Communication Engineering

Laboratory Manual

Premier University, Chittagong
Department of Electrical and Electronic Engineering

Premier University, Chittagong
Department of Electrical and Electronic Engineering

Course Code: EEE 310

Course Title: Communication Engineering Laboratory

Experiment number	Name of the experiment
01.	
02.	

Experiment No. 1

Double Sideband Suppressed Carrier AM Generation

Objective

- AM DSB modulation of audio signal using 455 kHz oscillator

Requirements

1. IT-4101 Trainer Board
2. 2 mm Patch Cords
3. Oscilloscope

Theory

In telecommunications, modulation is the process of conveying a message signal, for example, a digital bit stream or an analog audio signal, inside another signal that can be physically transmitted. In radio communications, cable TV systems, or the public switched telephone network, electrical signals can only be transferred over a limited passband frequency spectrum, with specific (non-zero) lower and upper cutoff frequencies. The primary motivation for modulation is to facilitate the transmission of the message signal over a communication channel with a prescribed passband. When the message or information-bearing signal is of an analog kind, we speak of *analog modulation* or *continuous wave modulation.*

A commonly used carrier is a sine wave, the source of which is physically independent of the source of the information-bearing signal. Modulation is done by varying the amplitude or angle of the sinusoidal carrier wave. On this basis, we can classify analog modulation into two broadly defined families: amplitude modulation and angle modulation.

In *amplitude modulation (AM),* the amplitude of the carrier is varied according to the information-bearing signal, as the name suggests. As the information signal increases in amplitude, the carrier wave is also made to increase the amplitude. Likewise, as the information signal decreases, then the carrier amplitude decreases.

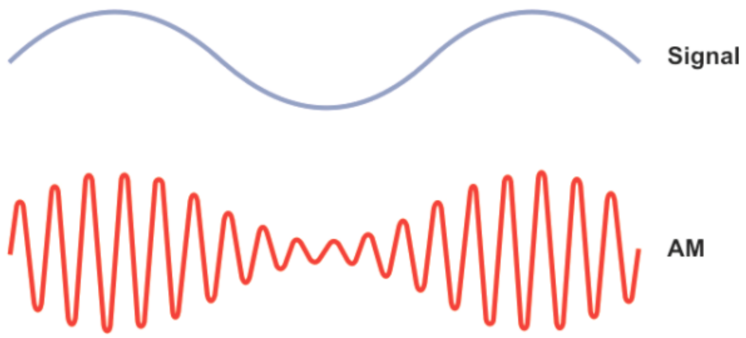

Fig-1

In Fig-1, we can see that the modulated carrier wave appears to ‘contain’ the information in its amplitude in some way.
The types of *amplitude modulation (AM):*

- Double-sideband with carrier (DSB-WC)
- Double-sideband suppressed carrier (DSB-SC)
- Single-sideband (SSB)

Procedure

1. Turn on the power of the IT-4101 module.
2. Make the *Interface Selector* Switch in the *Audio Generator Output* position.
3. Connect the oscilloscope probe to *Audio O/P (TP3)* and examine the low-frequency audio waveform. This audio frequency is a sine wave, which will be used as the modulating signal. The modulating frequency and amplitude can be varied by adjusting the *Audio Generator*’s *Amplitude* and *Frequency* preset. Set the amplitude and frequency to any convenient value. You can also listen to the audio signal by connecting the *Audio O/P (TP3)* with the *Audio Amplifier I/P (TP20)* and turning the *Speaker* ON. You can vary the volume by adjusting the Audio Amplifier Gain Adj.
4. Connect *455 kHz Local Oscillator O/P (TP5)* with the *Balanced Modulator-1 Carrier I/P (TP8)*. The local oscillator block generates the high-frequency carrier for the modulation process. The frequency is fixed at 455 kHz.
5. Now connect the *Audio O/P (TP3)* with the *Balanced Modulator-1 Audio I/P (TP7)*. The setup of the experiment should look like below:

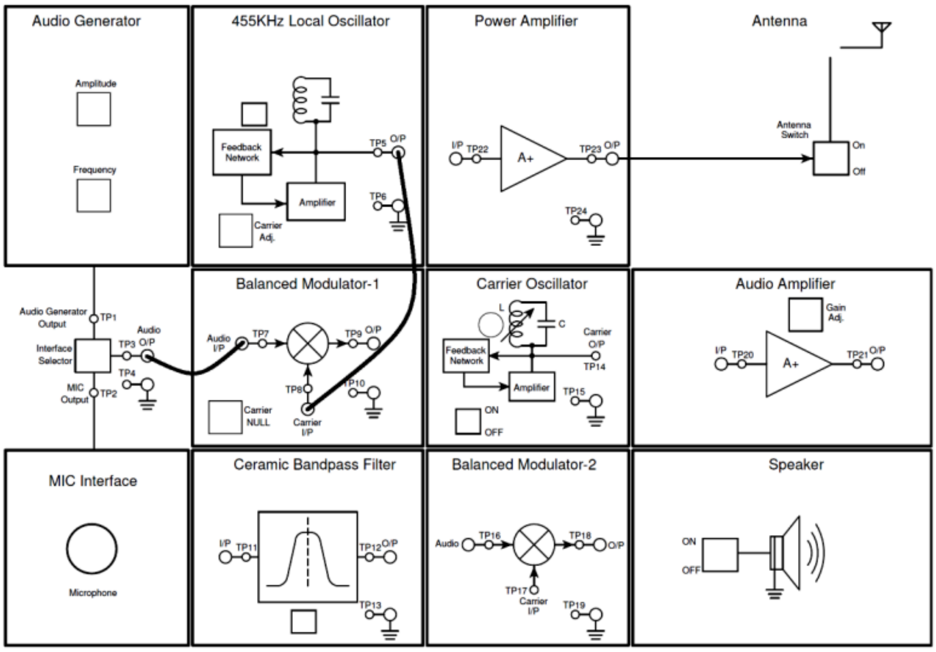

Fig-2

6. Connect the oscilloscope probe with *Balanced Modulator-1 O/P (TP9)* to observe the amplitude modulation. Adjust the *Carrier NULL* potentiometer to the center position until the waveform looks like this:

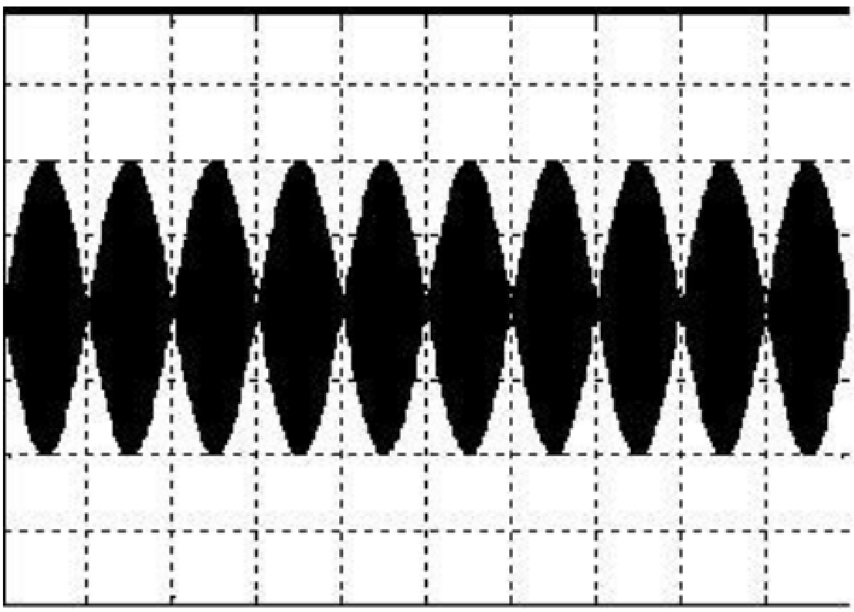

Fig-3

The output is a double sideband suppressed carrier AM waveform, which has been formed by amplitude modulating the 455 kHz carrier sine wave with the audio-frequency sine wave from the *Audio Oscillator.* Ideally, the frequency spectrum of this DSB-SC waveform should look like this:

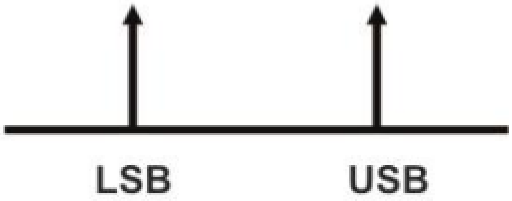

Fig-4

As we can see from Fig, the carrier component is absent in the DSB-SC modulated signal.

Below are the modulated wave (yellow) and its frequency spectrum (purple) seen on the digital oscilloscope. Here, we can see two sideband frequency peaks around the carrier frequency of 455 kHz.

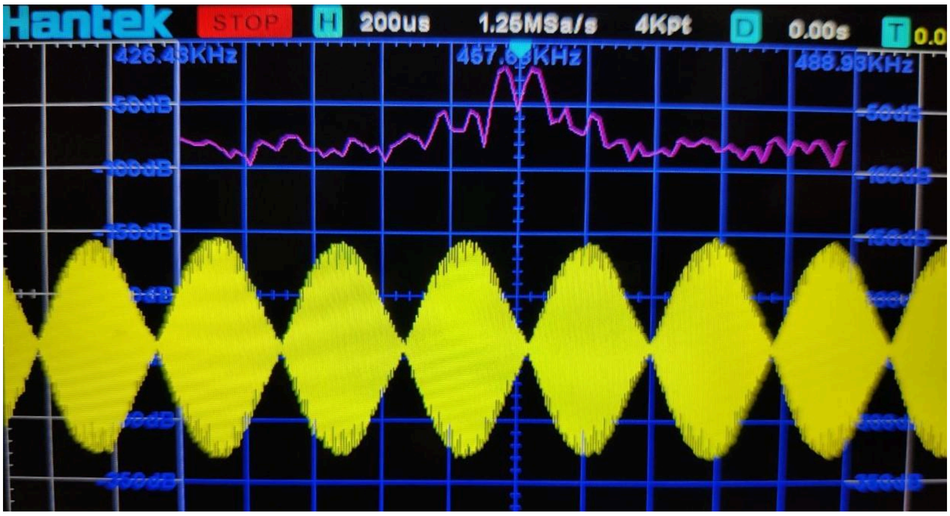

Fig-4

7. Examine the *Audio Output (TP3) (green)* together with the modulated waveform *(yellow)* on the oscilloscope.

Fig-5

Experimental Data

1. Measure the amplitude and frequency of the modulating signal and draw the signal.
2. Draw and observe the modulated wave and frequency spectrum on the oscilloscope.

Questions

1. Why is modulation necessary?
2. Why should the carrier waveform be a sinusoidal waveform?
3. Write down the mathematical formula for generating a DSB-SC wave.
4. What are sidebands?
5. What are the differences between DSB-SC and DSB-WC modulation? Which one is wasteful of transmitted power?

Experiment No. 2

Double Sideband Suppressed Carrier AM Detection

Objective

- To investigate the detection of double sideband amplitude modulated (AM) waveforms.

Requirements

1. IT-4101 and IT-4012 Trainer Boards
2. 2 mm Patch Cords
3. Oscilloscope

Theory

Demodulation is a crucial process in the reception of any amplitude-modulated signal. In DSB-WC modulation, the carrier wave is sent along with the sidebands. Owing to this, information pertaining to the message signal resides solely in the envelope of the modulated signal. We can harness this characteristic and employ a simple *envelope detection* technique for the demodulation of a DSB-WC-modulated signal.

In DSB-SC modulation, only the sidebands are present. The transmitted power is saved through the suppression of the carrier wave. But, the envelope of a DSB-SC modulated wave is different from the message signal, which means that simple demodulation using envelope detection is not a viable option for DSB-SC modulation.

The message signal m(t) can be recovered by multiplying the modulated signal *s(t)* with a locally generated sinusoidal wave and then low-pass filtering the product. It is assumed that the local oscillator signal is exactly coherent or synchronized, in both frequency and phase, with the carrier wave *c(t)* used in the product modulator used to generate the modulated wave *s(t)*. This method of modulation is known as *coherent detection* or *synchronous detection*.

Procedure

1. Position the *IT-4101* and *IT-4102* modules, with the *IT-4101* module on the left and a small gap between them.

2. Make the necessary connections for DSB modulation as given in experiment no. 1. Turn the *IT-4101* module ON and check the modulated waveform on the oscilloscope before proceeding.
3. Connect the *Balanced Modulator-1 O/P (TP9)* of the *IT-4101* module with the *Product Detector SSB I/P (TP9)* of the *IT-4102* module.
4. Connect the *455 kHz Local Oscillator O/P (TP5)* of *IT-4101* with the *Product Detector Carrier I/P (TP10)* of *IT-4102* module.
5. Connect the *Product Detector Audio O/P (TP11)* of the *IT-4102* module with the *Low Pass Filter LPF I/P (TP13)* of the *IT-4102* module.
6. Connect the *Low Pass Filter LPF O/P (TP14)* with the *Audio Amplifier Audio I/P (TP16)* of the *IT-4102* module.
7. Connect any two grounds of the *IT-4101* and *IT-4102* modules.
8. Observe the modulating audio signal and the detected audio signal *(Audio Amplifier Audio O/P TP17)* in two channels of the oscilloscope and check whether the two signals match. Change the amplitude and frequency of the modulating signal and observe the change in the detected signal. Below are the modulating signal (yellow) and the demodulated signal (green) as seen on the oscilloscope.

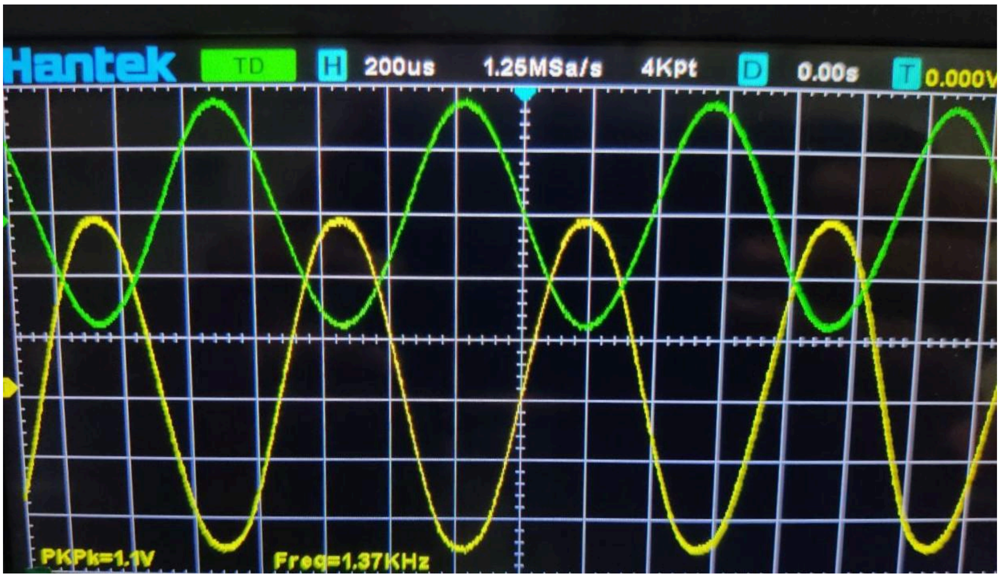

Fig-1

Experimental Data

1. Measure the amplitude and frequency of the demodulated signal and draw the signal.
2. Observe the *Product Detector Audio O/P (TP11)* and draw the signal.

Questions

1. Show mathematically how we can retrieve the message signal *m(t)* from the modulated signal *s(t)*.
2. Draw the frequency spectrum of the *Product Detector* output.

Experiment No. 3

Frequency Modulation

Objective

- To observe the frequency modulation using the *Varactor Modulator*

Requirements

1. IT-4103 Trainer Board
2. 2mm Patch Cords
3. Oscilloscope

Theory

Frequency Modulation (FM) conveys information over the carrier wave by varying its frequency. In analog applications, the instantaneous frequency of the carrier is directly proportional to the instantaneous value of the input signal. This modulation process does not affect the amplitude.

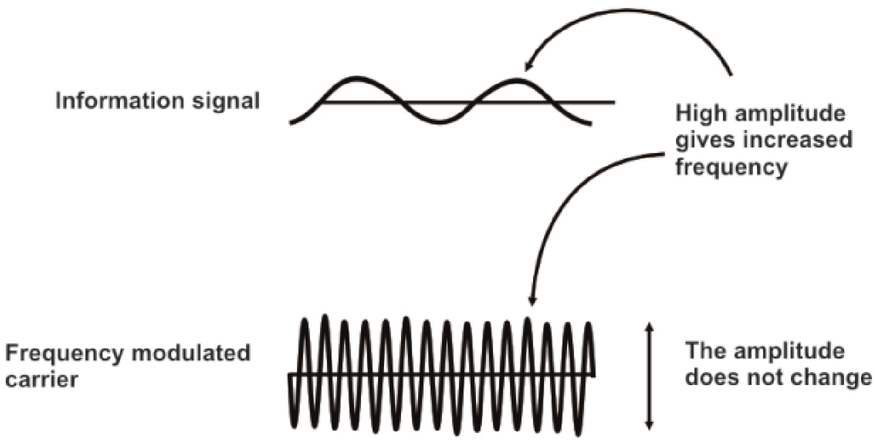

Fig-1

There are some distinct advantages of frequency modulation in communication systems. One advantage is that frequency modulation has no effect on the amplitude of the carrier wave. The demodulator only needs to observe the change in the frequency and can ignore any changes in amplitude. Electrical noise that shifts the amplitude, therefore, has no or little effect on an FM communication system.

Varactor Modulator: All FM transmitters function in much the same way. They include an RF oscillator to generate the carrier, and these oscillators employ a parallel-tuned circuit to determine the frequency of operation. A parallel resonant tank circuit is used, where the value of the inductance and capacitance determine the frequency of operation. The frequency of resonance is given by:

$$f = \frac{1}{2\pi\sqrt{LC}}$$

where L is the inductance and C is the capacitance. If we can vary the value of the capacitance according to the message signal, we can control the carrier frequency. A *Varactor Diode* is a way to do that.

A *varactor diode* is a semiconductor diode that is designed to behave as a voltage-controlled capacitor. When a semiconductor diode is reverse biased, no current flows and a junction capacitance is associated with it. We can control the width of the depletion region by the applied reverse bias voltage, which in turn controls the capacitance. If the message signal is applied to the varactor diode, the capacitance will, therefore, be varied accordingly, and the frequency will change.

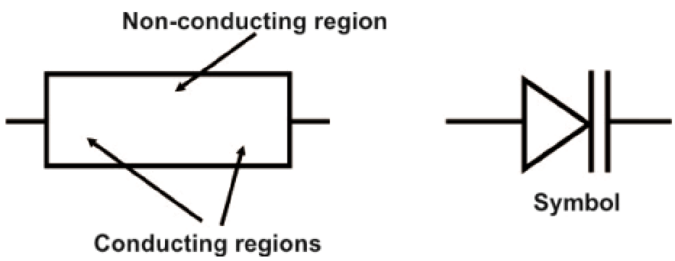

Fig-2

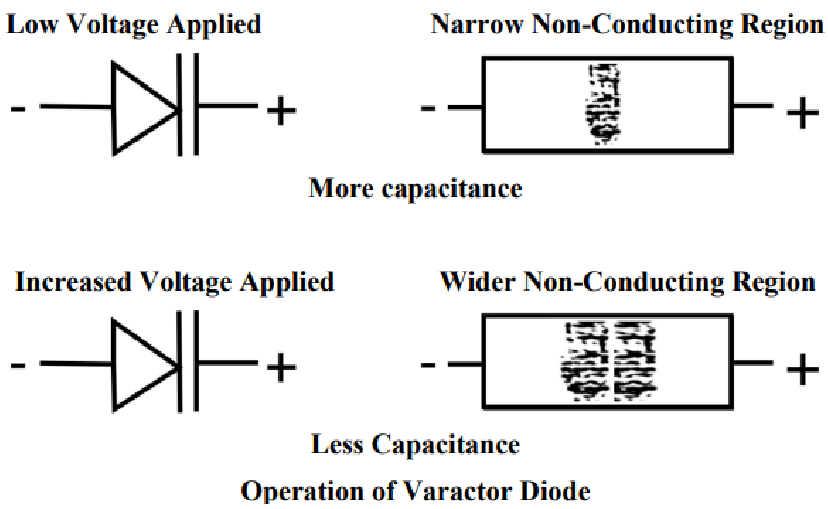

Fig-3

Procedures

1. Turn on the *IT-4103* module.

2. Connect *Audio Generator O/P (TP1)* to channel 1 of the Oscilloscope. This is the modulating wave. Check the output and adjust the *Frequency* knob to make the frequency around 1 kHz.

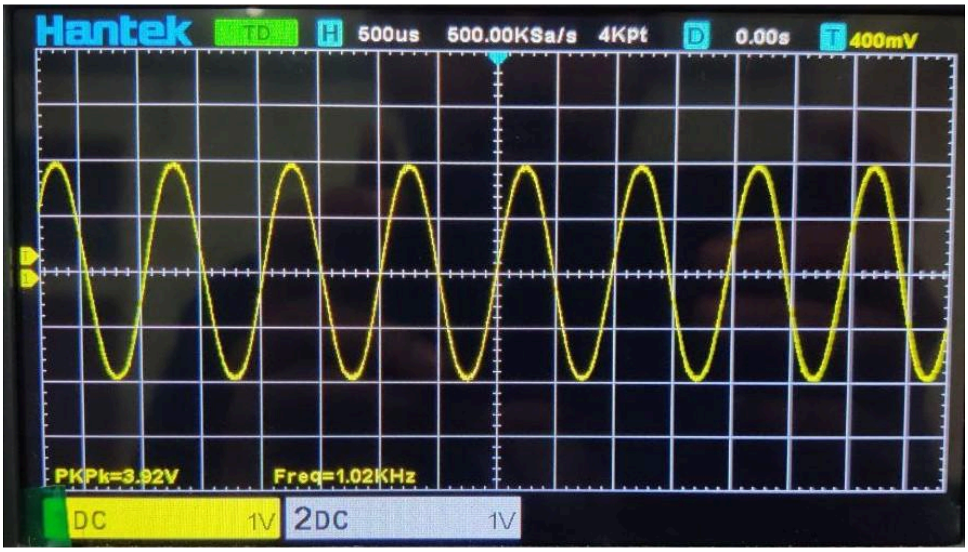

Fig-4: Message Signal

3. Connect the *Varactor Modulator O/P (TP9)* to channel 2 of the Oscilloscope. Observe the signal and calculate the frequency.
4. Connect the *Audio Generator O/P (TP1)* with the *Varactor Modulator I/P (TP8).* The connection should look like below:

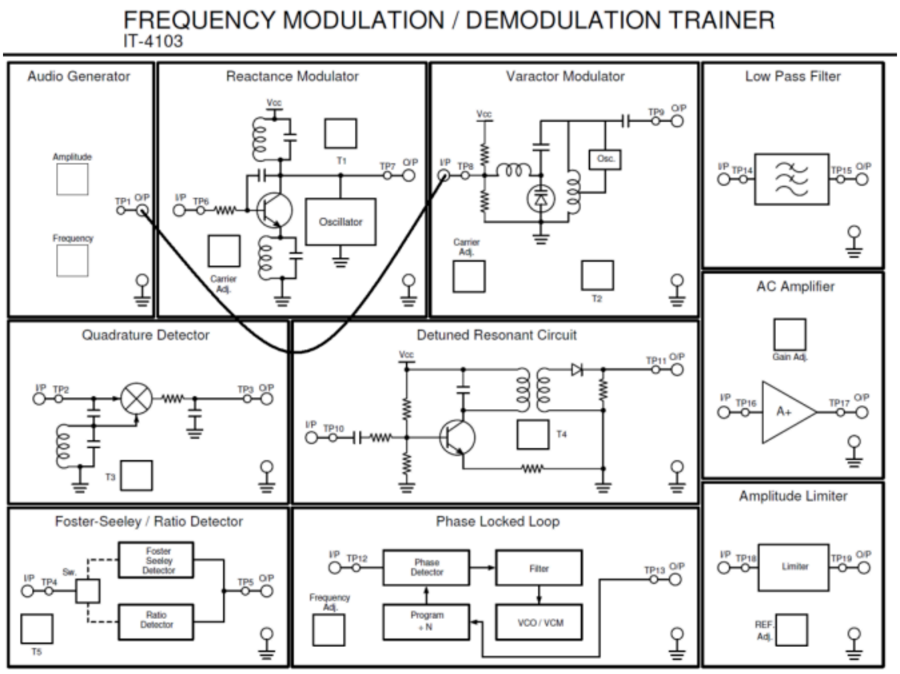

Fig-4

5. Observe the *Varactor Modulator O/P (TP9)* on the oscilloscope. Vary the audio signal's amplitude and observe the frequency variation.

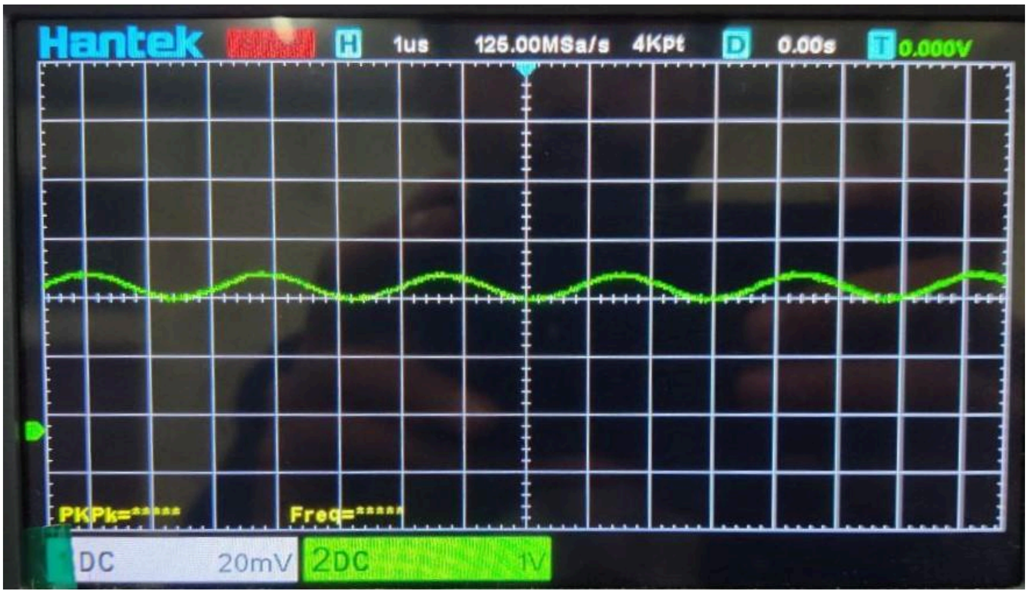

Fig-5: FM wave

Experimental Data

1. Measure the amplitude and frequency of the audio signal and draw the signal.
2. Measure the amplitude and frequency of the carrier signal and draw the signal.
3. Observe the frequency-modulated signal on the oscilloscope and draw the signal.

Questions

1. Write down the mathematical formula of frequency modulated wave.
2. What’s the frequency sensitivity factor? Why is the frequency deviation negligible, as observed in the experiments?

Experiment No. 4

Frequency Demodulation

Objective

- To observe frequency demodulation using the *Phase-Locked Loop*

Requirements

1. IT-4103 Trainer Board
2. 2 mm Patch Cords
3. Oscilloscope

Theory

Frequency demodulation is the process by means of which the original message signal is recovered from an incoming FM wave. For frequency demodulation, we need a device whose output amplitude is sensitive to variations in the instantaneous frequency of the input FM wave in a linear manner.
Phase-Locked Loop: A phase-locked loop is a feedback system whose operation is closely linked to frequency modulation. It retrieves the message signal indirectly.

Procedure

1. Turn on the *IT-4103* module.
2. Make the necessary connections for frequency modulation as given in experiment no. 3. Check the modulated waveform on the oscilloscope before proceeding.
3. Connect the *Varactor Modulator O/P (TP9)* with the *Phase-Locked Loop I/P (TP12)*. Observe the *Phase-Locked Loop O/P (TP13)* on the oscilloscope.

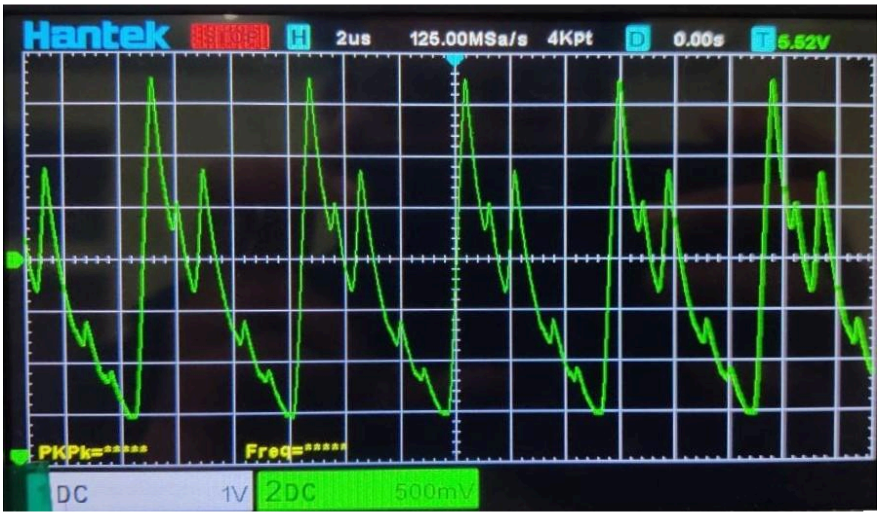

Fig-1: Phase-Locked Loop Output

4. Connect the *Phase-Locked Loop O/P (TP13)* with the *Low Pass Filter I/P (TP14)*.
5. Connect the *Low Pass Filter O/P (TP 15)* with the *AC Amplifier I/P (TP15)*. At this point, the connection should look like below:

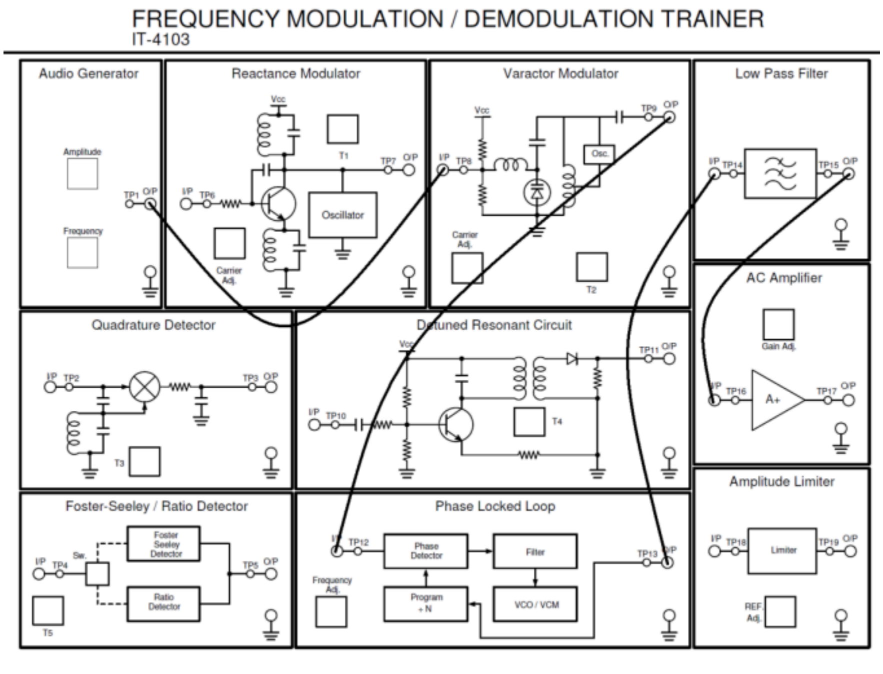

Fig-1

6. The *AC Amplifier O/P (TP17)* is the demodulated signal. Observe this signal on the oscilloscope in one channel and the *Audio Generator O/P (TP1)* on the other and check whether the two signals match.

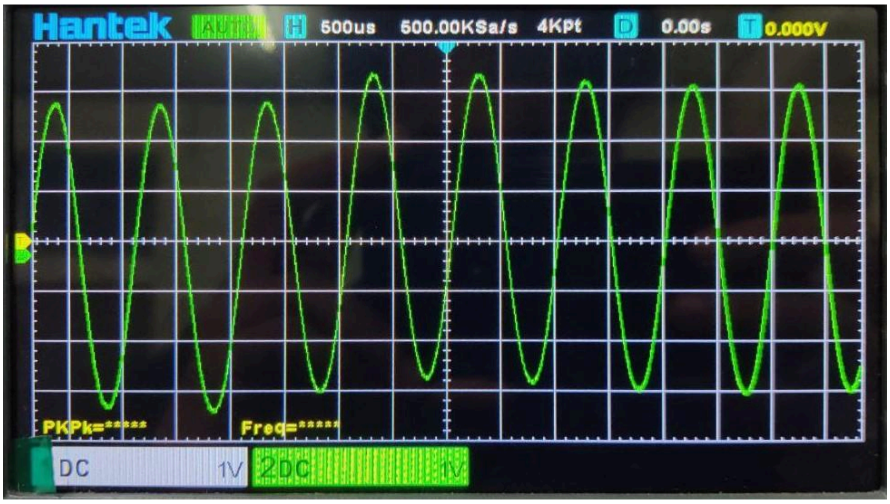

Fig-2: Demodulated wave

Experimental Data

1. Measure the amplitude and the frequency of the frequency demodulated signal and draw the signal.
2. Observe the *Phase-Locked Loop O/P (TP13)* on the oscilloscope and draw the signal.

Questions

1. Sketch the block diagram of the phase-locked loop.
2. What conditions must be satisfied when the control signal of the *Voltage-Controlled Oscillator* of the phase-locked loop is zero?

Experiment No. 5

Study of Signal Sampling and Reconstruction Techniques

Objective

- To observe pulse amplitude modulation and the effect of sampling frequency on signal reconstruction

Requirements

1. IT-4201 Trainer Board
2. 2 mm Patch Cord
3. Oscilloscope

Theory

Pulse Modulation is another form of modulation. In analog pulse modulation, a periodic pulse train is used as a carrier wave, and some characteristics of each pulse (amplitude, duration, or position) vary continuously in accordance with the corresponding *sample* value of the message signal.
Sampling a signal in the time domain makes the spectrum of the signal periodic in the frequency domain. Through the use of the sampling process, an analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time. Specifically, in the *Pulse Amplitude Modulated (PAM)* system, a train of pulses corresponding to the samples of each signal is modulated in amplitude according to the amplitude of the message.
Here, an information signal is sent through an ideal switch operated by a control signal. When the switch is open, the voltage is zero; when it is closed, the output voltage is equal to the instantaneous signal voltage. The pulse's duty cycle (sample width) depends on how long the switch is closed.

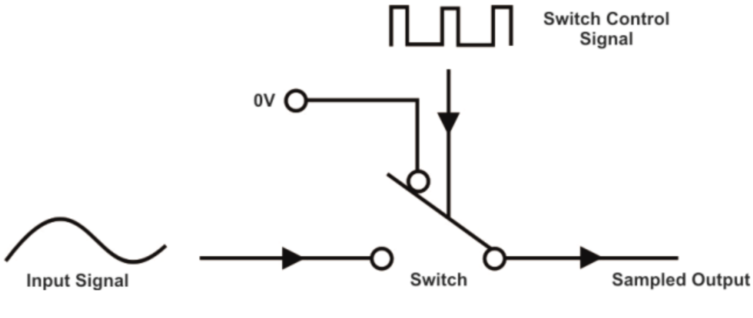

Fig 1: Basic Sampling Process

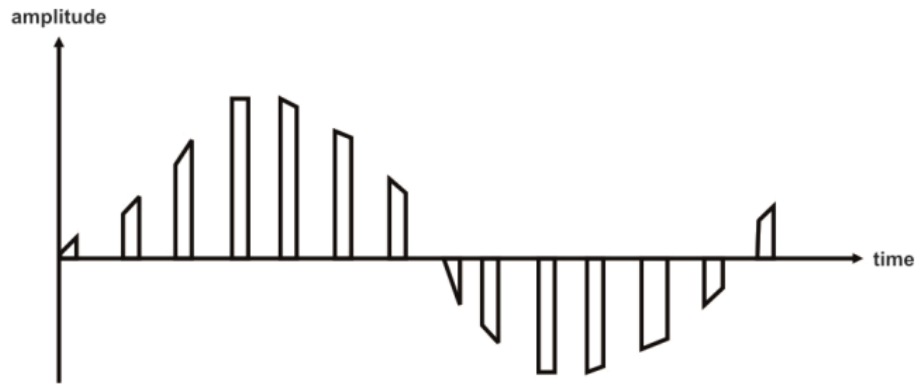

Fig 2: Sampled Output

For sampling, we have to keep in mind the sampling principle. It states that the intelligence signal can be reconstructed by filtering when the sampling signal frequency *(F_s)* is greater than twice the maximum intelligence signal frequency *(F_m)*. $F_s=2F_m$ is called the *Nyquist Rate.* The frequency response of the low-pass filter on the receiver side must be capable of passing the maximum intelligence signal to reconstruct the intelligence signal frequency while rejecting the sideband frequencies of the sampling signal to reconstruct the intelligence free of distortion. If the signal is sampled at a rate lower than the Nyquist criterion, then there is an overlap between the information signal and the sidebands of the harmonics. Thus, the higher and the lower frequency components get mixed and cause unwanted signals to appear at the demodulator output. This phenomenon is termed *aliasing* or *fold-over distortion.*

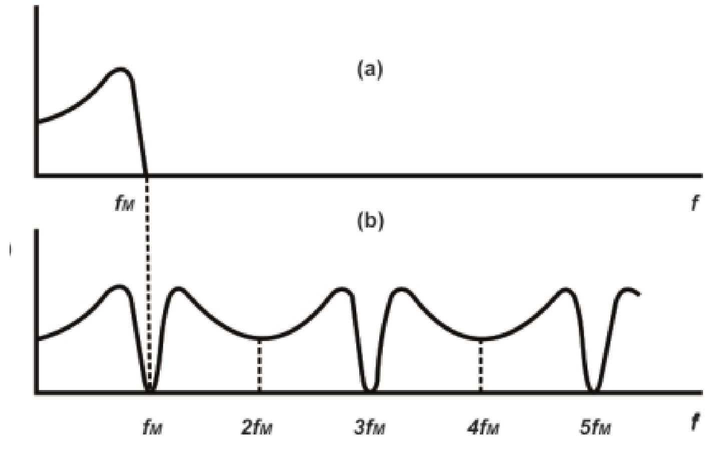

Fig 3: Effect of sampling at the Nyquist rate on the frequency response

Procedure

1. Turn On the power of the *IT-4201* module.
2. Connect the oscilloscope with the *Audio Generator Audio O/P (TP1).* Adjust the *Frequency* knob to approximately 1kHz. Adjust the *Amplitude*

knob in the center position. The signal, as seen on the oscilloscope, is shown below:

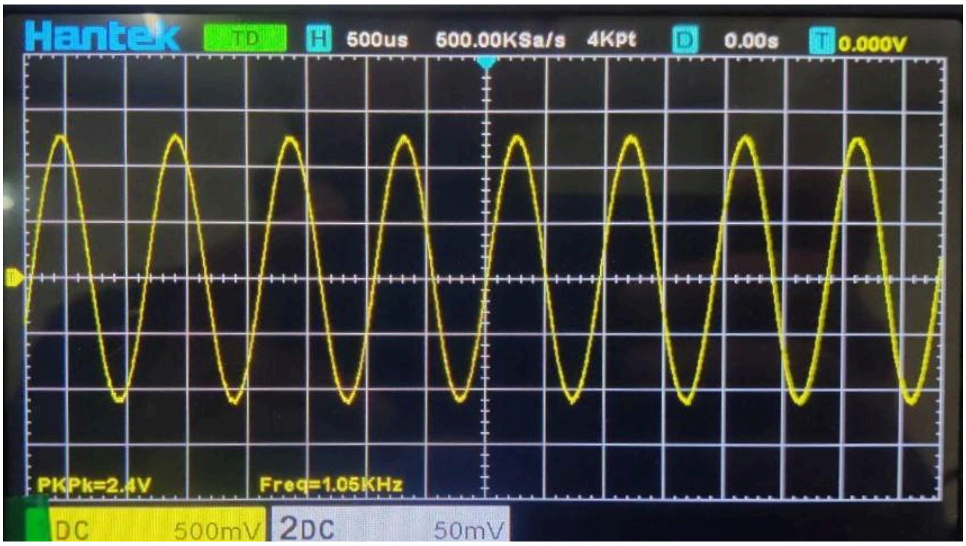

Fig-4: Audio Signal

3. Connect the oscilloscope probe to *Clock Generator O/P (TP2)* and verify a clock of frequency 4.096 MHz.

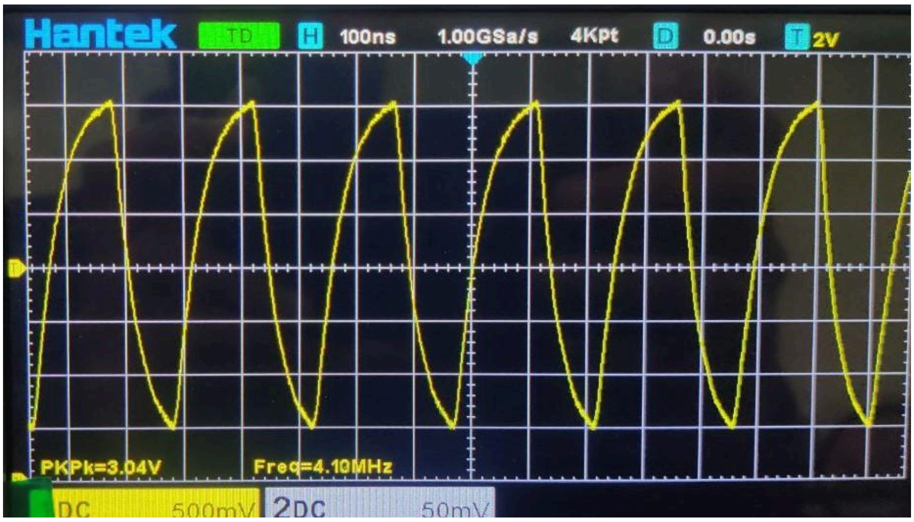

Fig-5: Clock Generator O/P

4. Connect the *Clock Generator O/P (TP2)* with the *Channel Selection I/P (TP3).* Select the 16 kHz channel by selecting the switch *SW1*. *SW1* is used to select the frequency to be output through *MUX.* When you press *SW1*, the channel selection LED will indicate the channel selected for output and the corresponding frequency LED will also be ON. You can change the frequency of the pulse train by pressing the *SW1* switch.
5. Observe the *Channel Selection O/P (TP12)* on the oscilloscope. Check the frequency of the wave.

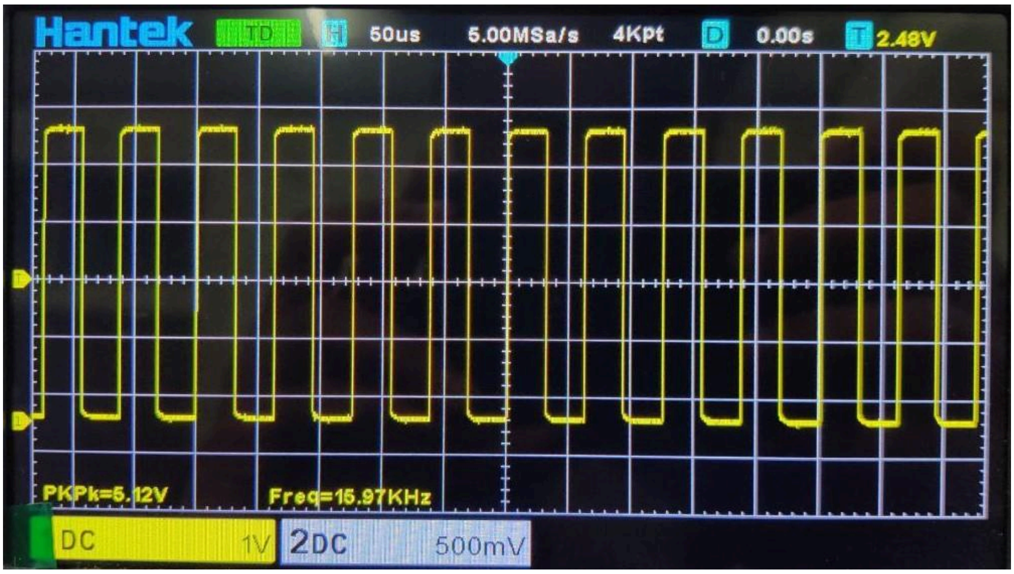

Fig-6: *Channel Selection O/P*

6. Connect *the Channel Selection O/P (TP12) with the Sampling Pulse I/P (TP14). Also,* connect the *Audio Generator Audio O/P (TP1)* with the *Sampling Signal I/P (TP13)*.
7. Observe *Sampling Sample O/P (TP15)* on the oscilloscope.

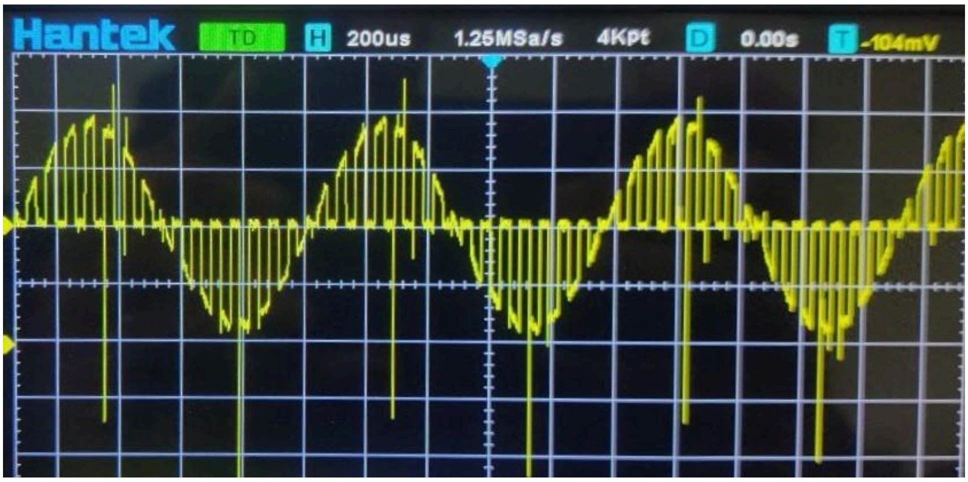

Fig-7: *Sampled Signal Output at 16 kHz*

8. Change the sampling frequency to 8 kHz, 4 kHz, and 2 kHz by successive presses of the frequency selector switch *SW1*. Observe the output on the oscilloscope. It can be observed that as we decrease the sampling frequency, the sampled signal gradually 'loses' its sinusoidal appearance.

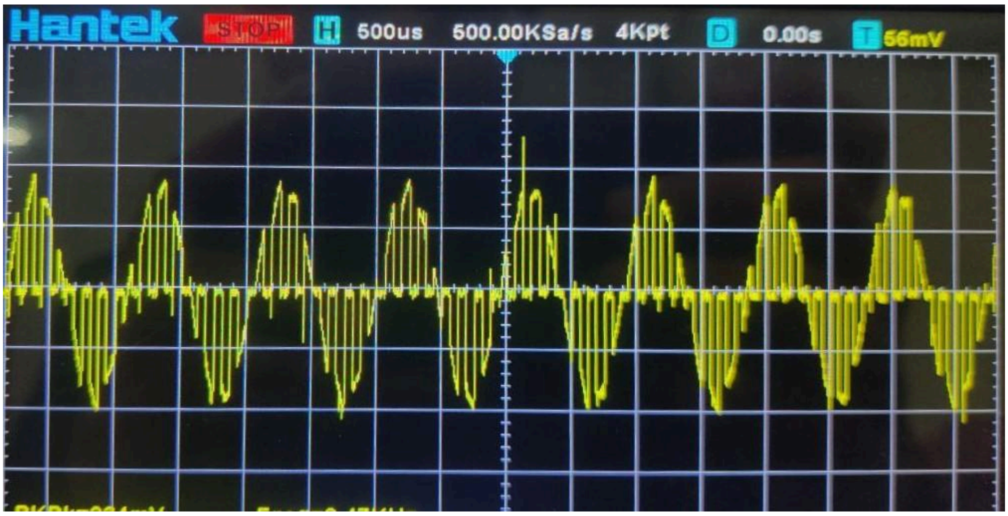

Fig-8: *Sampled Signal Output at 8 kHz*

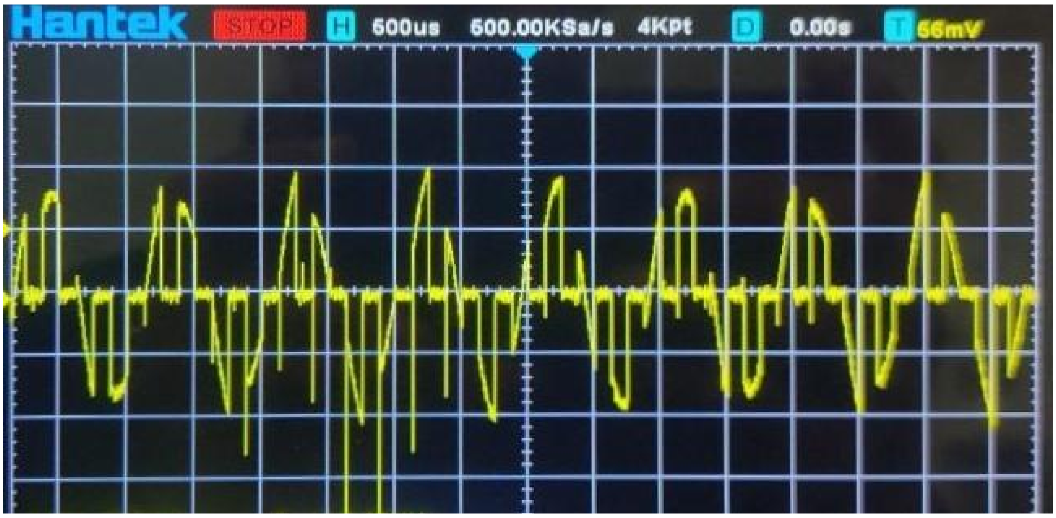

Fig-8: *Sampled Signal Output at 4 kHz*

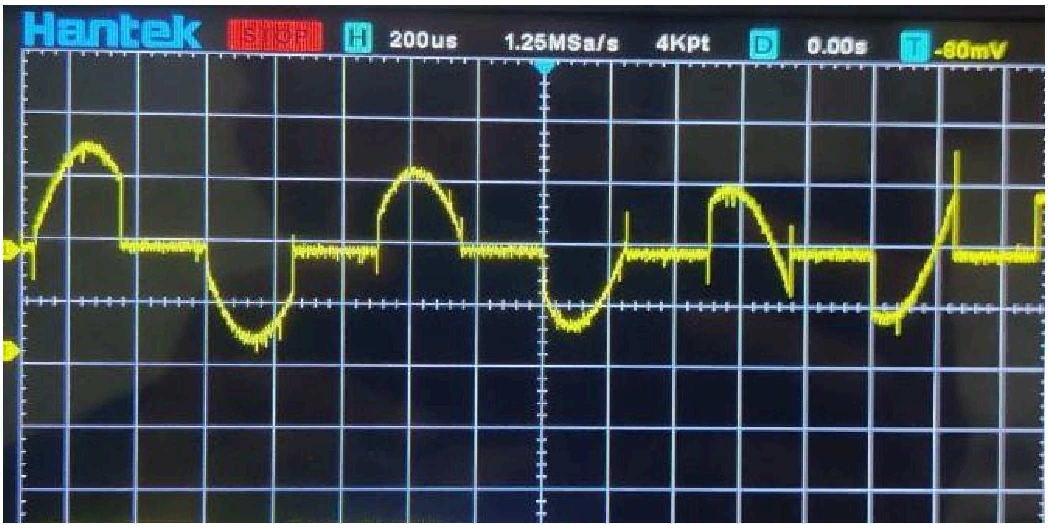

Fig-8: *Sampled Signal Output at 2 kHz*

9. You can also observe the pulses from the *Channel Selection Block (TP12)* along with the *Sample O/P (TP15)* on the oscilloscope to clearly visualize the PAM process as shown below:

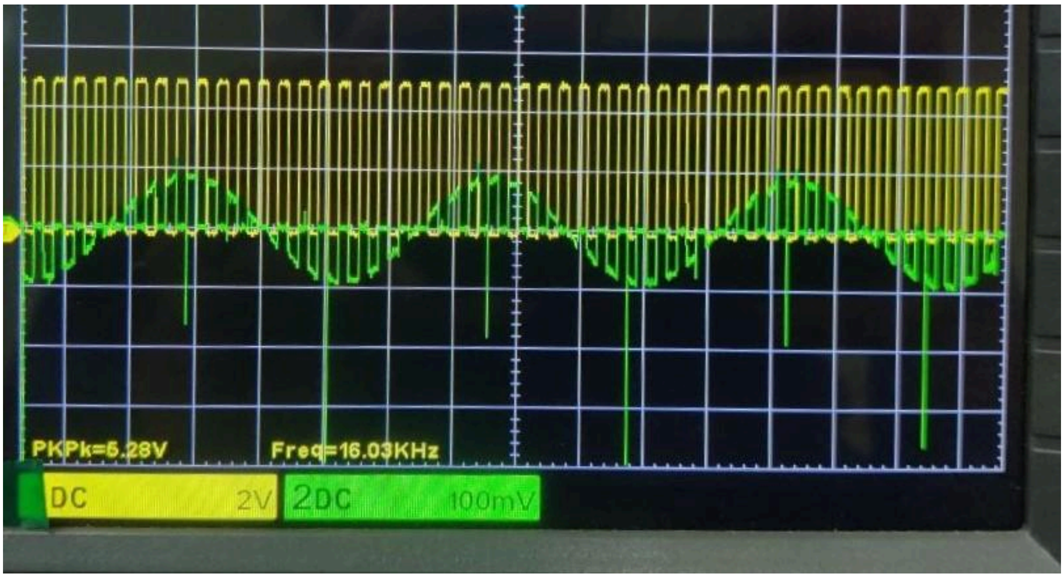

Fig-9: *Sampled Signal Output at 16 kHz (Green) along with sampling pulses (yellow)*

10. Now, we shall start reconstructing the sampled signal. We know that the sampling process generates high-frequency periodic copies of the spectrum of the message signal, the period of which depends on the sampling frequency. So, for reconstruction, we need to attenuate the high-frequency copies, for which we need low-pass filtering. Here, using the *IT-4201 Trainer Board,* we shall do two levels of filtering. First, connect the *Sample O/P (TP15)* with the *LowPass Filter-1 I/P (TP31).*
11. Connect the *LowPass Filter-1 O/P (TP32)* with the *AC Amplifier-1 I/P (TP27)* to increase the signal level. Adjust the gain as necessary.
12. Connect the *AC Amplifier-1 O/P (TP28)* with the *LowPass Filter-2 I/P (TP33).*
13. Connect the *LowPass Filter-2 O/P (TP34)* with the *AC Amplifier-2 I/P (TP29)* for the second level of gain adjustment.
14. Observe the reconstructed signal on the oscilloscope at AC Amplifier-2 O/P (TP30). Change the sampling frequency by pressing the *Channel Selection Switch SW1* and observe its effect on the reconstructed signal.

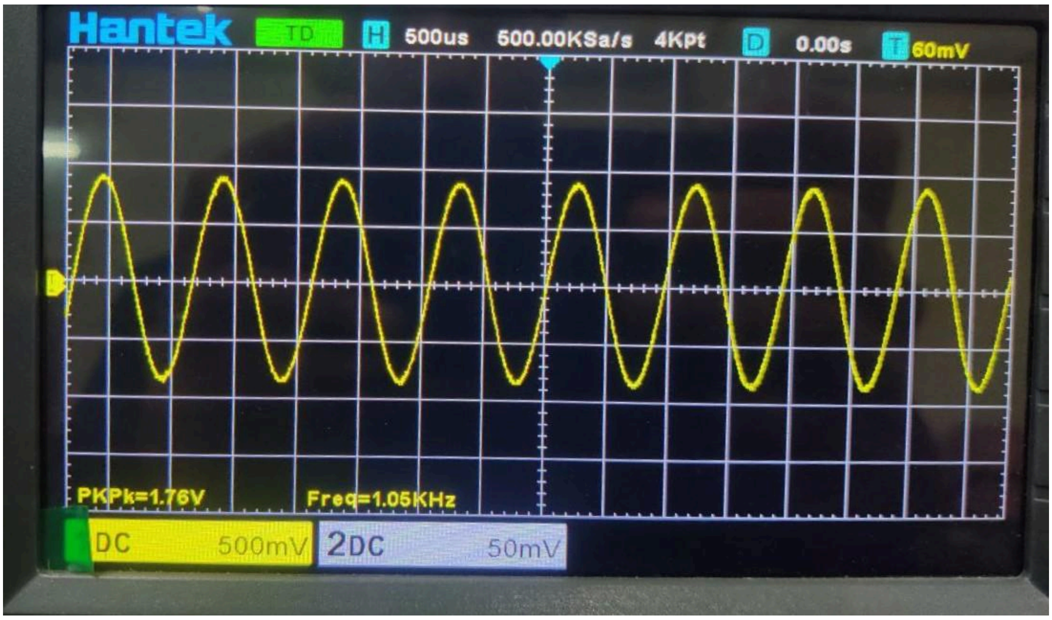

Fig-10: The reconstructed signal when the sampling frequency is 16 kHz

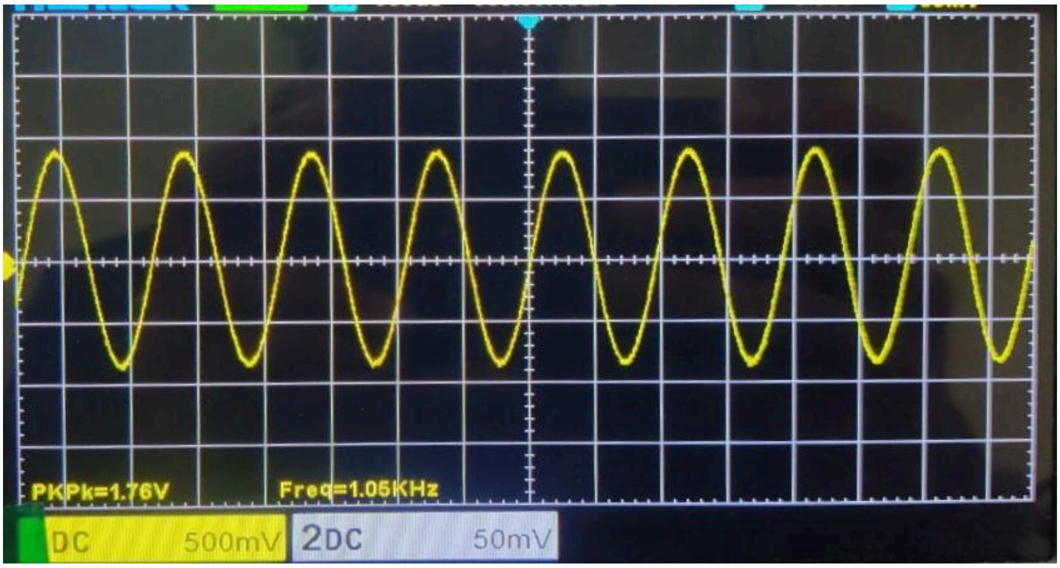

Fig-11: The reconstructed signal when the sampling frequency is 8 kHz

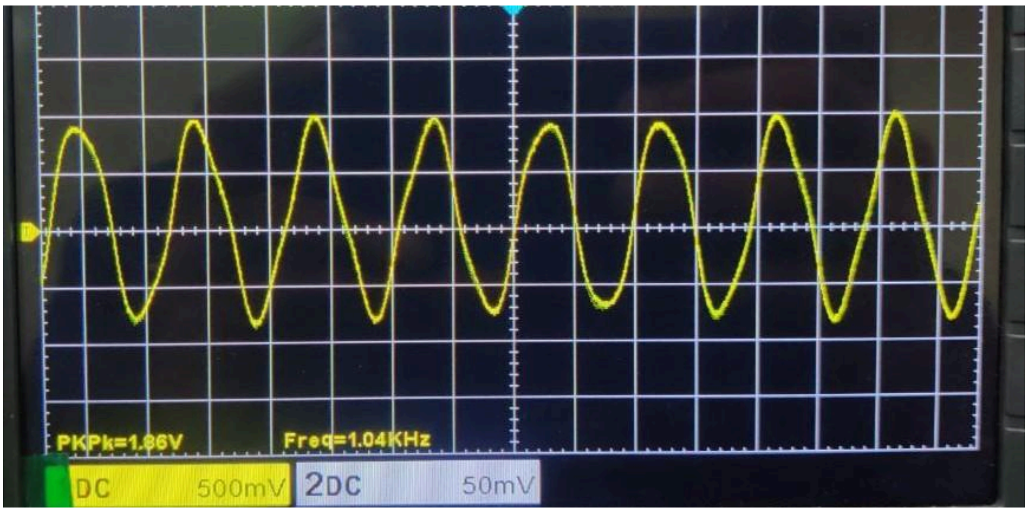

Fig-12: The reconstructed signal when the sampling frequency is 4 kHz

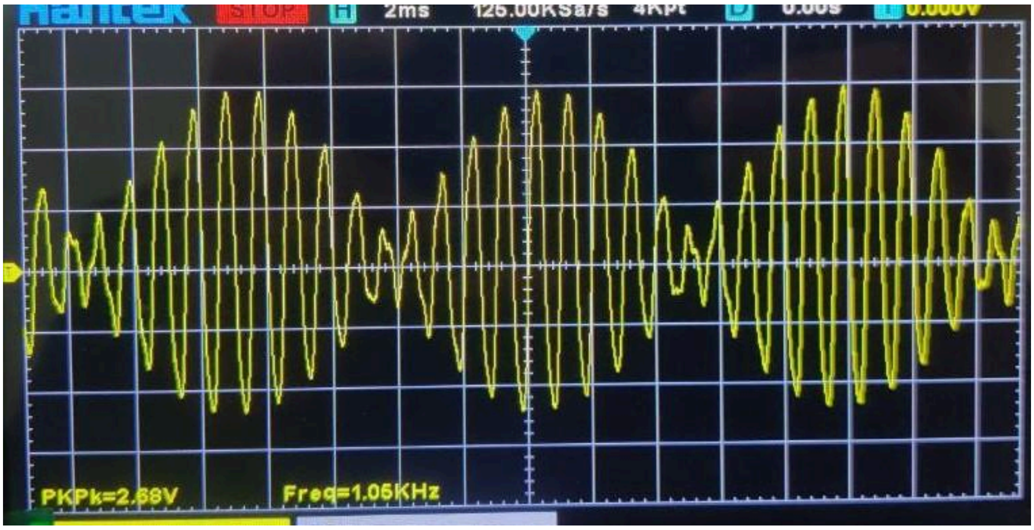

Fig-13: The reconstructed signal when the sampling frequency is 2 kHz

As can be seen from the outputs, once the sampling frequency drops to 4kHz, the signal gets distorted. Although the theoretical lower limit of the sampling frequency is twice the highest frequency component present in the original message signal (which is slightly above 2 kHz in our case), the lowpass filter we use is never ideal, meaning that they don't have the abrupt frequency response as is assumed in Nyquist theorem. Other effects also come into play, like equipment tolerances, amplifier frequency responses, etc. Finally, at 2 kHz sampling, we get a totally distorted signal.

Experimental Data

1. Observe the frequency and amplitude of the message signal.
2. Observe the clock generator output
3. Observe the Channel Selection output, i.e., the sampling pulses
4. Observe the sampled output
5. Observe and measure the frequency and amplitude of the reconstructed signal.

Questions

1. What is the Sampling theorem or Nyquist criterion?
2. What is a low-pass filter? Write the transfer function of a typical lowpass filter.

Experiment No. 6

Pulse Code Modulation of Audio Tune

Objective

- To observe pulse code modulation of an audio signal

Requirements

1. PCM Tx Trainer Board (IT-4203)
2. PCM Rx Trainer Board (IT-4204)
3. Oscilloscope
4. 2 mm patch cords

Theory

Pulse Code Modulation (PCM) is the most basic form of digital pulse modulation. In PCM, a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time and amplitude. The basic operations performed in the transmitter of a PCM system are sampling, quantization, and encoding. The quantizing and encoding operations are usually performed in the same circuit, which is called an analog-to-digital converter.

The sampling process involves sampling the incoming message signal with a train of rectangular pulses. According to the sampling theorem, the sampling rate must be greater than twice the highest frequency component of the message signal to ensure perfect construction of the message signal at the receiver.

Analog signals, like voice signals, have a continuous range of amplitudes, and thus, samples have a continuous amplitude range, in other words, an infinite number of amplitude levels. This original continuous signal can be approximated by a signal constructed of discrete amplitudes selected on a minimum error basis from an available set. The existence of a finite number of discrete amplitude levels is a fundamental condition of digital pulse modulation.

Then, we go on to the encoding process. We represent each amplitude level as a *symbol or code element* in binary format. Each symbol consists of a certain number of bits. This number is *bits-per-sample*. Suppose we have R bits per sample in a PCM system. By using such a code, we can represent a total of 2^R distinct numbers.

In PCM systems, we need frames to organize and structure the digital data representing the sampled analog data. Frames provide synchronization points within the digital stream, allowing the receivers to identify the boundaries of each frame and adequately decode the sampled values. Frames help differentiate between data streams of two different samples.

Procedures

1. Turn on the power of the IT-4203 module.
2. Connect the oscilloscope with the *Audio Generator O/P (TP1).* Adjust the *Gain* knob in the center position. The signal should look like this: *(Intelligence signal photo to be inserted later)*
3. Connect the oscilloscope with the *Clock Generator O/P (TP5).* Observe that the clock is 64 kHz.

Fig-1

4. Connect the *Clock Generator O/P (TP5)* with the *Frame Sync. Generator Clock I/P (TP13)*.
5. Observe the *Frame Sync O/P (TP14)* on the oscilloscope. As the PCM system of the *IT-4203* is an 8-bit A/D system, each PCM frame will consist of 8 clock cycles. That means that after every eight clock cycles, there will be a frame bit output that will synchronize the data stream. The frame synchronization bit (yellow) and clock generator output (green), as seen on the oscilloscope, are shown in the figure below. We can verify that frame synchronization bits appear after every eight clock cycles.

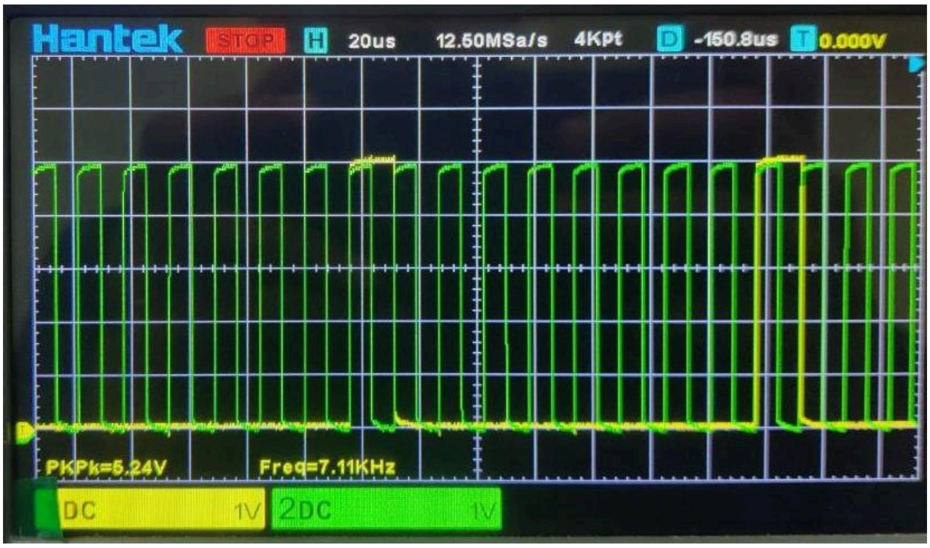

Fig-2

6. Connect the *Audio Generator O/P (TP1)* with the *Sample and Hold Analog I/P (TP9).* Also, connect the *Frame Sync. Generator Frame Sync O/P (TP14)* with the *Frame Sync I/P (TP11).*
7. Observe the *Sample and Hold O/P (TP14)* on the oscilloscope.

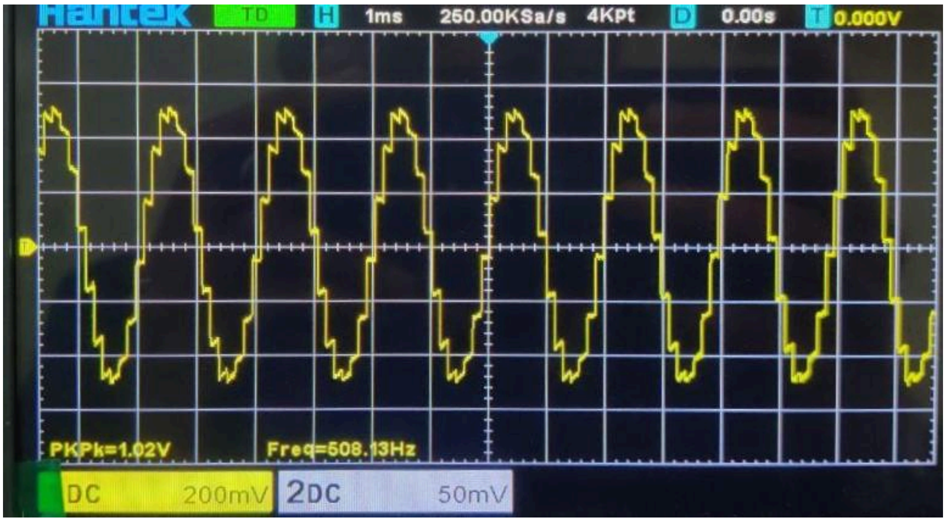

Fig-3

When we observe the sample and hold output together with the frame synchronization output, it’s clear that the analog data is sampled at every frame bit.

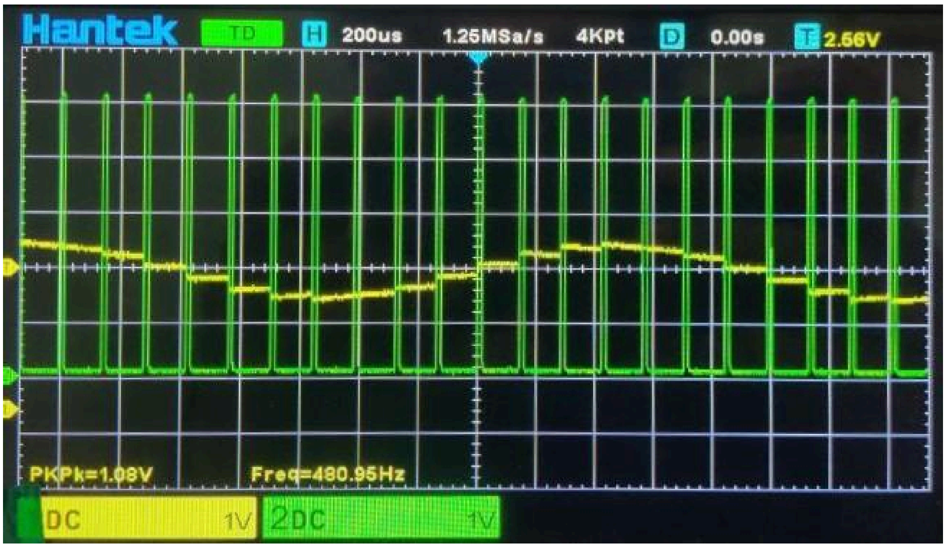

Fig-4: Frame bit (green) and sample and hold output (yellow)

8. Connect the *Sample and Hold O/P (TP10)* with the *Analog to Digital Converter Sample and Hold I/P (TP12)*. Observe that *Buffer* Output LEDs start to blink fast, indicating the appearance of the data stream.
9. Observe the *Multiplexer PCM O/P (TP15).* Here, we can observe the data stream generated as a result of pulse code modulation of the audio generator block output. From the figure below, we can observe each sinusoidal wave sample, the corresponding 8-bit PCM data, and the frame bit. Thus, we have successfully constructed a PCM output from an analog signal.

Fig-5: Sample and Hold output (yellow), PCM output (green)

10. Now, we shall start the reconstruction process of the audio signal. Connect the *Clock Generator O/P (TP5), F/S Frame Sync O/P (TP14),* and the *PCM O/P (TP 15)* of the *IT-4203* module with the *Clock I/P*

(TP10), F/S I/P (TP9), and the PCM I/P (TP8) of the *IT-4204* module. Also, make a connection between any two ground inputs of the modules.

11. Observe the *S/Hold O/P (TP6)* on the oscilloscope.

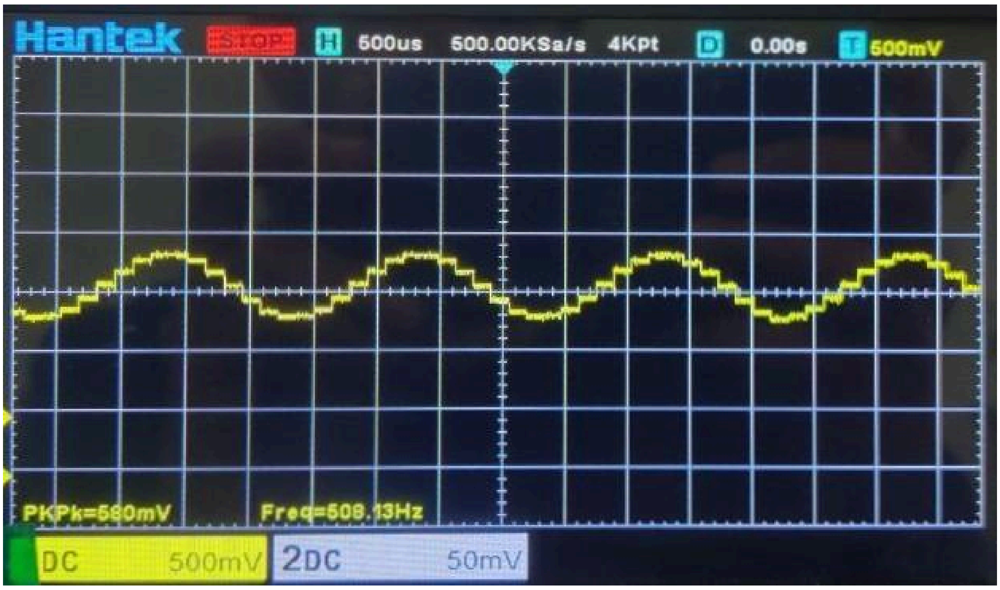

Fig-6

12. Connect the *S/Hold O/P (TP6)* with the *Low Pass Filter I/P (TP15).* This will remove the high-frequency content of the sample and hold output.
13. Connect the *Low Pass Filter O/P (TP16)* with the *AC Amplifier I/P (TP12)*. Observe the *AC Amplifier O/P (TP13)* on the oscilloscope. You can adjust the gain by rotating the *Gain Adj.* knob. You can also connect the *Low Pass Filter O/P (TP16)* with the *Audio Amplifier I/P (TP11),* adjust the gain and hear the sound.

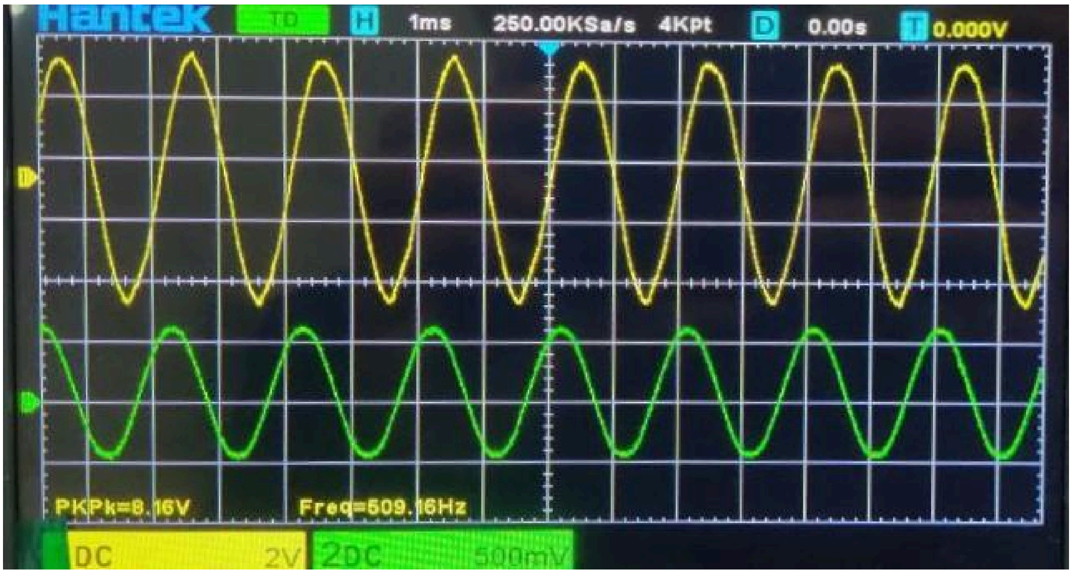

Fig-7: Received signal (yellow), original message signal (green)

Experimental Data

1. Observe the frequency and amplitude of the audio signal and draw the signal.
2. Observe the frequency of the clock signal and draw the signal.
3. Observe the frame synchronization bits and draw them along with the clock signal.
4. Observe and draw the sample and hold output.
5. Observe and draw the PCM output along with the sample and hold output.
6. Observe and draw the received signal along with the original message signal.

Questions

1. Draw the block diagram of the essential elements of the PCM system.
2. What is uniform and non-uniform quantization?

Experiment No. 7

Pulse Code De- De-Modulation of Audio Tune

Objective

- To retrieve the original audio signal from the pulse-code modulated signal.

Requirements

- PCM Tx Trainer Board (IT-4203)
- PCM Rx Trainer Board (IT-4204)
- 2mm Patch Cords
- Oscilloscope

Theory