DEPARTMENT OF PURE AND APPLIED PHYSICS

B.Sc. (ELECTRONICS) course

Academic Year- 2023-24

Semester-VI

Core 14: Communication Electronics Lab

Course Code: PLUFLT2 Credit: 2

Experiments:

1. Study of Amplitude Modulation and Demodulation.
2. Study of Frequency Modulation and Demodulation.
3. Study of TDM (Time division multiplexing).
4. Study of Differential Coded Pulse Code Modulation.
5. Study of Adaptive Predictive Differential Coded Pulse Code Modulation.
6. Study of Phase Shift Keying modulation techniques.

Experiment 1: Amplitude Modulation and Demodulation

Introduction:

In radio transmission, it is necessary to send audio signal (e.g. Music,speech etc) from a broad casting station over great distances to a receiver. This communication of audio signal does not employ any wire and is sometimes called wireless. The audio signal cannot be sent direetly over the air for appreciable distance. Even if the audio signal is converted into electrical signal, the later cannot be sent very far without employing large amount of power. The energy of a wave is directly proportional to its frequency. At audio frequencies (20Hz to 20 KHz) the signal power is quite small and radiation is not practicable.
The radiation of electrical energy is practicable.

only at high fiequencies e.g. above 20 KHz. The high frequency signals can be sent thousands of miles even with comparatively small power.Therefore, if audio signal is to be transmitted properly, some means must be devised which, will permit transmission to occur at high frequencies while it simultaneously allows the carrying of audio signal. This is achieved by imposing electrical audio signal on high frcquency carrier. The resultant waves 'are known as modulated waves or radio waves and the process is called modulation. At the radio receiver, the audio signal is extracted from the modulated wave by the process called demodulation. The signal is then amplified and reproduced into sound by the loudspeaker.

MODULATION

Modulation is a process of mixing a signal with a sinusoid to produce a new signal. This new signal, conceivably, will have certain benefits of an un-modulated signal, especially during transmission. If we look at a general function for a sinusoid:

$$f(t)=A \sin(wt+j) \quad \text{eqn.A}$$

We can see that this sinusoid has 3 parameters that can be altered, to affect the shape of the graph. The first term, A, is called the magnitude, or amplitude of the sinusoid. The next term, w is known as the frequency, and the last term, j is known as the phase angle. All 3 parameters can be altered to transmit data. The sinusoidal signal that is used in the modulation is known as the carrier signal, or simply "the carrier". The signal that is being modulated is known as the "data signal'. It is important to notice that a simple sinusoidal carrier contains no information of its own.

A high frequency carrier wave is used to carry the audio signal which is done by changing some characteristic of carrier wave in accordance with the signal. Under such conditions; the audio signal will be contained in the resultant wave. In modulation, some characteristic of a carrier wave is changed in accordance with the intensity (i.e. Amplitude) of the signal. The resultant wave is called modulated wave or radio wave and contains the audio signal.Therefore, modulation permits the transmission to occur at high frequency while it simultaneously allows the carrying of the audio signals.

Types of Modulation

There are 3 different types of modulation: Amplitude modulation,Frequency modulation, and Phase modulation. From the eqn.A we can see that there are three variable factors in it and so modulation can be done to all this three parameters.

NEED FOR MODULATION

Modulation is extremely necessary in communication system due to the following reasons.

1. PRACTICAL ANTENNA LENGTH

In order to transmit a wave effectively, the length of the transmitting antenna should be approximately equal to the wavelength of the wave.

Now Wavelength = Velocity/ frequency = 3x10^8/ Frequency

As the audio frequencies range from 20Hz to 20 KHz, therefore, if they are transmitted directly into space, the length of the transmitting antenna required would be extremely large. For instance, to radiate a frequency of 20 KHz directly into space, we would need an antenna length of 3 x 10^8 / 20 x 10^3= 15,000 meters. This is too long antenna to be constructed practically. For this reason, it is impracticable to radiate audio signal directly into space. On the other hand, if a carrier wave say of

1000 KHz is used to carry the signal, we need an antenna length of 300 meters only and this size can be easily constructed.

2. OPERATING RANGE

The energy of a wave depends upon its frequency. The greater the frequency of the wave, the greater is the energy possessed by it. As the audio signal frequencies are small, therefore these cannot be transmitted over large distance if radiated directly into space. The only practical solution is to modulate a high frequency carrier wave with audio signal and permit the transmission to occur at this high frequency (i.e. carrier frequency).

3. WIRELESS COMMUNICATION

One desirable feature of radio transmission is that it should be carried without wires i.e. radiated into space. At audio frequencies radiation is not practicable because the efficiency of radiation is poor.However, efficient radiation of electrical energy is possible at high frequencies (>20 KHz). For this reason, modulation is always done in communication systems.

AMPLITUDE MODULATION

Amplitude modulation (AM) is a technique used in electronic communication, most commonly for transmitting information via a radio carrier wave. AM works by varying the strength of the transmitted signal in relation to the information being sent. For example, changes in the signal strength can be used to specify the sounds to be reproduced by a loudspeaker, or the light intensity of television pixels. (Contrast this with frequency' modulation, also commonly used for. sound transmissions, in which the frequency is varied; and phasc modulation, often used in remote controls. in which the phase is varied)

In the mid-1870s, a form of amplitude modulation—initially called “undulatory currents"—was the first method to successfully produce quality audio over telephone lines. Beginning with Reginald Fessenden's audio demonstrations in 1906, it was also the original method used for audio radio transmissions, and remains in use today by many forms of communication-"AM" is often used to refer to the medium wave broadcast band.

Forms of amplitude modulation:

As originally developed for the electric telephone. amplitude modulation was used to add audio information to the low-powered direct current flowing from a telephone transmitter to a receiver. As a simplified explanation, at the transmitting end, a telephone microphone was used to vary the strength of the transmitted current, according to the frequency and loudness of the sounds received. Then, at the receiving end of the telephone line, the transmitted electrical current affected an electromagnet, which strengthened and weakened in response to the strength of the current. In turn, the electromagnet produced vibrations in the receiver diaphragm, thus closely reproducing the frequency and loudness of the sounds originally heard at the transmitter.

In contrast to the telephone, in radio communication what is modulated is a continuous wave radio signal (carrier wave) produced by a radio transmitter. In its basic form, amplitude modulation produces a signal with power concentrated at the carrier frequency and in two adjacent sidebands. This process is known as heterodyning. Each sideband is equal in bandwidth to that of the modulating signal and is a mirror image of the other.

Amplitude modulation that results in two sidebands and a carrier is often called double sideband amplitude modulation (DSB-AM). Amplitude modulation is inefficient in terms of power usage and much of it is wasted. At least two-thirds of the power is concentrated in the carrier signal, which carries no useful information (beyond the fact that a signal is present); the remaining power is split between two identical sidebands, though only one of these is needed since they contain identical information.

To increase transmitter efficiency, the carrier can be removed(suppressed) from the AM signal. This produces a reduced-carrier transmission or double-sideband suppressed-carrier (DSBSC) signal.

A suppressed-carrier amplitude modulation scheme is three times more power-efficient than traditional DSB-AM. If the carrier is only partially suppressed, a double-sideband reduced-carrier (DSBRC)signal results. DSBSC and DSBRC signals need their carrier to be regenerated (by a beat frequency oscillator, for instance) to be demodulated using conventional techniques.

Even greater efficiency is achieved-—-at the expense of increased transmitter and receiver complexity—by completely suppressing both the carrier and one of the sidebands. This is single-side hand modulation, widely used in amateur radio due to its efficient use of both power and bandwidth.

A simple form of AM often used for digital communications is onoff keying, a type of amplitude-shift keying by which binary data is represented as the presence or absence of a carricr wave. This is commonly used at radio frequencies to transmit Morse code, referred to as continuous wave (CW) operation.

Modulation index

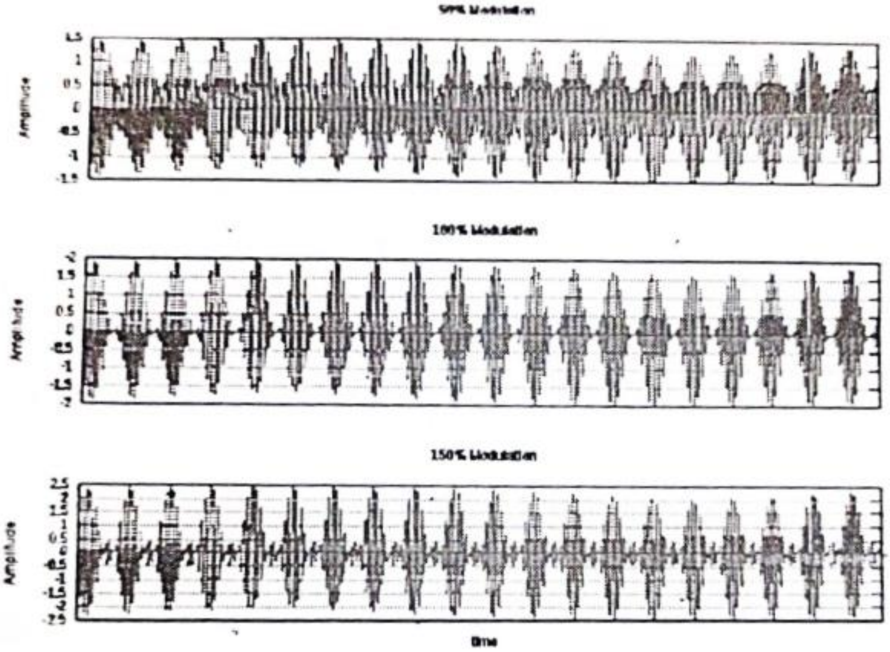
It can be defined as the measure of extent of amplitude variation about an unmodulated maximum carrier. As with other modulation indices, in AM, this quantity, also called modulation depth, indicates by how much the modulated variable varies around its 'original level. For AM, it relates to the variations in the carrier amplitude and is defined as:

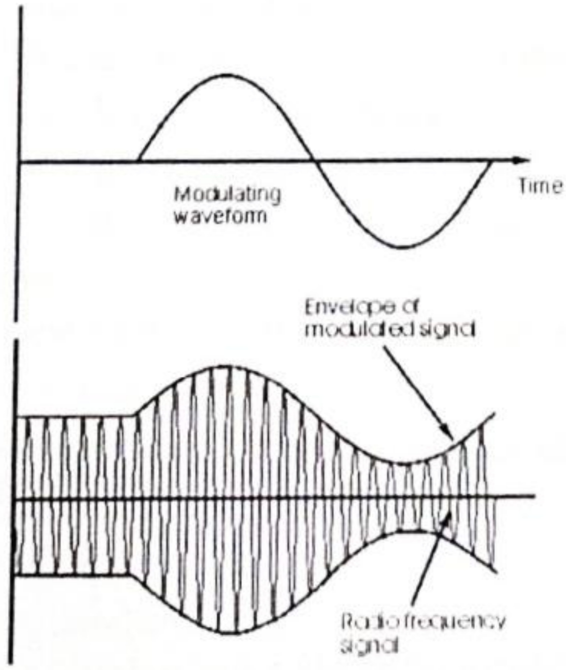
h= peak value of m(T)/ M = M/A

T Where A and were introduced above.

So if h = 0.5, the carrier amplitude varies by 50% above and below its unmodulated level, and for h = 1.0 it varies by 100%. To avoid distortion in the A3E transmission mode, modulation depth greater than 100% must be avoided. Practical transmitter systems will usually incorporate some kind of limiter circuit, such as a VOGAD,to ensure this. However, AM demodulators can be designed to detect the inversion (or 180 degree phase reversal) that occurs when modulation exceeds 100% and automatically correct for this effect.

Variations of modulated signal with percentage modulation are shown below. In cach image, the maximum amplitude is higher than in the previous image. Note that the scale changes from one image to the next.


100% Modulation
150% Modulation
Amplitude


Modulating
waveform
Time
Envelope of
modulated signal
Radio frequency
signal

TO STUDY AMPLITUDE MODULATION THEORY:

Amplitude modulation (AM) is a method of impressing data onto an alternating-current (AC) carrier waveform. The highest frequency of the modulating data is normally less than 10 percent of the carrier frequency. The instantaneous amplitude (overall signal power) varies depending on the instantaneous amplitude of the modulating data.

Test Points:

TPI : CARRIER FREQUENCY INPUT

TP2 : SIGNAL INPUT

TP3 : AM OUTPUT

TP4 : AM DE-MOD OUTPUT

Note: It is recommended not to change the adjustients Positions. The settings are all Factory set.

PROCEDURE

1. Connect the AC Supply to the Kit.

2. Connect the Sine Wave Generator output to the 'Modulating

Input' (TP2) post of AM modular block.

3. Connect the Carrier signal output from Carrier Generator to the Carrier I/p(TP1) post of AM Modulator block.

4. Switch ON the power.

5. Connect the AM O/P (TP3) to the AM I/P of AM

DEMODULATOR section.

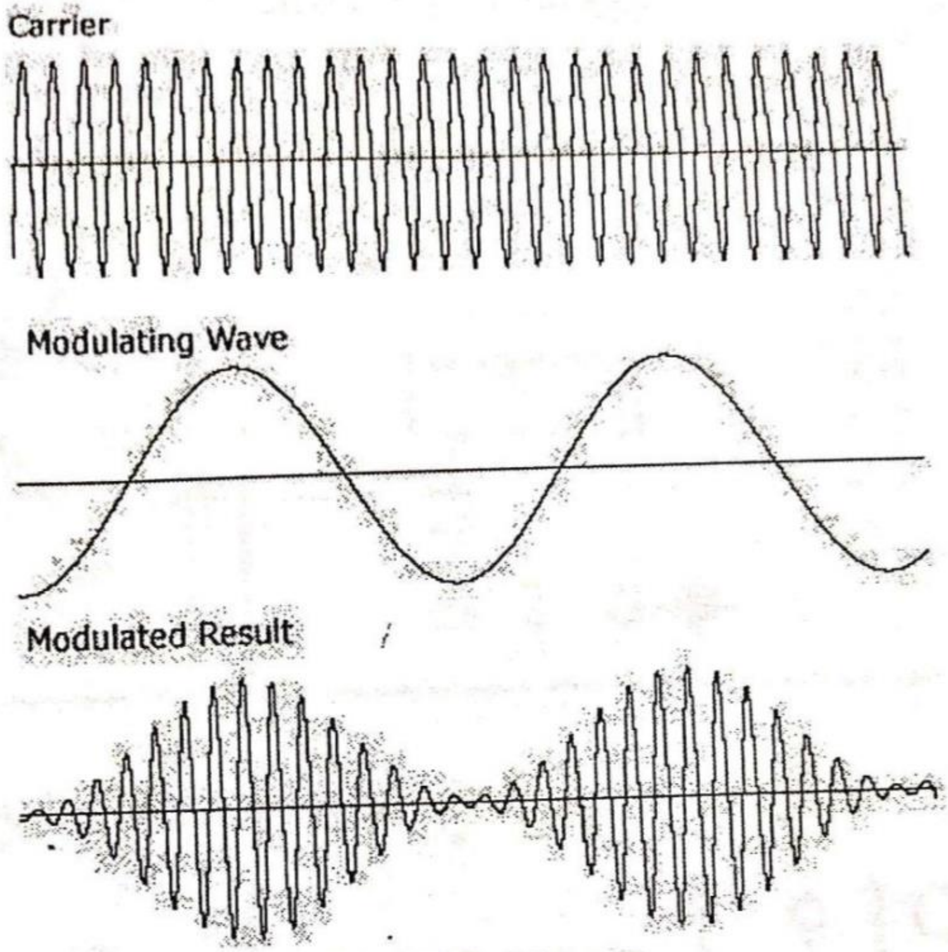
6. Observe the following waveforms on oscilloscope

a. Sine Wave Generator O/P.

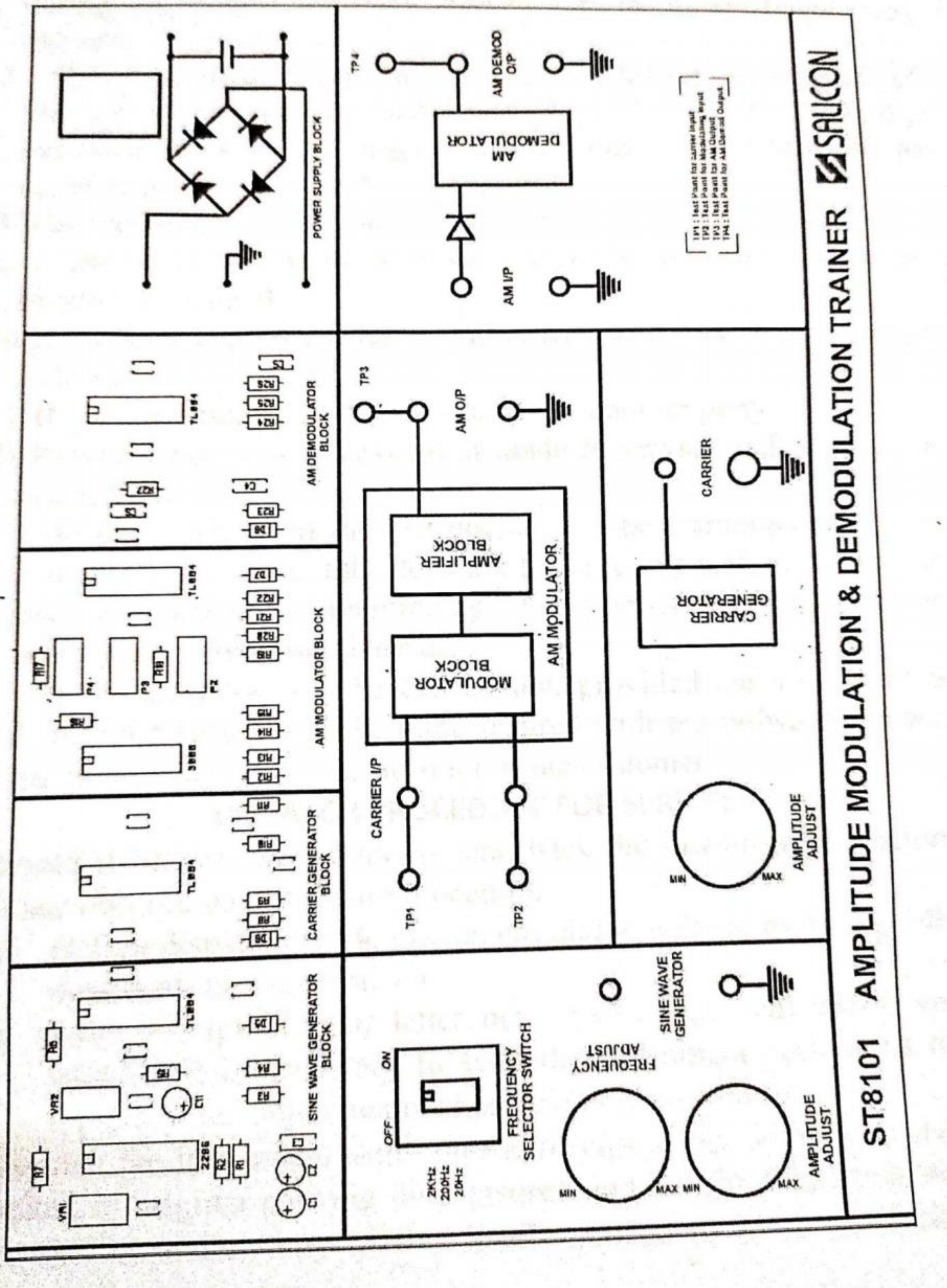
b. Carrier Frequency O/P.

c. AM O/P at AM Modulator.

d. AM Demodulated signal at AM DEMOD O/P.(TP4)


Carrier
Modulating Wave
Modulated Result

BLOCK DIAGRAM



Experiment 2 : Study of frequency Modulation and Demodulation

In radio transmission, it is necessary to send audio signal (e.g. Music, speech etc.) from a broad casting station over great distances to a receiver. This communication of audio signal does not employ any wire and is sometimes called wireless. The audio signal cannot be sent directly over the air for appreciable distance. Even if the audio signal is converted into electrical signal, the later cannot be sent very far without employing large amount of power. The energy of a wave is directly proportional to its frequency. At audio frequencies (20Hz to 20 KHz) the signal power is quite small and radiation is not practicable.

The radiation of electrical energy is practicable only at high frequencies e.g. above 20.KHz. The high frequency signals can be sent thousand of miles even with comparatively small power. Therefore, if audio signal is to be transmitted properly, some means must be devised which, will permit transmission to occur at high frequencies while it simultaneously allows the carrying of audio signal. This is achieved by imposing electrical audio signal on high frequency carrier. The resultant waves are known as modulated waves or radio waves and the process is called modulation. At the radio receiver, the audio signal is extracted from the modulated wave by the process called demodulation. The signal is then amplified and reproduced into sound by the loudspeaker.

MODULATION

Modulation is a process of mixing a signal with a sinusoid to produce a new signal. This new signal, conceivably, will have certain benefits often un-modulated signal, especially during transmission. If we look at a general function for a sinusoid:

$F(t)=A \sin(wt+j)$ eqn.A

We can see that this sinusoid has 3 parameters that can be altered, to affect the shape of the graph. The first term, A, is called the magnitude, or amplitude of the sinusoid. The next term, w is known as the frequency, and the last term, j is known as the phase angle. All 3 parameters can be altered to transmit data. The sinusoidal signal that is used in the modulation is known as the carricr signal, or simply "the carrier". The signal that is being modulated is known as the "data signal". It is important to notice that a simple sinusoidal carrier contains no information of its own.

A high frequency carrier wave is used to carry the audio signal which is done by changing some characteristic of carrier wave in accordance with the signal. Under such conditions, the audio signal will be contained in the resultant wave. In modulation, some characteristic of a carrier wave is changed in accordance with the intensity (i.e. Amplitude) of the signal. The resultant wave is called modulated wave or radio wave and contains the audio signal.Therefore, modulation permits the transmission to occur at high frequency while it simultaneously allows the carrying of the audio signals.

Types of Modulation

There are 3 different types of modulation: Amplitude modulation, Frequency modulation, and Phase modulation. From the eqn.A we can see that there are three variable factors in it and so modulation can be done to all this three parameters.

NEED FOR MODULATION

Modulation is extremely necessary in communication system due to the following reasons.

1. PRACTICAL ANTENNA LENGTH

In order to transmit a wave effectively, the length of the transmitting antenna should be approximately equal to the wavelength of the wave.

Now Wavelength = Velocity/Frequency = 3x10^8 meters/ Frequency (Hz)

As the audio frequencies range from 20Hz to 20 KHz, therefore, if they are transmitted directly into space, the length of the transmitting antenna required would be extremely large. For instance, to radiate a frequency of 20 KHz directly into space, we would need an antenna length of 3 x 10 1 20 x 103 = 15,000 meters. This is too long antenna to be constructed practically. For this reason, it is impracticable to radiate audio signal directly into space. On the other hand, if a carrier wave say of 1000 KHz is used to carry the signal,we need an antenna length of 300 meters only and this size can be easily constructed.

2. OPERATING RANGE

The energy of a wave depends upon its frequency. The greater the frequency of the wave, the greater is the energy possessed by it. As the audio signal frequencies are small, therefore these cannot be transmitted over large distance if radiated directly into space. The only practical solution is to modulate a high frequency carrier wave with audio signal and permit the transmission to occur at this high frequency (i.e. carrier frequency).

3. WIRELESS COMMUNICATION

One desirable feature of radio transmission is that it should be carried without wires i.e. radiated into space. At audio frequencies radiation is not practicable because the. efficiericy of radiation is poor.However, efficient radiation of electrical energy is possible at high frequencies (>20 KHz). For this reason, modulation is always done in communication systems.

FREQUENCY MODULATION

frequency modulation (FM) conveys information over a carrier wave by varying its instantaneous frequency (contrast this with amplitude modulation, in which the amplitude of the carrier is varied while its frequency remains constant). In analog applications, the difference between the instantaneous and the base frequency of the carrier is directly proportional to the instantaneous value of the input signal. Digital data can be sent by shifting the carrier's frequency among a set of discrete values, a technique known as frequency-shift keying.
Frequency modulation can be regarded as phase modulation where the carrier phase modulation is the time integral of the FM modulating signal.

Suppose the baseband data signal (the message) to be transmitted is

$$X_m(t)$$

As is restricted in amplitude to be

$$|X_m(t)| \leq 1$$

And the sinusoidal carrier is

$$X_c(t) = Ac \cos (2\Pi f_c t)$$

Where f_c is the carrier's base frequency And A_c is the carrier's amplitude. The moderator combines the carrier with the base band data design to get the transmitted signal.

$$
\begin{aligned}
y(t) &= A_c \cos\left(2\pi \int_0^t f(\tau)d\tau\right) \\
&= A_c \cos\left(2\pi \int_0^t [f_c + f_\Delta x_m(\tau)]\, d\tau\right) \\
&= A_c \cos\left(2\pi f_c t + 2\pi f_\Delta \int_0^t x_m(\tau)d\tau\right) \quad \ldots \quad (1)
\end{aligned}
$$

In this equation, f(T)is the instantaneous frequency of the oscillator and fΔ the frequency deviation, which represents the maximum shift away from fe in one direction, assuming $x_m(t)$ is limited to the range ±l.

Although it may seem that this limits the frequencies in use to f_c+f_Δ . this neglects the distinction between instantaneous frequency, and spectral frequency. The frequency spectrum of an actual FM signal has components extending out to infinite frequency, although they become negligibly small beyond a point.

Sinusoidal baseband signal

Whilst it is an over-simplification, a baseband modulated signal may be approximated by a sinusoidal Continuous Wave signal with a frequency fm. The integral of such a signal is

$$
x_m(t) = \frac{A_m \cos(2\pi f_m t)}{2\pi f_m}
$$

Thus, in this specific case, equation one above simplifies to

$$
y(t) = A_c \cos\left(2\pi f_c t + \frac{f_\Delta}{f_m} \cos(2\pi f_m t)\right)
$$

Where the amplitude AM of the modulating sinusoid is represented by the peak deviation FΔ(see frequency deviation).

The harmonic distribution of a sine wave carrier modulated by such a sinusoidal signal can be represented with bassel function. This provides a basic for a mathematical understanding of frequency modulation in frequency domain.

Modulation index as with the other modulation indices. This quantity indicates by how much the modulated variable varies around its unmodulated level. It relates to the variations in the frequency of the carrier signal

$$h = \frac{\Delta f}{f_m} = \frac{f_\Delta |x_m(t)|}{f_m}$$

where f_m is the highest frequency component present in the modulating signal $x_m(t)$, and Δf the Peak frequency-deviation, i.e. the maximum deviation of the instantaneous frequency from the carrier frequency. If h 《 1, the modulation is called narrowband FM, and its bandwidth is approximately $2f_m$. If h > 1, the modulation is called wideband FM and its bandwidth is approximately 2fΔ.

While wideband FM uses more bandwidth, it can improve signal-to-noise ratio significantly. With a tone-modulated FM wave, if the modulation frequency is held constant and the modulation index is increased, the (non-negligible) bandwidth of the FM signal increases but the spacing between spectra stays the same: some spectral components decrease in strength as others increase. If the frequency deviation is held constant and the modulation frequency increased the spacing between spectra increases.

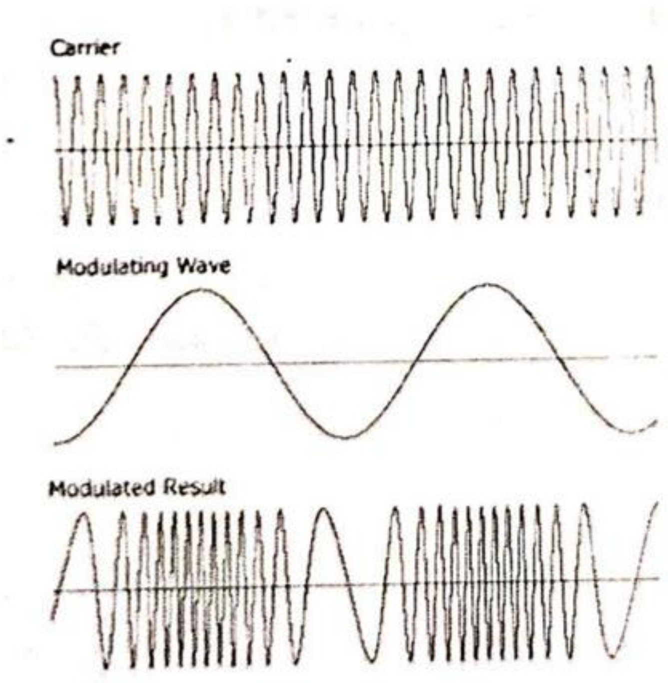
Carson's rule

Main article: Carson bandwidth rule

A rule of thumb. Carson's rule states that nearly all (-98%) of the power of a frequency-modulated signal lies within a bandwidth B_T of

$B_T = 2(\triangle f + f_m)$

Where △f, as defined above, is the peak deviation of the instantaneous frequency f(t) from the center carrier frequency f_e.



TO STUDY FREQUENCY MODULATION

THEORY

frequency modulation (FM) conveys information over a carrier wave by varying its instantaneous frequency (contrast this with amplitude modulation, in which the amplitude of the carrier is varied while its frequency remains constant).

Test Points:

TPI : MODULATING INPUT

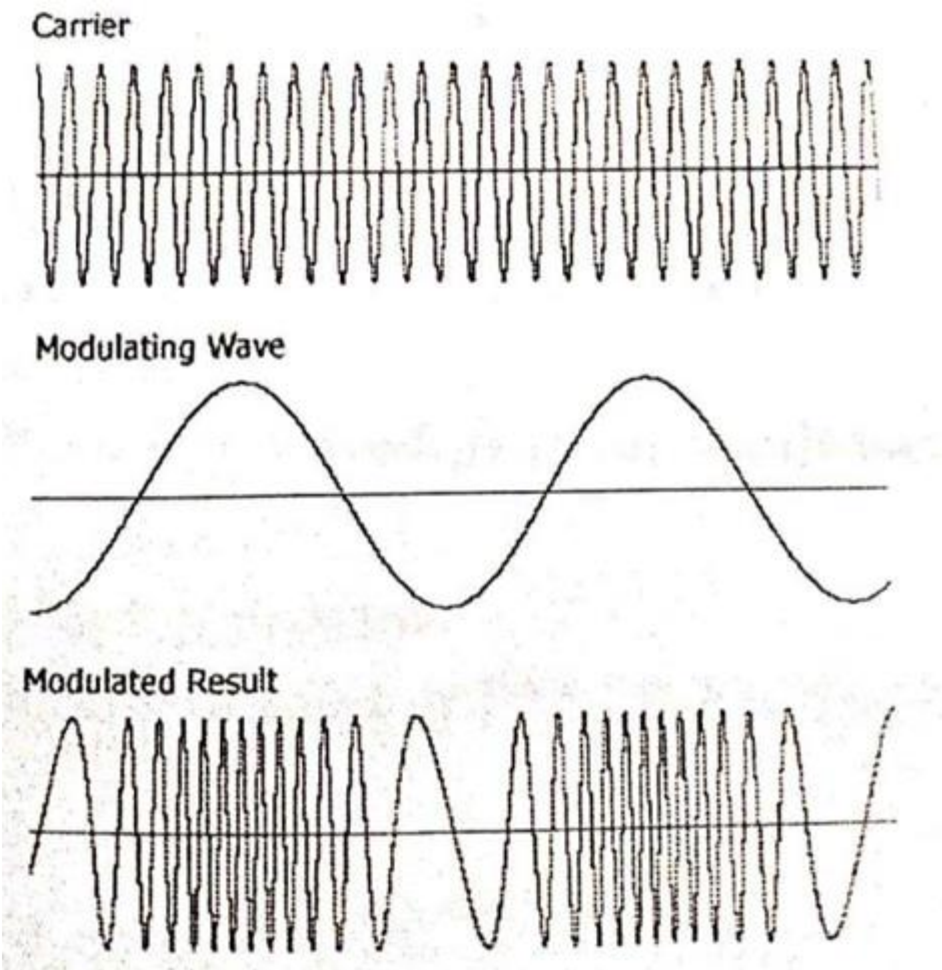
TP2: FM MODULATION OUTPUT

Note : It is recommended not to change the adjustments Positions . The settings are all Factory set.

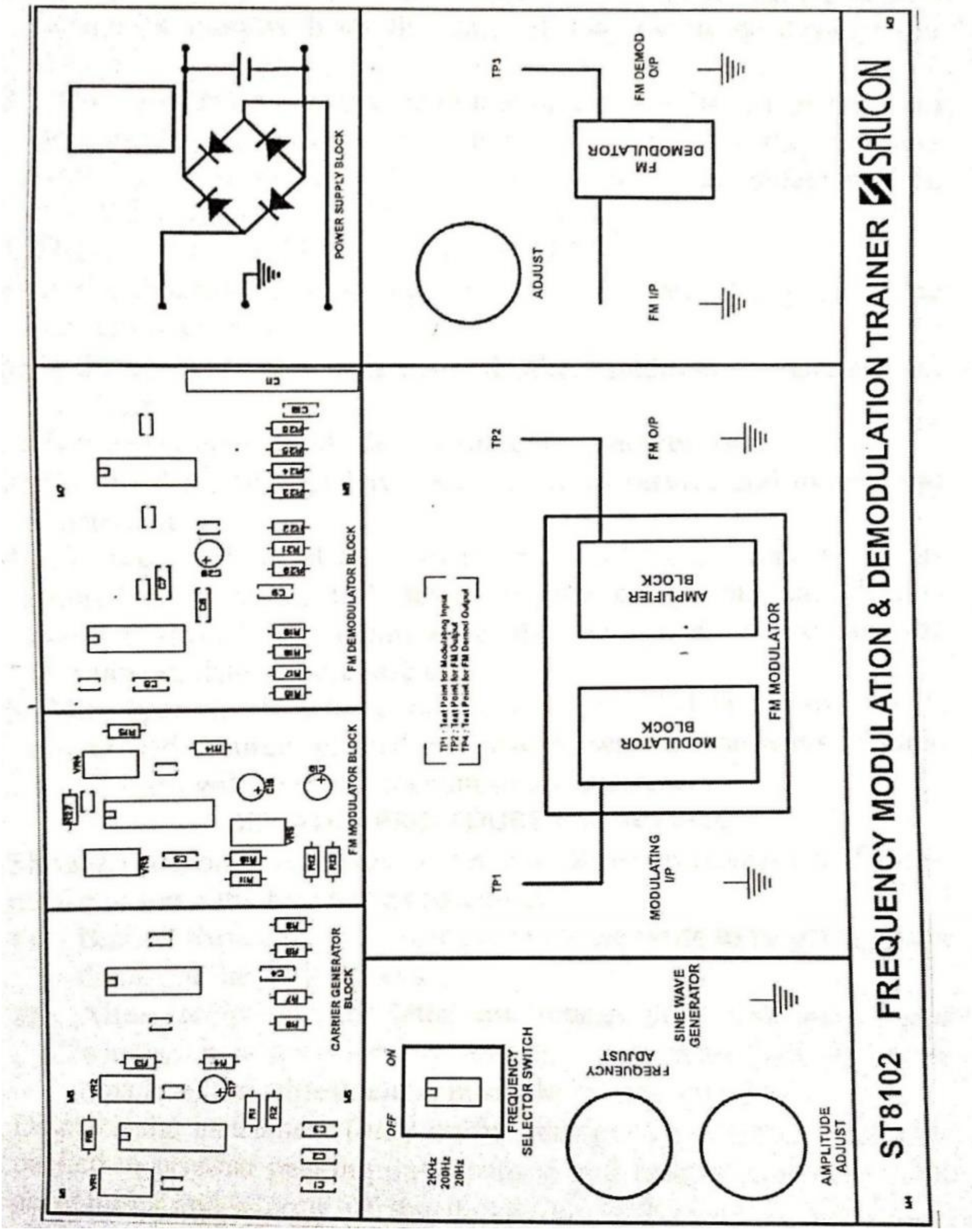
PROCEDURE

1. Connect the AC Supply to the Kit.
2. Connect the SINE WAVE GENERATOR output to the
'Modulating Input' Post of FM modulator block.3. Connect all grounding points of the blocks.

4. Switch ON the power.
5. Observe the following waveforms on oscilloscope.
a. Sine Wave Generator O/P.
b. FM O/P at FM Modulator.



BLOCK DIAGRAM



Experiment 3 : Time -Division Multiplexing

0BJECTIVE

To demonstrate Time Division Multiplexing and demultiplexing process using Pulse amplitude modulation signals.

HARDWARE REQUIRED

TDM Trainer Kit-ST2102

2. CRO

3. Patch Chords

4. Probes

INTRODUCTION

An important feature of pulse-amplitude modulation is a conservation of time. That is, for a given message signal, transmission of the associated PAM wave engages the communication channel for only a fraction of the sampling interval on a periodic basis Hence, some of the time interval between adjacent pulses of the PAM wave is cleared for use by the other independent message signals on a time-shared basis. By so doing, we obtain a time-division multiplex system (TDM), which enables the joint utilization of a common channel by a plurality of independent message signals without mutual interference.

Each input message signal is first restricted in bandwidth by a low-pass pre-alias filter to remove the frequencies that are nonessential to an adequate signal representation. The pre-alias filter outputs are then applied to a commutator, which is usually implemented using electronic switching circuitry. The function of the commutator is two-fold: (1) to take a narrow sample of each of the N input messages at a rate fs that is slightly higher than 2W, where W is the cutoff frequency of the pre-alias filter, and (2) to sequentially interleave these N samples inside a sampling interval Ts 1/fs.

Indeed, this latter function is the essence of the time-division multiplexing operation. Following the commutation process, the multiplexed signal is applied to a pulse-amplitude modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over the communication channel.

At the receiving end of the system, the received signal is applied to a pulse- amplitude demodulator, which performs the reverse operation of the pulse amplitude modulator. The short pulses produced at the pulse demodulator output are distributed to the appropriate

low-pass reconstruction filters by means of a de commutator, which operates in synchronism with the commutator in the transmitter. . This synchronization is essential
for satisfactory operation of the TDM system, and provisions have to be made for it.

1.4 BLOCK DIAGRAM

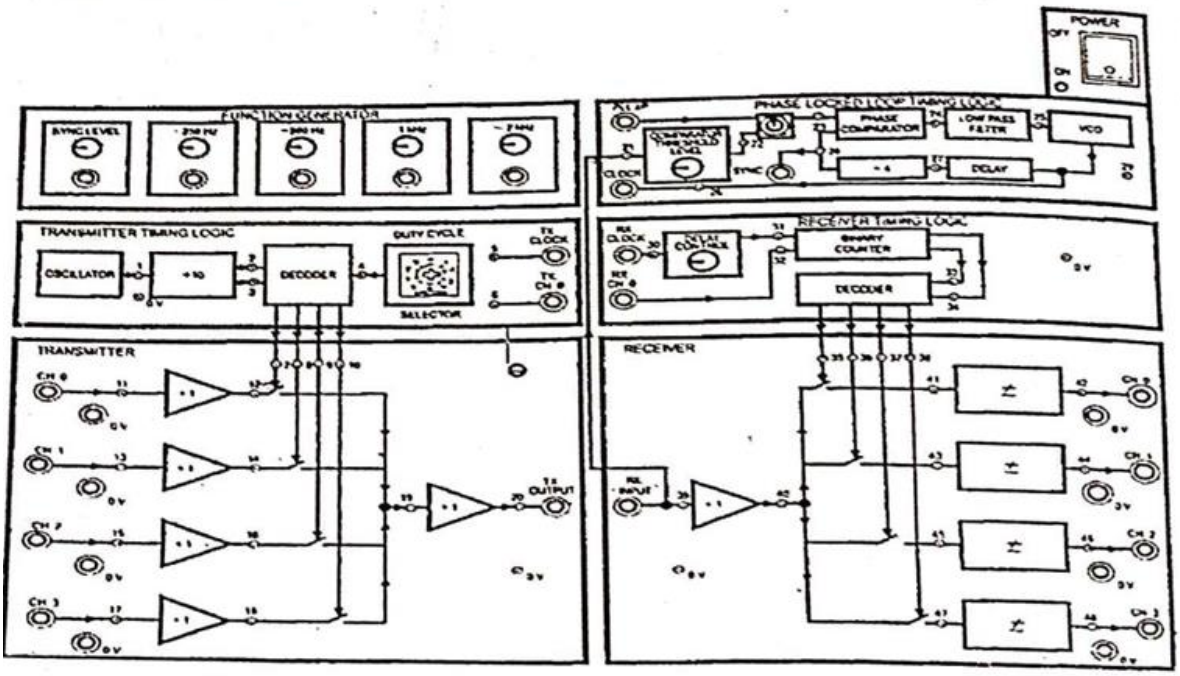


Figure 1.1 TDM Trainer Kit –ST2102 Block Diagram

PRELAB QUESTIONS

1. What is multiplexing?
2. Mention the types of multiplexing?
3. What is the need for multiplexing?
4. What is the bit rate of T1,T2,T3 and T4 carrier systems?
5. Compare synchronous and asynchronous TDM.
6. What are the functions of commutator switch?
7. Give the advantages of multiplexing.

TEST PROCEDURE:

1. Take the signals from the function generator and give it to the channels (CHO..CH3) present in the transmitter using patch chords. Note down the amplitude and time period of each signal.
2. Measure the amplitude and time period at the transmitter output point
3. Using a patch chord, connect transmitter output to receiver input.
4. For synchronization purpose, connect the transmitter clock and receiver clock and also

transmitter CHO and receiver CHO.

5. See the output before the filter and after the filter for all the channels connected.

TRANSMITTER SECTION

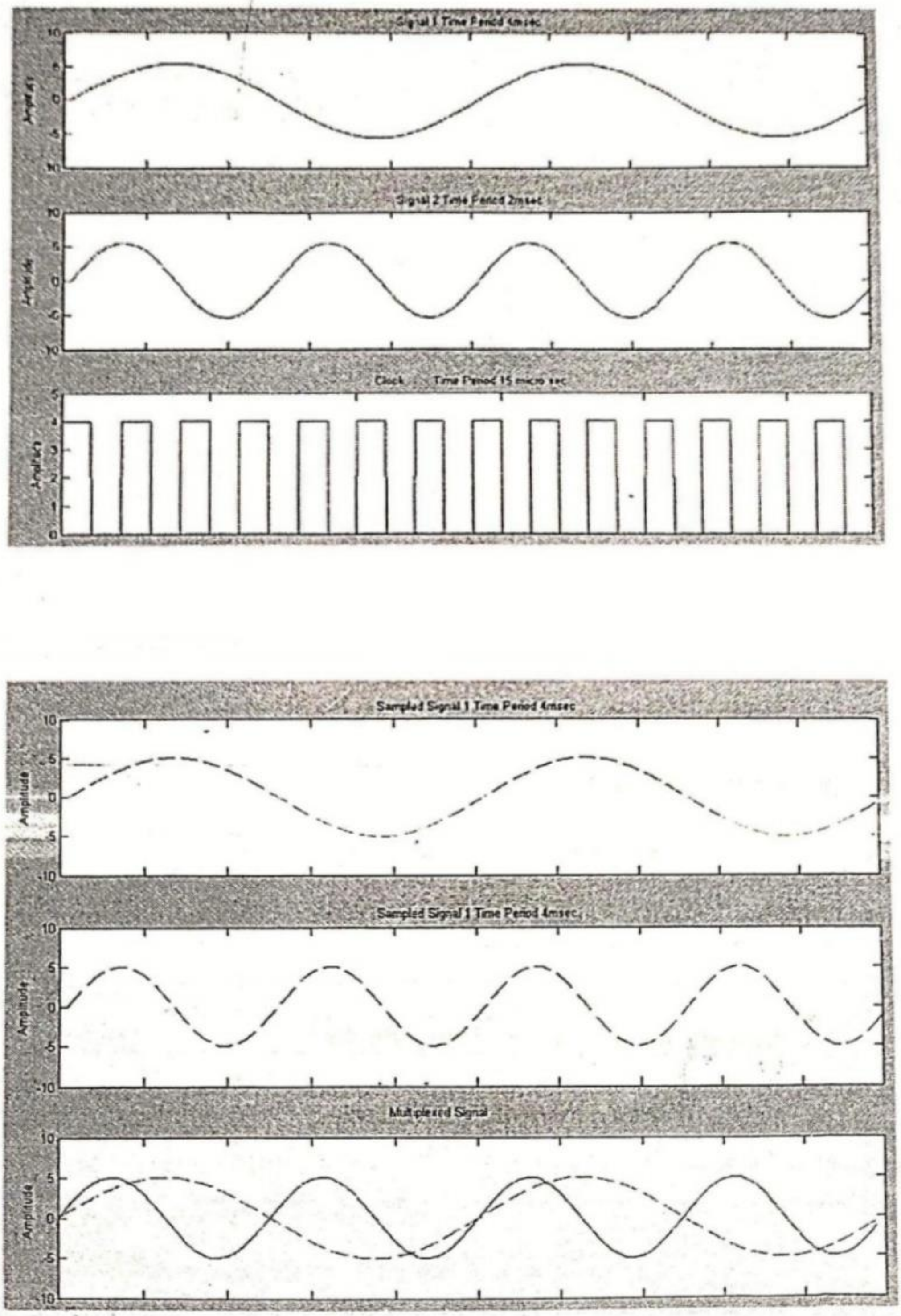


Figure 1.2 TDM Multiplexed Signal

Receiver section

MODEL GRAPH:

RECEIVER SECTION

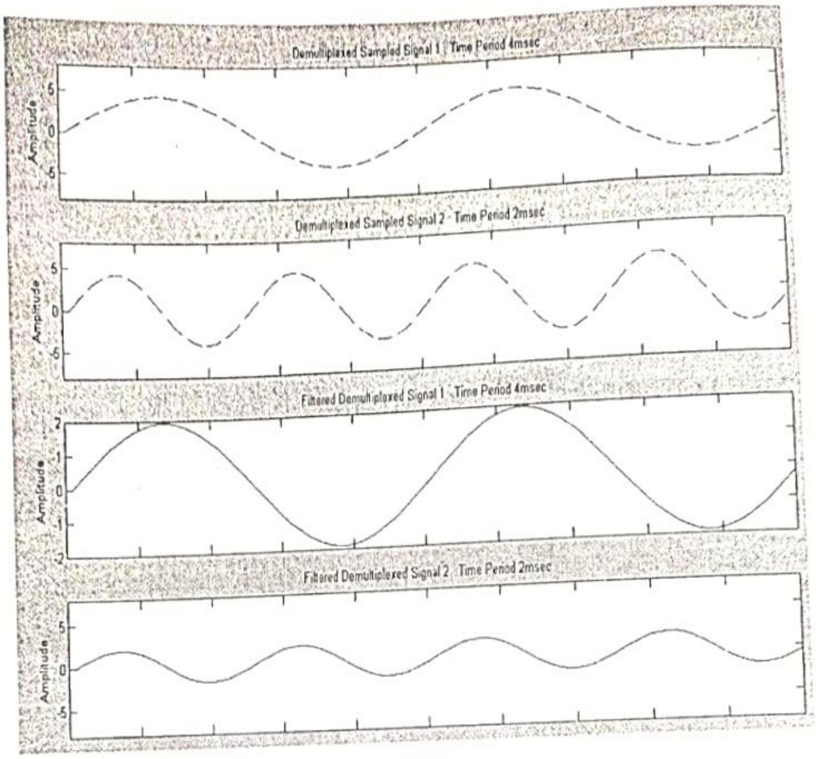
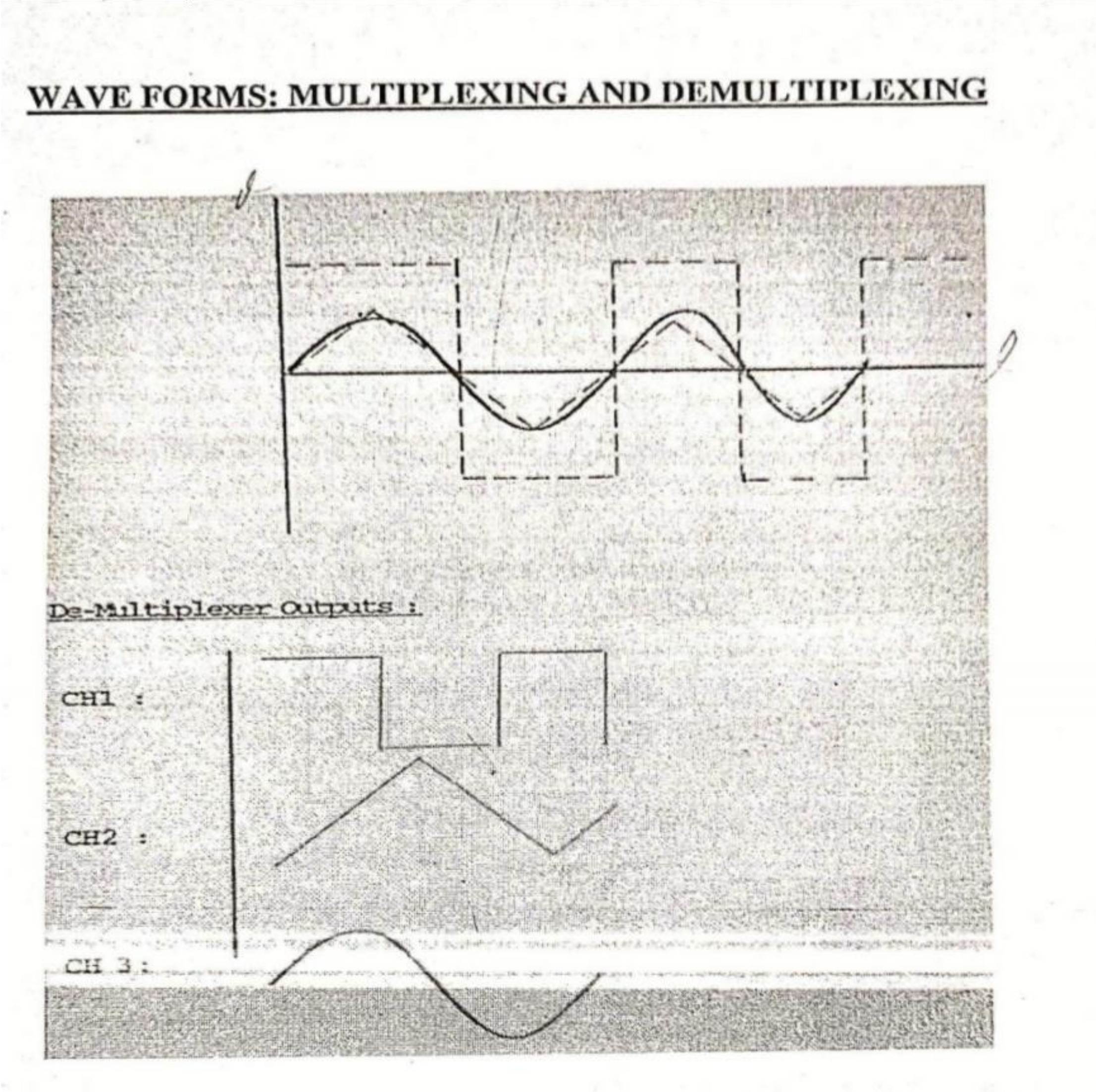


Figure 1.3 TDM Received Signal

1.8. OBSERVATION

Transmitter Section		Receiver Section	
Signal 1		**Demultiplexed Signal 1**	
Amplitude	**Time Period**	**Amplitude**	**Time Period**
Signal 2		**Demultiplexed Signal 2**	
Amplitude	**Time Period**	Amplitude	**Time Period**
Transmitter Output		**Filtered Demultiplexed Signal 1**	
Amplitude	**Time Period**	**Amplitude**	**Time Period**
		Filtered Demultiplexed Signal 1	
		Amplitude	**Time Period**



Procedure:

1. Switch on Time Division Multiplexing and De Multiplexing Trainer.
2. Connect the sine wave to channel-1, square wave to channel -2 and triangle wave to channel-3, terminals of 8 to 1 Multiplexer.
3. Observe the multiplexer output on channel -1 of a CRO.
4. Connect mux output to de-mux input.
5. Observe corresponding signal outputs at channel-2 of CRO.

Result

The operation of TDM is observed and the output waveforms are verified.

LAB RESULT

Time division multiplexing and de-multiplexing using PAM signals were performed and respective waveforms were plotted.

POST LAB QUESTIONS

1. Two signals g1(t) and g2(t) are to be transmitted over a common channel by means of time division multiplexing. The highest freq of g1(t) is 1 KHz and that g2(t) is 1.5 KHz. What is the minimum value of the permissible sampling rate? Justify your answer.
2. How is synchronization achieved in TDM?
3. Twenty four voice signals are sampled uniformly and then time division multiplexed. The sampling operation uses flat top samples with 1ps duration. The multiplexing operation includes provision for synchronization by adding an extra pulse of sufficient amplitude and also 1 us duration. The highest frequency component of cach voice signal is 3.4KHz.
   a. Assuming a sampling rate of 8' KHz, Find the spacing between successie pulses of the multiplexed signal.
   b. Repeat your calculation assuming the use of nyquist rate sampling.

4.What is the major drawback of digital communication?

5.Three signals ml,m2 and m3 are to be multiplexed, m1 and m2 have a 5 KHz bandwidth and m3 has 10KHz bandwidth. Design a commutator switching system so that each signal is sampled at its Nyquist rate..

6. Define bandwidth expansion factor.

Experiment 4: Study of Differential Coded Pulse Code Modulation.

OBJECTIVE:

Study of differential pulse code modulation technique

THEORY:

DPCM is a good way to reduce the bit rate for voice transmission. However it causes some other problems that deals with voice quality. DPCM quantizes and encodes the difference between a previous sample input signal and a current sample input signal. DPCM quantizes the difference signal using uniform quantization. Uniform quantization generates an SNR that is small for small input sample signals and large for large input sample signals. Therefore, the voice quality is better at higher signals.

The first part of DPCM works exactly like PCM (that is why it is called differential PCM). The input signal is sampled at a constant sampling frequency (more than the input frequency). Then these samples are modulated. At this point, the DPCM process takes over. The sampled input signals are stored in what is called a predictor. The predictor takes the stored sample signal and sends it through a differentiator. The differentiator compares the previous,sample signal and sends its difference to the quantizing and coding phase of PCM (this phase can be a uniform quantizing or companding with A-law or µ-law). After quantizing and coding, the difference signal is transmitted to its final destination. At the receiving end of the network, everything is reversed. First the difference signal is decoded and- dequantized. This difference is added to the sample signal stored in the predictor and send through a low-pass filter that reconstructs the original input signal.

EQUIPMENTS:

ADCL-07 Kit

Connecting Chords

Power Supply

20MHz Dual Trace Oscilloscope

NOTE: KEEP THE SWITCH FAULTS IN OFF POSITION.

PROCEDURE:

1. Refer to the block diagram (Fig.1) and carry out the following connections and switch settings.

2. Connect power supply in proper polarity to the kit ADCL-07 and switch it ON.

3. Keep the clock frequency at 512KHz, by changing the jumper position of JP1 in the clock generator section.

4. Keep the amplitude of the onboard sine wave, of frequency 500Hz to 1Vpp.
   DPCM modulation

5. Connect the 500Hz sine wave to the IN post of Analog Buffer.

6. Connect OUT post of Analog Buffer to IN post of DPCM modulator section.

7. Observe the sample output at the given test point. The input signal is sampled at the clock frequency of 16KHz.

8. observe the linear predictor output at the PREDICTED OUT post of the linear predictor in the DPCM modulator section.

9. Observe the diferential pulse code modulated data (DPCM) at the DPCM OUT post of the DPCM modulator section.

10. Observe the DPCM data at DPCM OUT post by varying input signal from 0 to 2V.

DPCM demodulation:

11. Connect the DPCM modulated data from the DPCM OUT post of the DPCM modulator to the IN post of the DPCM demodulator.
12. Observe the demodulated data at the output of summation block.
13. Observe the integrated demodulated data at the DEMOD OUT post of the DPCM demodulator.
14. Connect the demodulated data from the DEMOD OUT post of the DPCM demodulator to the IN post of the low-pass filter.
15. Observe the reconstructed signal at the OUT post of the filter. Use RST switch for clear observation of output.
16. Now, simultaneouslyreduce the clock frequencies from 512KHz to 256KHz, 128KHz and

64KHz by changing the jumper position of JP1 and observe the difference in the DPCM modulated and demodulated data. As the frequency of clock decreases, DPCM demodulated data at DEMOD OUT becomes distorted.
Observe various waveforms as mentioned below

OBSERVATION:

ON KITADCL-07

Observe the following waveforms on the oscilloscope and plot on the paper.

1) 500Hz, 1Vpp input sine wave.
2) Sampled out at the provided test point SAMPLER OUT.
3) Linear predictor out at PREDICTED OUT post.
4) DPCM data at DPCM OUT post.
5) Line interface out at the given output test point of line interface block in DPCM demodulator.
6) Demodulated DPCM data at the output test point of summation block in DPCM demodulator.
7) Integrated demodulated data at the DEMOD OUT post of the DPCM demodulator.
8) Reconstructed sine wave at the OUT post of the filter.
9) Observe the data at different clock rates.

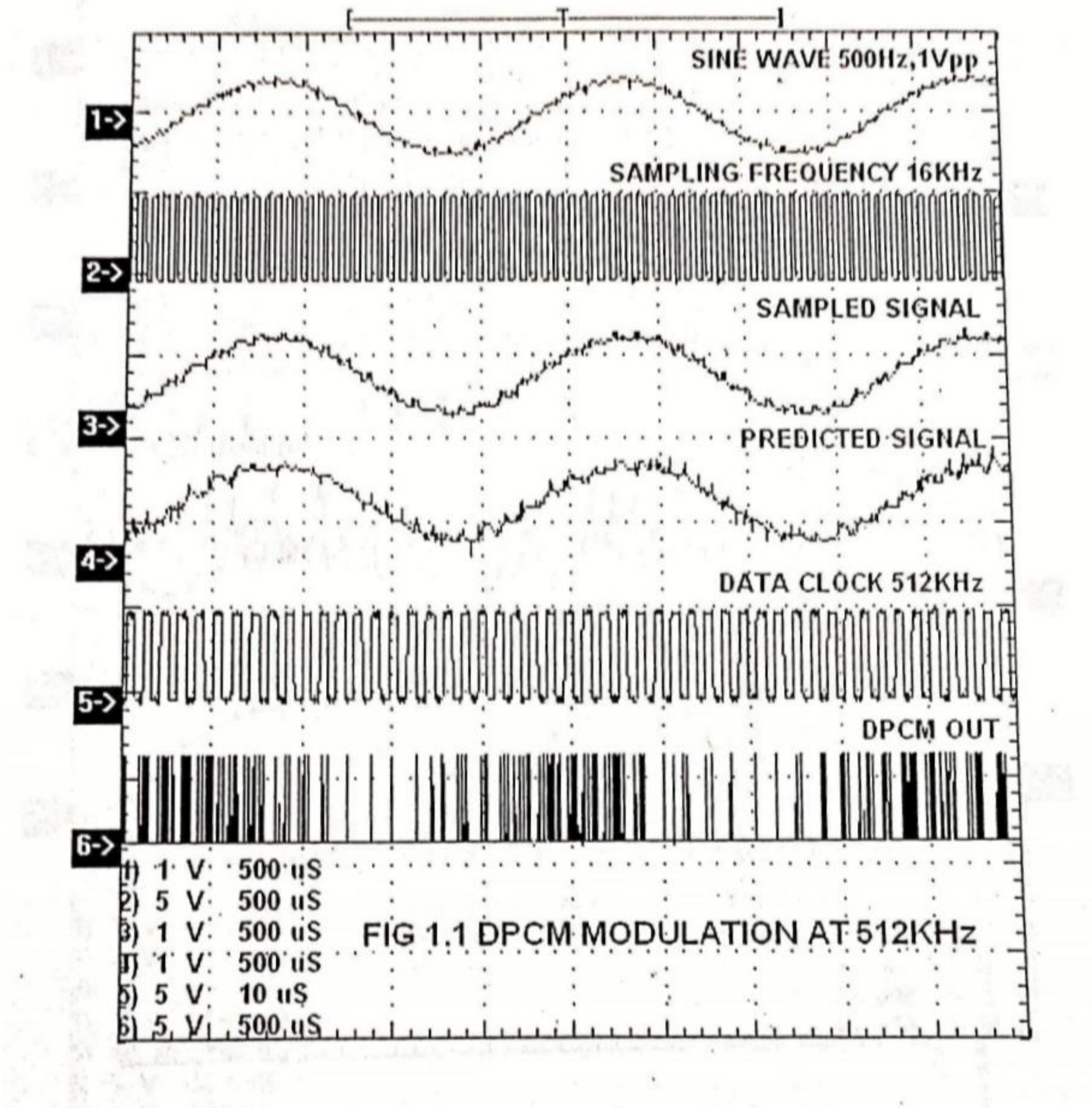


FIG 1.1 DPCM MODULATION AT 512KHz

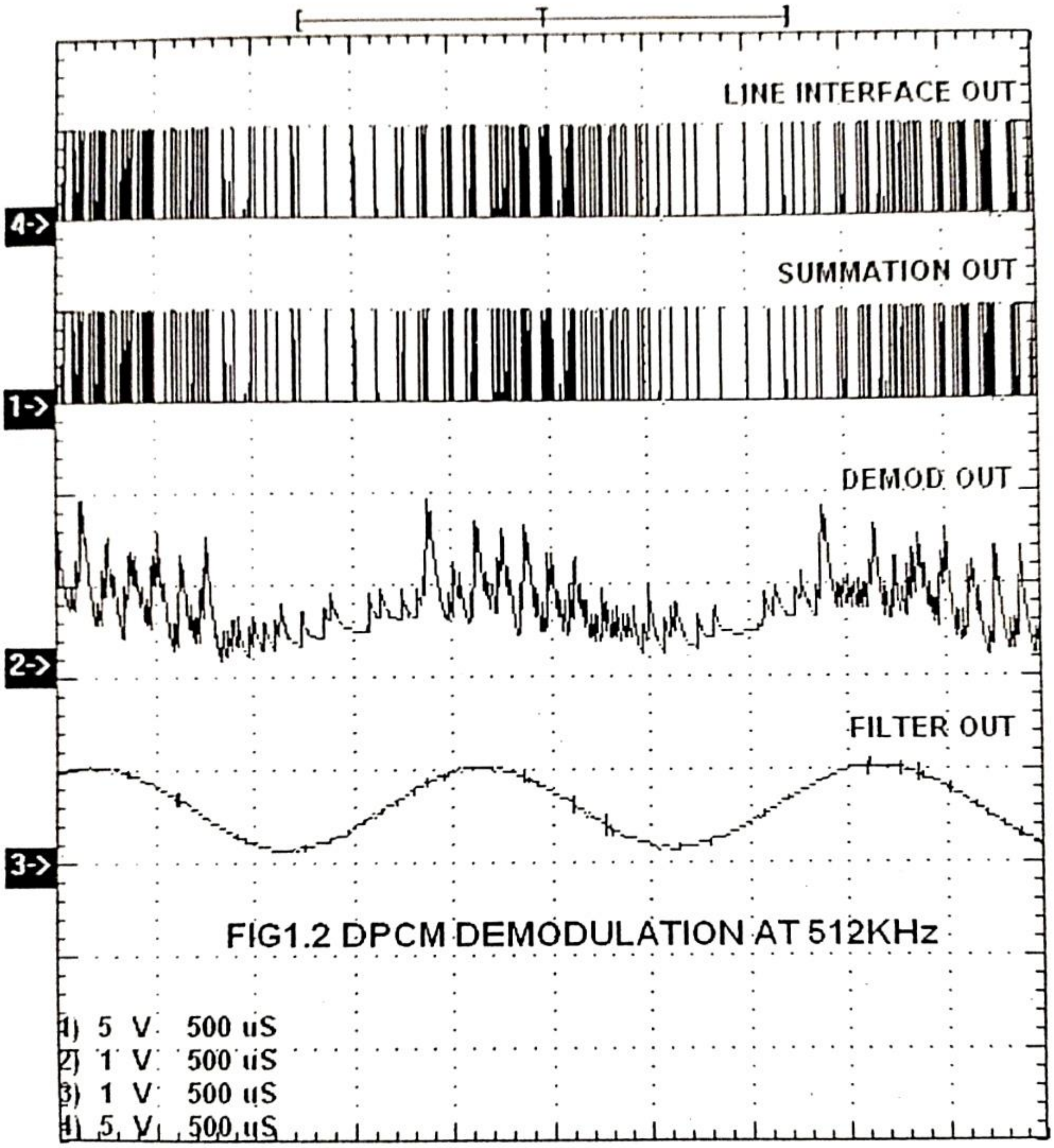


FIG1.2 DPCM DEMODULATION AT 512KHz

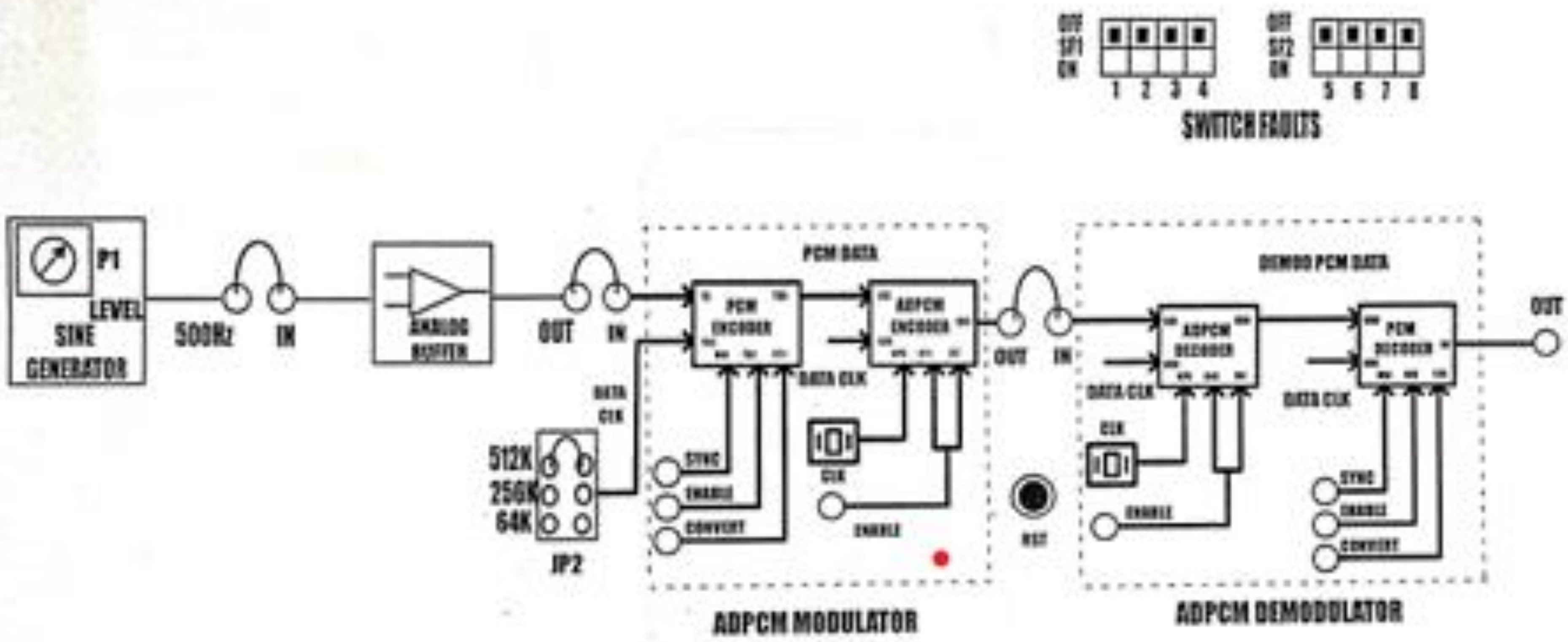


FIG. 2 BLOCK DIAGRAM FOR STUDY OF ADAPTIVE DIFFERENTIAL PULSE CODE MODULATION AND DEMODULATION

Experimnt 5: Experiment 5 Study of Adaptive Predictive Differential Coded Pulse Code Modulation

OBJECTIVE:
To study the adaptive differential pulse code modulation technique.
THEORY:
Since most of the signals generated by the human voice are small. Voice quality needs to focus on small signals. To solve this problem, adaptive DPCM is developed. Adaptive DPCM adapts the quantization levels of the difference signal that generated at the time of DPCM process. When the difference signal is low, ADPCM increases the size of the quantization levels. If the difference signal is high, ADPCM decreases the size of the quantization levels. So, ADPCM adapts the quantization level to the size of the input difference signal. This generates an SNR that is uniform through out the dynamic range of the difference signal. Using ADPCM reduces bit rate of voice transmission.

CODEC IC is used for analog to digital conversion. The whole process takes place inside the CODEC. Present techniques of voice communication standards such as A-law / μ-law companded PCM voice coding at 64kbps.If the amplitude of the signal is small; quantization levels have to be closely spaced. This gives proper resolution. But if the signal amplitude is large than this fine resolution will result in increasing the number of code bits. The analog input is fed into the CODEC; the output is in digital form. The digital data output is in Pulse Code Modulated form. The CODEC chips used exhibit both A-law and p-law companding techniques. Here, we select this A-law and μ-law companding with the help of a switch.

The PCM data is fed to the ADPCM CODEC IC. This CODEC is used to reduce the data rate required to transmit a PCM encoded analog / voice signal while maintaining the voice fidelity and intelligibility of the PCM signal. It also uses a filter to attempt to predict the next PCM input value.based on previous PCM input is values. The error between the predicted and the true PCM input value is the information that is sent to the other end of the line. The adaptive filter adapts to the statistics of the signals presented to it. Here also the whole process takes place inside the CODEC.

CODEC IC is used for ADPCM demodulation also. The ADPCM data is fed into this CODEC along with clocks for synchronization. Demodulated PCM data is derived. This PCM data is fed into the PCM CODEC filter IC. The received digital data is again converted into analog form by the chip.

EQUIPMENTS:

ADCL-07 Kit
Connecting Chords
Power Supply
20MHz Dual Trace Oscilloscope
NOTE: KEEP THE SWITCH FAULTS IN OFF POSITION.

PROCEDURE:

ADPCM modulation:

Refer to the block diagram and carry out the following connections and switch settings.

1. Connect power supply in proper polarity to the kit ADCL-07 and switch it ON.

2. Keep the clock frequency at 512KHz, by changing the jumper position of JP2 in the clock generator section.

3. Keep the amplitude of the onboard sine wave, of frequency 500Hz to 1Vpp.

4. Connect the 500Hz sine wave to the IN post of Analog Buffer.

5. Connect the OUT post of buffer to the IN post of ADPCM modulator section.

6. Observe the PCM data at the given test point in the ADPCM modulator section

7. Observe the ADPCM data in DUAL mode at the OUT post of ADPCM modulator section with sync pulse and with input sine wave.

ADPCM demodulation:

8. Connect the ADPCM demodulated data from the OUT post of the ADPCM modulator to the IN post of the ADPCM demodulator.

9. Observe the demodulated PCM data at the provided test point. Observe the reconstructed sine wave at the OUT post of the ADPCM demodulator. Use RST switch for clear observation of output.

10.Now, simultaneously reduce the clock frequencies from 512KHz to 256KHz and 64KHz by changing the jumper position of JP2 and observe the difference in the data and the reconstructed sine wave. As the frequency of clock decreases, DPCM modulated data at filter OUT becomes distorted.

11.Observe various waveforms as mentioned below.

OBSERVATION:

ON KIT ADCL-07

Observe the following waveforms on oscilloscope and plot on the paper.

1. Input sine wave 500Hz, 1Vpp.
2. PCM data at different frequencies at the given test point on the ADPCM modulator section.
3. ADPCM data at different frequencies the OUT post of the ADPCM modulator section.
4. Demodulated PCM data at different frequencies at the given test point of ADPCM demodulator section.
5. Reconstructed sine wave at different frequencies at the OUT post of the ADPCM demodulator section.

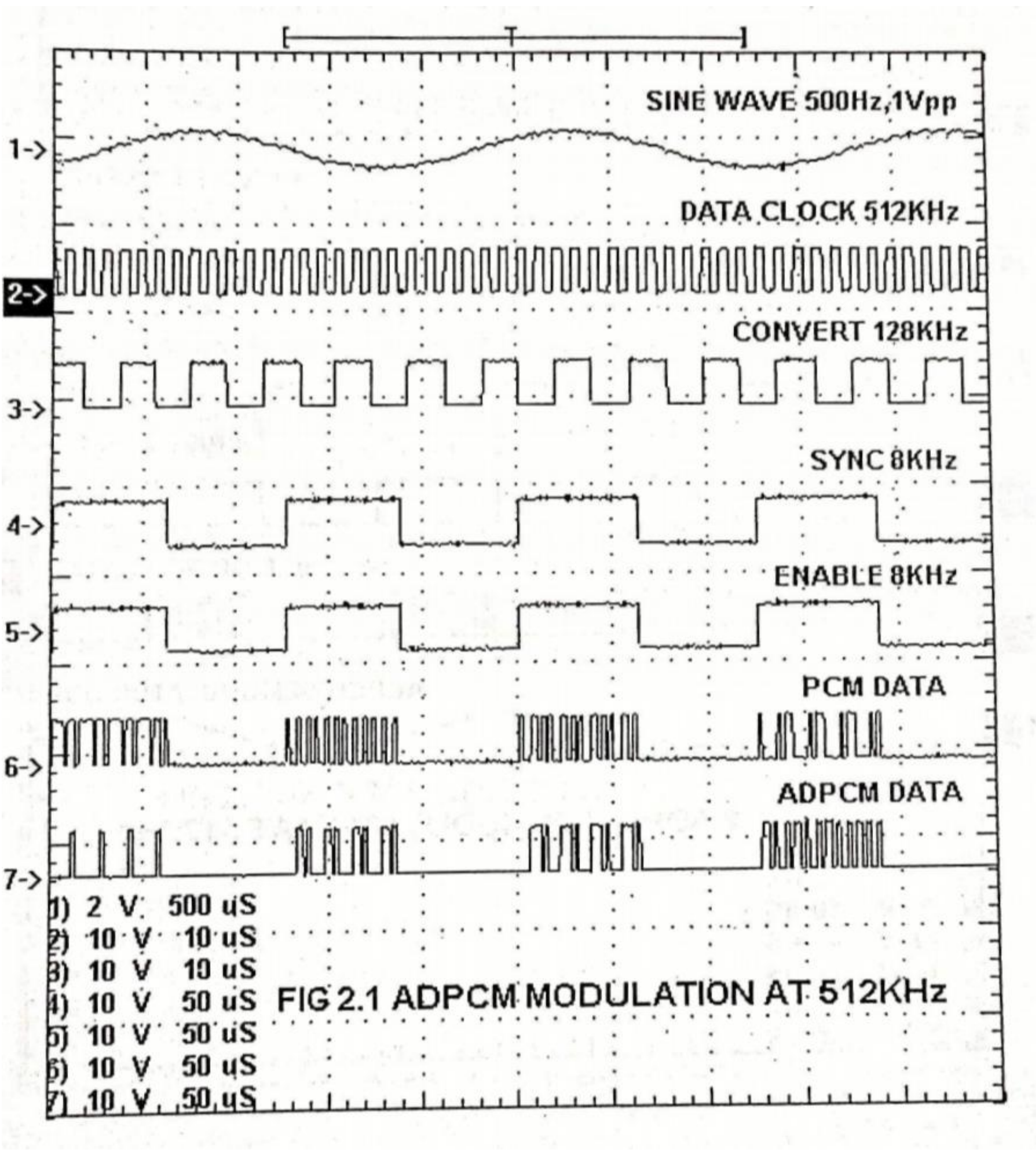


FIG 2.1 ADPCM MODULATION AT 512KHz

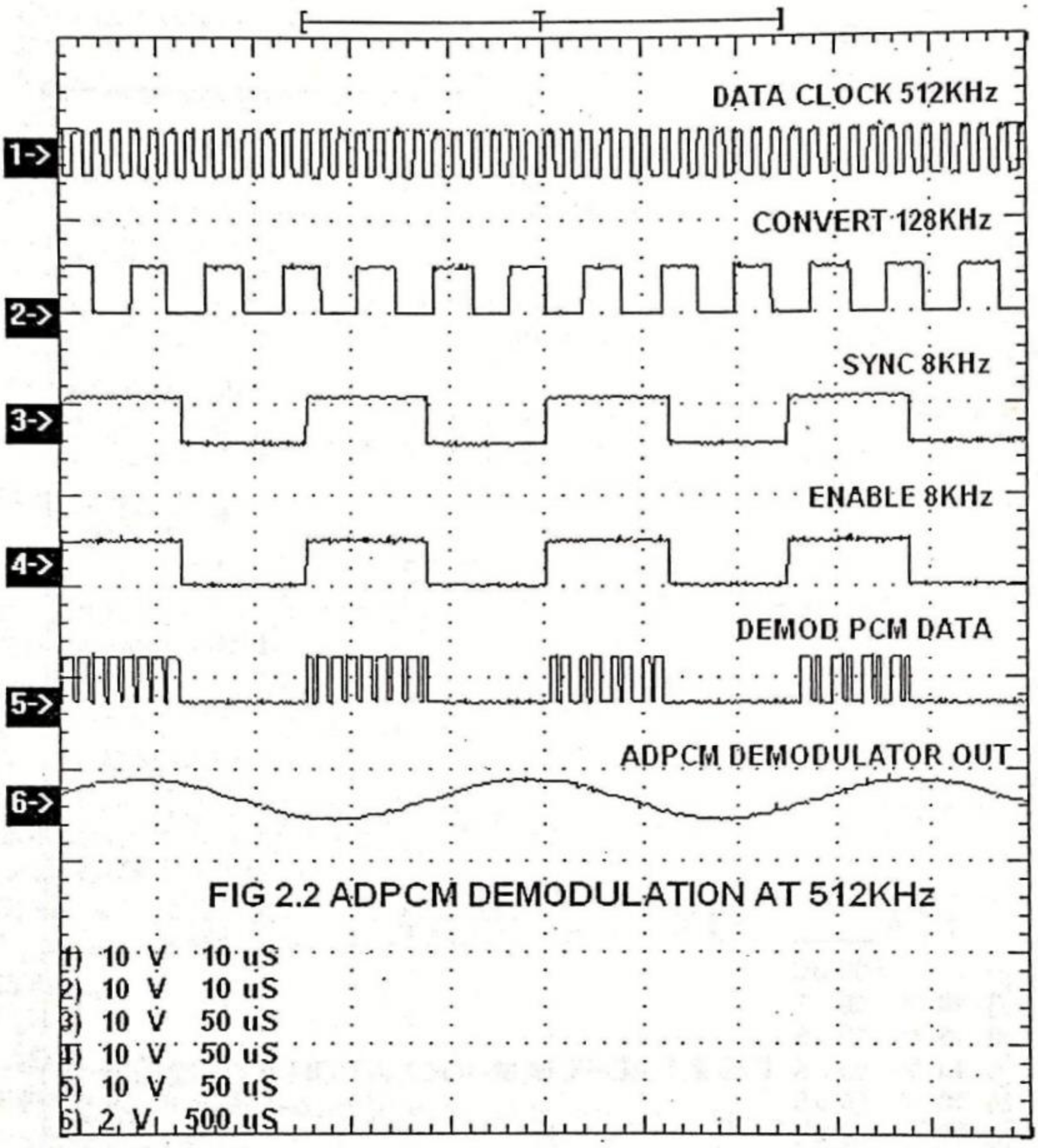


FIG 2.2 ADPCM DEMODULATION AT 512KHz

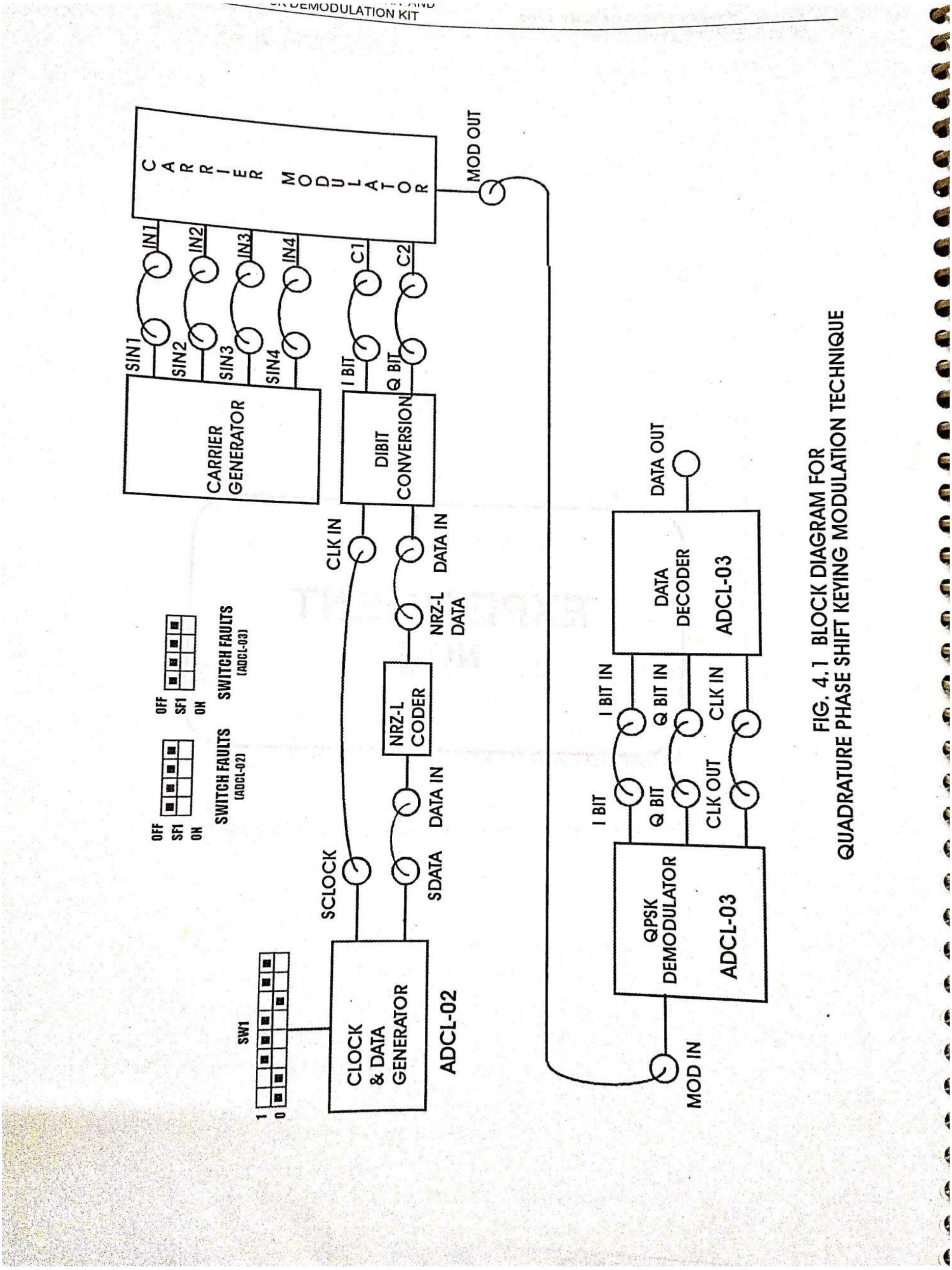


FIG. 4.1 BLOCK DIAGRAM FOR QUADRATURE PHASE SHIFT KEYING MODULATION TECHNIQUE

clock having different phases. At the output of adder the signals consisting the envelope corresponds to the I & Q bit. Envelope detector then filters the high Frequency components and recovers I & Q bit. These recovered 1 & Q bits having exactly same phase & frequency compared to transmitter | & Q bit. These I & Q bits then applied to data decoder logic to recover the original NRZ-L data pattern.

EQUIPMENTS:

Experimenter Kits ADCL-02 & ADCL-03.

Connecting Chords.

Power supply.

20MHz Dual Trace Oscilloscope.

NOTE: KEEP THE SWITCH FAULTS IN OFF POSITION.

PROCEDURE:

1. Refer to the block diagram (Fig.4.1) and carry out the following connections and switch settings.
2. Connect power supply in proper polarity to the kits ADCL-02 and ADCL-03 and switch it on.
3. Select Data pattern of simulated data using switch SW1.
4. Connect SDATA generated to DATA IN of the NRZ-L CODER
5.Connect NRZ-L DATA to DATA IN of the DIBIT CONVERSION
6. Connect SCLOCK to CLK IN of the DIBIT CONVERSION.
7. Connect the dibit data I & Q bit to control input C1 and C2 of CARRIER MODULATOR respectively.

NOTE : Adjust I & Q bit as shown in Fig.4.2A by operating RST Switch on ADCL-02 before connecting it to C1 & C2

8. Connect carrier component to input of CARRIER MODULATOR as follows:

a. SIN 1 to IN 1

b. SIN 2 to IN 2

C.SIN 3 to IN 3

d. SIN 4 to IN 4

9. Connect QPSK modulated signal MOD OUT on ADCL-02 to the MOD IN Of the QPSK DEMODULATOR on ADCL-03. NOTE : Adjust Recovered | 8 Q bit on ADCL-03 as per ADCL-02 by RST Switch on ADCL-03.
10.Connect I BIT, Q BIT & CLK OUT outputs of QPSK Demodulator to I BIT IN, Q BIT IN & CLK IN posts of Data Decoder respectively.

11.Observe various waveforms as mentioned below (Fig. 4.2).

NOTE: If there is mismatch in input & Recovered Data, then adjust that Data by RST Switch on ADCL-03.

OBSERVATION:

Observe the following waveforms on oscilloscope and plot it on the paper.

ON KIT ADCL-02

1. Input NRZ-L Data at DATA INPUT
2. Carrier frequency SIN 1 to SIN4
3. Dibit pair generated data I bit & Q bit at DIBIT CONVERSION
4. QPSK modulated signal at MOD OUT.

ON KIT ADCL-03

1. Output of first squarer at SQUARER 1.
2. Output of second squarer at SQUARER 2.

Experiment 6: Study of Phase Shift Keying modulation techniques.

QUADRATURE PHASE SHIFT KEYING MODULATION TECHNIQUES.

OBJECTIVE:

Study of Carrier Modulation Techniquos by Quadrature Phase Shift Keying method.

THEORY:

In this modulation, called Quadraturo PSK (QPSK) or 4 PSK the sine carrier takes 4 phase values, soparated of 90 dog. and dotormined by the combinations of bit pair (Dibit) of the binary data signal. Tho data aro codod into Dibit by a circuit generating:

1. A data signal I (in phase) consisting in voltage levels corresponding to the value of the first bit of the considerod pair, for duration equal to 2 bit intervals.
2. A data signal Q (in quadrature) consisting in voltage levels corresponding to the value of the second bit of the pair, for duration equal to 2 bit intervals.

The block diagram of the modulator used on the module is shown in the fig.4.1 four 500KHz sine carriers, shifted between them of 90 deg, are applied to modulator. The data (signal | & Q) reach the modulator from the Dibit generator The instantaneous value of I and Q data bit generates a symbol. Since I and Q can take either 0 or 1 value, maximum 4 possible symbols can be generated (00,01,10. and 11). According to the symbol generated one of the four-sine carrier will be selected. The relation between the symbol generated and sine carrier is shown in table.

DIBIT	PHASE SHIFT
00	180 deg
01	90 deg
10	270 deg
11	0 deg

A receiver for the QPSK signal is shown in fig. synchronous detection is required and hence it is necessary to locally regenerate the carriers. The scheme for carrier regeneration is similar to that employed in BPSK. In that earlier case we squared the incoming the signal, extracted the waveform at twice the carrier frequency by filtering, and recovered the carrier by frequency dividing by two. In the present case, it is required that the incoming signal be raised to the fourth power after which filtering recovers a waveforms at four times the carrier. The incoming signal also applied to the sampler followed by an adder and envelope detectors. Two adders add the sampled QPSK signal, sampled by the clock having different phases. At the output of adder the signals consisting the envelope corresponds to the | & Q bit. Envelope detector then filters the high bits then applied to data decoder logic to recover the original NRZ-L data pattern.

EQUIPMENTS

Experimenter Kits ADCL-02 & ADCL-03.
Connecting Chords.
Power supply.
20MHz Dual Trace Oscilloscope.

NOTE: KEEP THE SWITCH FAULTS IN OFF POSITION.
PROCEDURE:

1. Refer to the block diagram (Fig.4.1) and carry out the following connections and switch settings.
2. Connect power supply in proper polarity to the kits ADCL-02 and ADCL-03 and switch it on.
3. Select Data pattern of simulated data using switch SW1.
4. Connect SDATA generated to DATA IN of the NRZ-L CODER.
5. Connect NRZ-L DATA to DATA IN of the DIBIT CONVERSION
6. Connect SCLOCK to CLK IN of the DIBIT CONVERSION.
7. Connect the dibit data I & Q bit to control input C1 and C2 of CARRIER MODULATOR respectively. NOTE : Adjust I & Q bit as shown in Fig.4.2A by operating RST Switch on ADCL-02 before connecting it to C1 & C2.
8. Connect carrier component to input of CARRIER MODULATOR as follows:

a. SIN 1 to IN 1
b. SIN 2 to IN 2
c. SIN 3 to IN 3
d. SIN 4 to IN 4

9. Connect QPSK modulated signal MOD OUT on ADCL-02 to the MOD IN Of the QPSK DEMODULATOR on ADCL-03. NOTE : Adjust Recovered |&Q bit on ADCL-03 as per ADCL-02 by RST Switch on ADCL-03.
10.Connect I BIT, Q BIT & CLK OUT outputs of QPSK Demodulator to I BIT IN, Q BIT IN & CLK IN posts of Data Decoder respectively.
11.Observe various waveforms as mentioned below (Fig.4.2). NOTE: If there is mismatch in input & Recovered Data, then adjust that Data by RST Switch on ADCL-03.

OBSERVATION:

Observe the following waveforms on oscilloscope and plot it on the paper.
ON KIT ADCL-02
1. Input NRZ-L Data at DATA INPUT.
2. Carrier frequency SIN 1 to SIN 4
3. Dibit pair generated data I bit & Q bit at DIBIT CONVERSION
4. QPSK modulated signal at MOD OUT.
ON KIT ADCL-03
1. Output of first squarer at SQUARER 1.
2. Output of second squarer at SQUARER 2.
3. Four sampling clocks at the output of SAMPLING CLOCK GENERATOR.
4. Two adder outputs at the output of ADDER.
5. Recovered data bits (I & Q bits) at the output of ENVELOP DETECTORS.
6. Recovered NRZ-L data from I & Q bits at the output of DATA DECODER.

SWITCH FAULTS:
Note: Keep the connections as per the procedure. Now switch corresponding fault switch button in ON condition & observe the different effect on the output. The faults are normally used one at a time.
1. Put switch 1 of SF1 (ADCL-02) in Switch Fault section to ON position. This will open capacitor for filtering of SIN 1. Thus amplitude of SIN 1 and SIN 3 gets reduced.
2. Put switch 2 of SF1 (ADCL-02) in Switch Fault section to ON position. This will disable control signal C1 going to Modulator IC. Modulator will not able to modulate the signal properly.
3. Put switch 3 of SF1 (ADCL-02) in Switch Fault section to ON position. This will open the input of EX OR gate used in differential encoder 1. Due to this random data is generated at the output of differential encoder 1.

4. Put switch 4 of SF1 (ADCL-02) in Switch Fault section to ON position. This will remove the clock signal (125 KHz-180 deg.) in the generation of Q bit data. This disable the generation of Q bit data at the output of dibit conversion.
5. Put switch 2 of SF1 (ADCL-03) in Switch Fault section to ON position. This will remove Pull up resistor from envelope detector of I-Bit. I-Bit generation gets disabled.
6. Put switch 3 of SF1 (ADCL-03) in Switch Fault section to ON position. This will remove one of the sampling clocks to sampler. Thus QPSK signal doesn't get sampled properly and due to this QPSK demodulated data also gets disturbed.
7. Put switch 4 of SF1 (ADCL-03) in Switch Fault section to ON position. This will remove the sampling input of sample. So I bit can not be observed and recovered data also gets disturbed.

CONCLUSION:

In BPSK we deal individually with each bit of duration Tb. In QPSK we lump two bits together to form a SYMBOL. The symbol can have any one of four possible values corresponding to two-bit sequence 00, 01, 10, and 11. We therefore arrange to make available for transmission tour distinct sianals. At the receiver each signal represents one symbol and, correspondingly, two bits. When bits are transmitted, as in BPSK, the signal changes occur at the bit rate. When symbols are transmitted the changes occur at the symbol rate which is one-half the bit rate. Thus the symbol time is $T_s = 2T_b$.

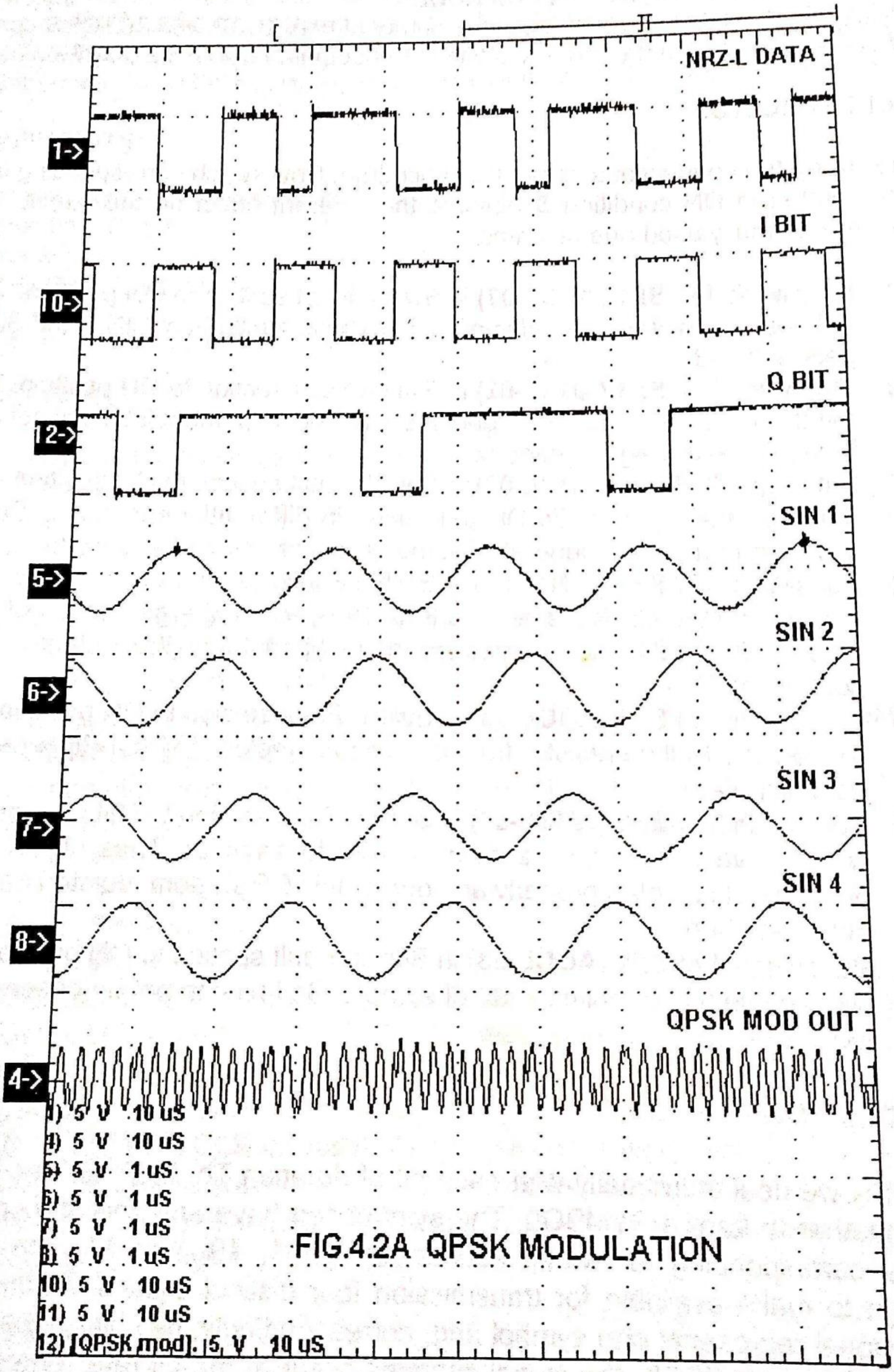


FIG.4.2A QPSK MODULATION

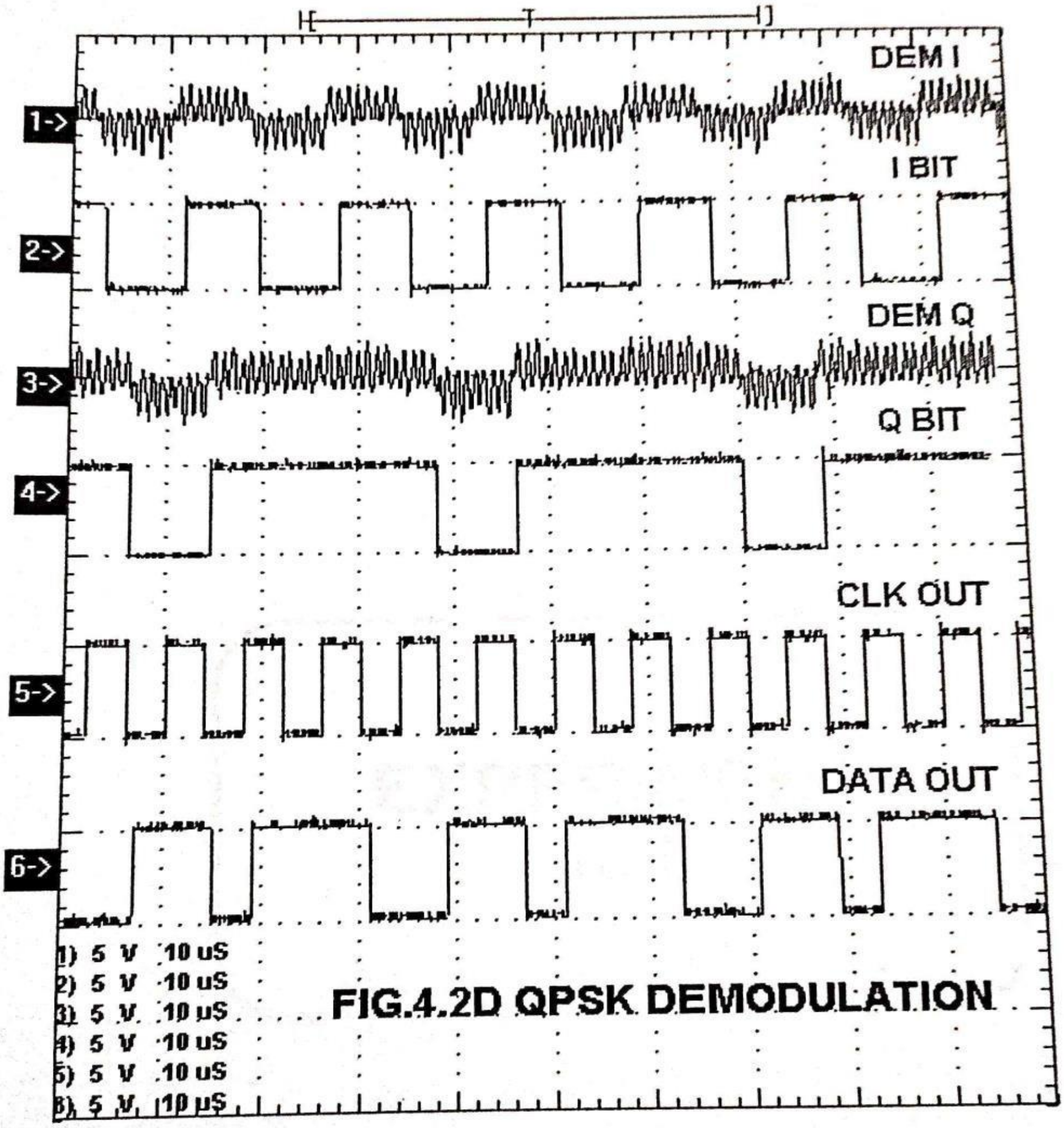


FIG.4.2D QPSK DEMODULATION