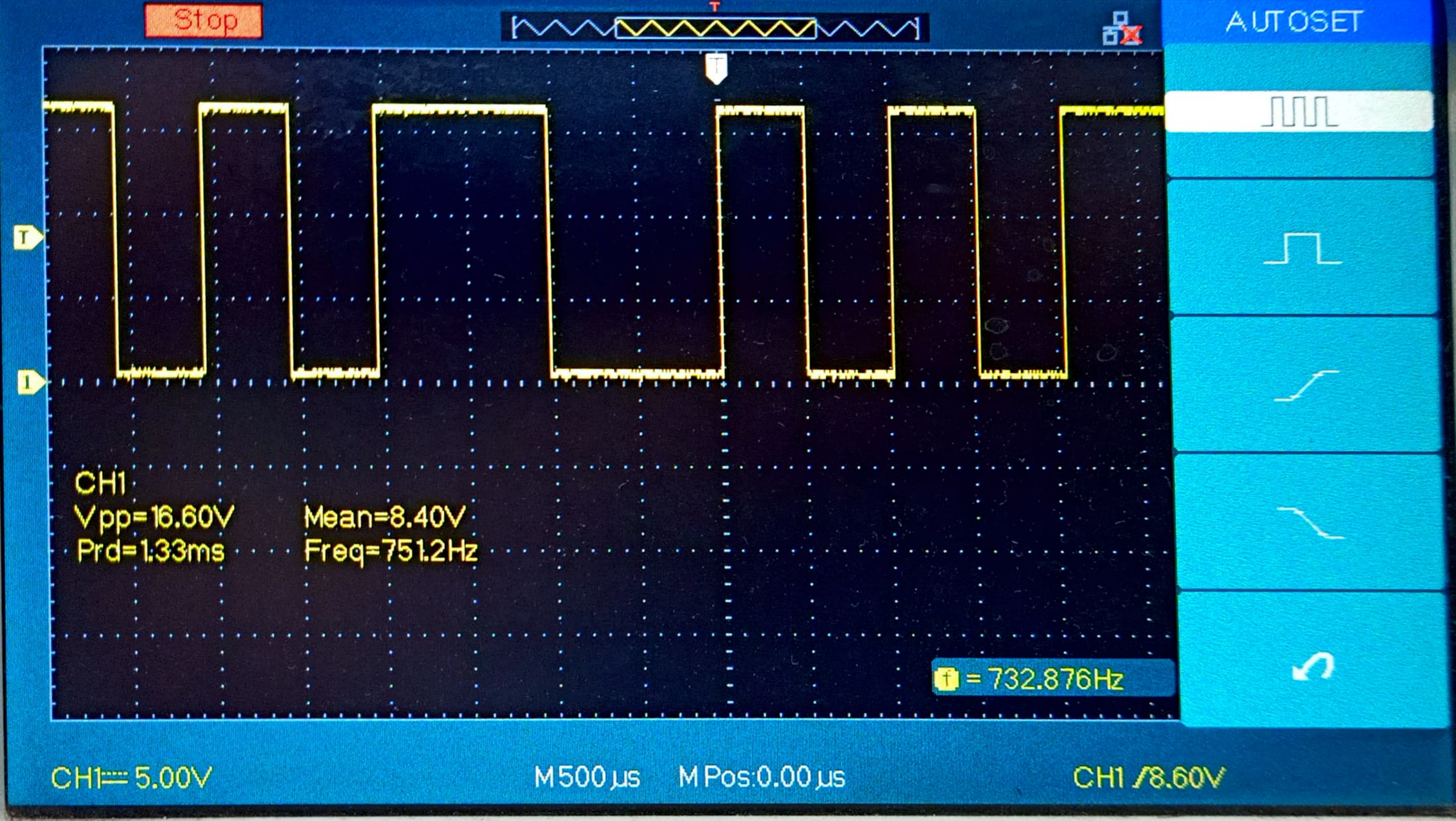
3. Flat top sampling

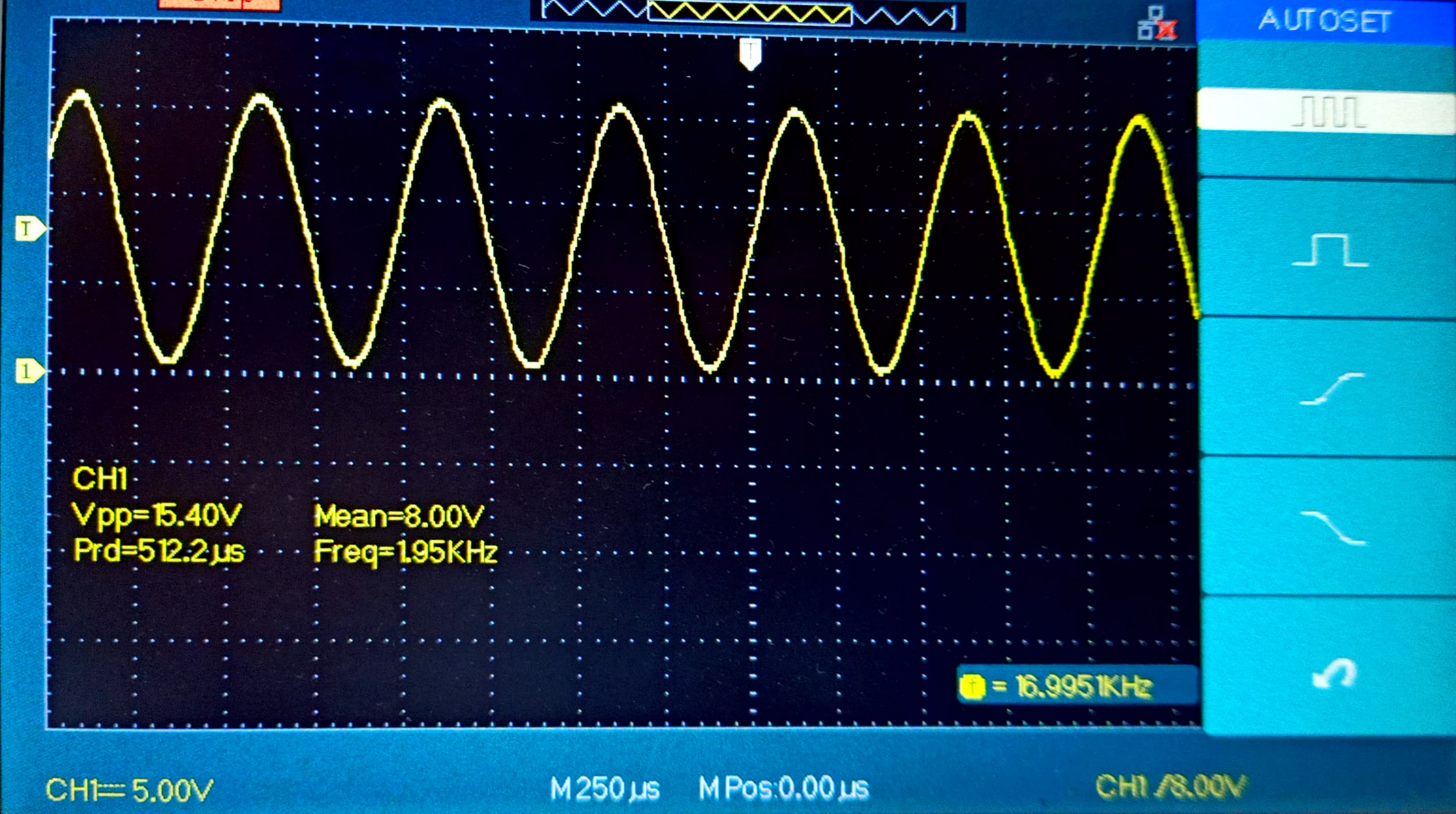
6. **What is under sampling?**

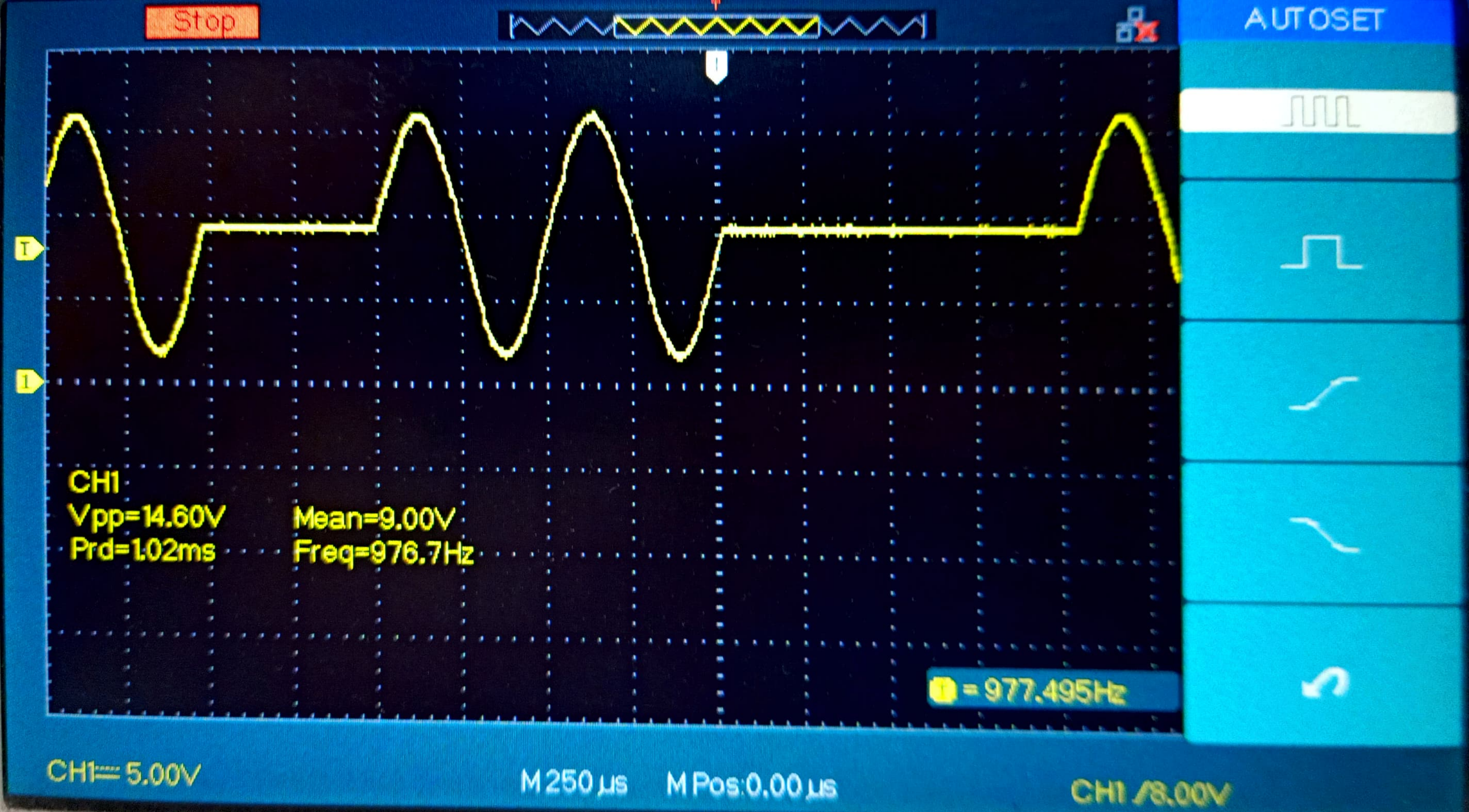
Ans: When the sampling rate is lower than or equal to the Nyquist rate, a conditiondefined as under sampling, it is impossible to rebuild the original signal according to the sampling theorem.

7. **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 intothe reconstructed signal and cause distortion of the signal Frequencies "fold" aroundhalf the sampling frequency. This type of signal distortion is called aliasing.


Stop
AUTOSET
CH1
Vpp=16.60V
Mean=8.40V
Prd=1.33ms
Freq=751.2Hz
= 732.876Hz
CH1 5.00V
M 500 µs
M Pos:0.00 µs
CH1 /8.60V


AUTOSET
CH1
Vpp=15.40V
Mean=8.00V
Prd=512.2µs
Freq=1.95KHz
= 16.9951KHz
CH1═5.00V
M 250µs
M Pos:0.00µs
CH1 /8.00V

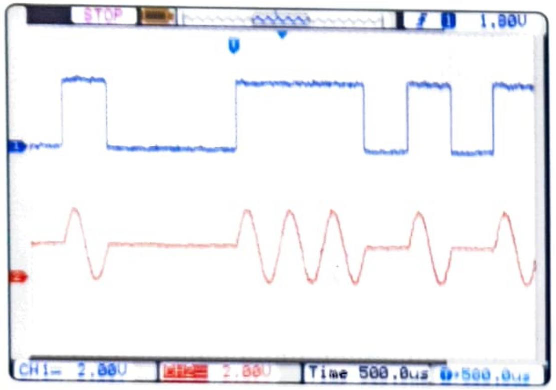

Stop
AUTOSET
CH1
Vpp=14.60V
Mean=9.00V
Prd=1.02ms
Freq=976.7Hz
= 977.495Hz
CH1 5.00V
M 250 µs
M Pos:0.00 µs
CH1 /8.00V


Stop
AUTOSET
CH2
Vpp=5.28A
Prd=1.02ms
Mean=1.72A
Freq=976.5Hz
CH1 5.00V
CH2 1.00A
M 1.00ms
M Pos:0.00µs
CH2 1.84A

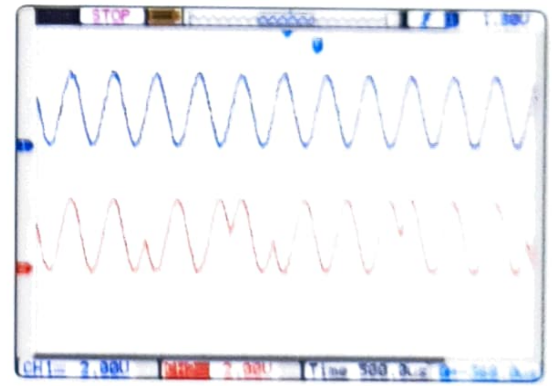

Stop
AUTOSET
CH2
Vpp=3.68A
Mean=1.68A
Prd=1.33ms
Freq=751.2Hz
= 732.876Hz
CH1 5.00V
CH2 2.00A
M 500 µs
M Pos:0.00 µs
CH2 1.76A

ASK, BPSK, DBPSK and FSK Modulator & Demodulator

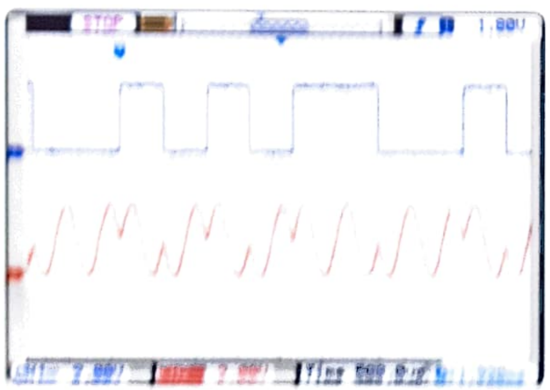
Amplitude-Shift Keying (ASK) or **On Off Keying (OOK)** is a form of modulation that represents digital data as variations in the amplitude of a carrier wave. The amplitude of an analog carrier signal varies in accordance with the bit stream (modulating signal), keeping frequency and phase constant.



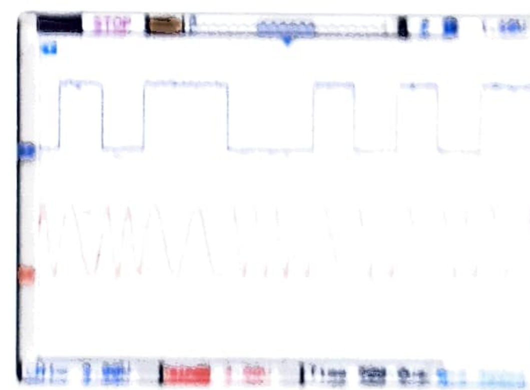
Differential Encoding is used to protect against this possibility. It is one of the simplest form of error protection coding done on a baseband sequence prior to modulation Then the phase of the carrier is varied to represent binary 1 or 0 of encoded data.

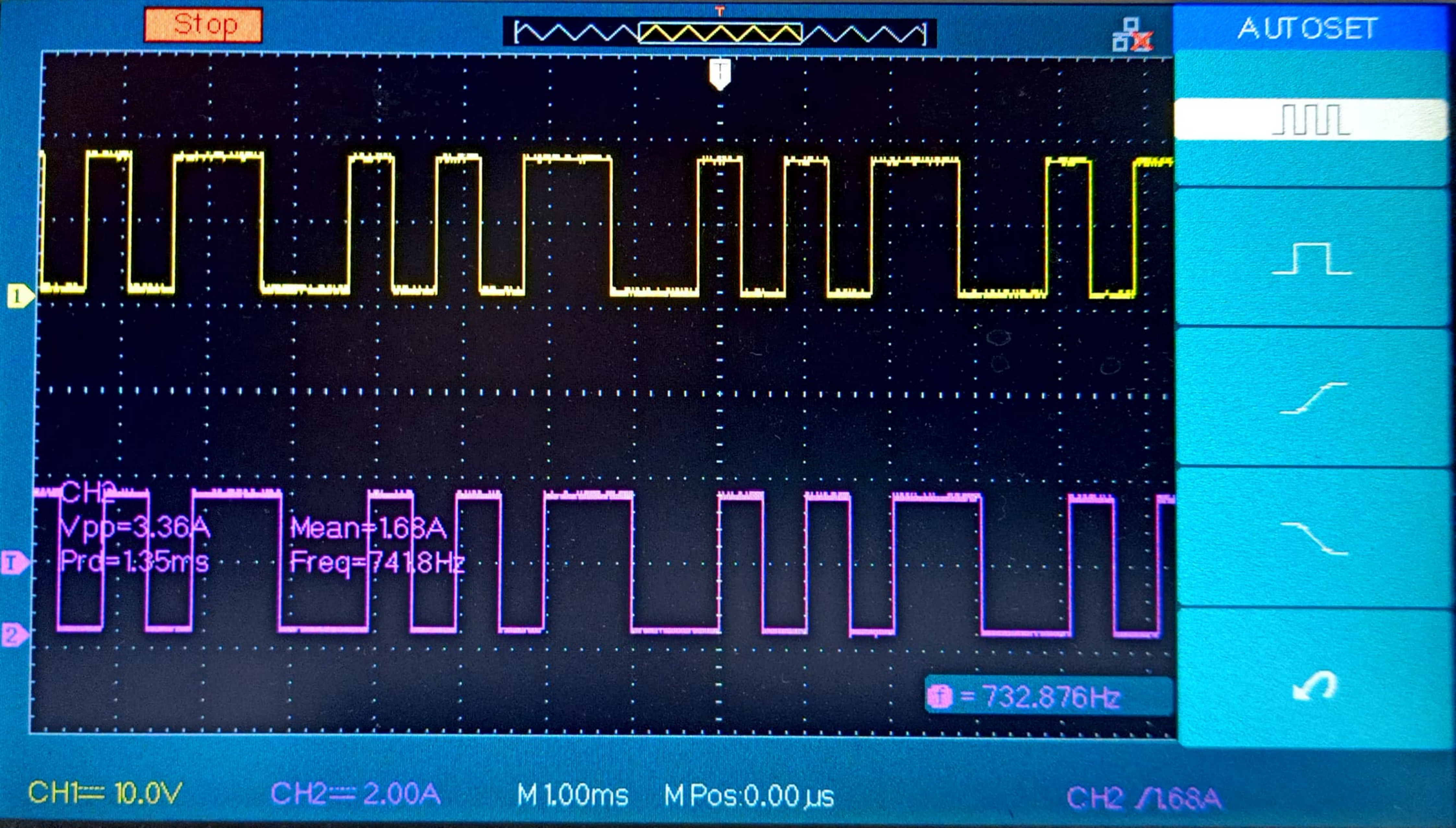


In **Binary Phase Shift Keying** modulation, the phase of the RF carrier is shifted 180 degrees in accordance with a digital bit stream. A "one" causes a phase transition, and a "zero" does not produce a transition.



In **Frequency Shift Keying** modulation, we change the frequency in response to information, one particular frequency for Logic 1 and another frequency for Logic 0.




Stop
AUTOSET
CH2
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Prd=1.35ms
Freq=741.8Hz
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CH1 10.0V
CH2 2.00A
M 1.00ms
M Pos:0.00µs
CH2 1.68A