Ahsanullah University of Science and Technology

Department of Electrical and Electronic Engineering

LABORATORY MANUAL
FOR
ELECTRICAL AND ELECTRONIC SESSIONAL COURSES

Student Name:
Student ID:

Course no: - EEE-3208
Course title: - Communication Theory Lab

For the students of
Department of Electrical and Electronic Engineering
3rd Year, 2nd Semester

Experiment no: **1**

Name of the Experiment: **Study of Amplitude Modulation (AM) and Demodulation. [Double Side Band (DSB) Transmission and Reception]**

(a) Objective:

1. To understand the various aspects of amplitude modulation.
2. To generate and observe the DSB-AM and DSBSC-AM signal.
3. To study the demodulation of DSB signal at the receiver.

(b) Equipments:

1) AM Trainer board (Anacom 1/1, Anacom1/2)
2) Oscilloscope
3) Power Supply

(c) Theory:

1. Modulation:

The process of modulation implies varying some characteristics of a high frequency sinusoidal voltage (called carrier voltage) in accordance with the instantaneous values of message voltage (Called the modulating voltage),

2. Need for modulation:

The size of the antenna conductor is inversely proportional to the carrier frequency. Higher the carrier frequency, the smaller the size and hence economical is the antenna structure needed. Further, higher the carrier frequency, the better is the selection of signal in the receiver. If there are several stations operating over this same frequency range, the program of different stations will get mixed up. Thus in order to keep the various signals separate, it is necessary to translate or shift them to different frequency spectrum. Each station is allocated a band of frequency. This also overcomes the drawbacks of poor radiation efficiency at low frequency.

3. Amplitude Modulation (AM):

In amplitude modulation, the amplitude of the carrier wave varies in accordance with the instantaneous values of the modulating voltage.
Amplitude modulation produces a signal with power concentrated at the carrier frequency and in two adjacent sidebands. 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. DSBSC 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-sideband modulation, widely used in amateur radio due to its efficient use of both power and bandwidth.

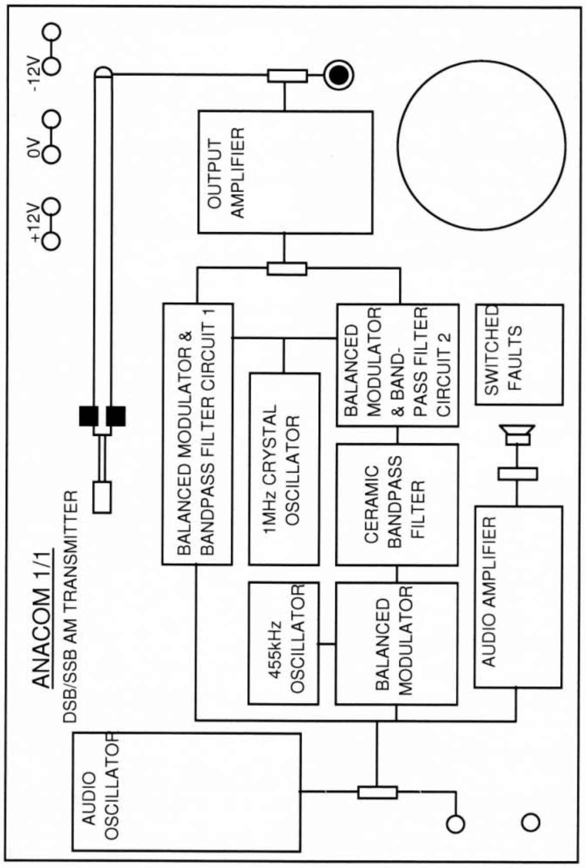


Fig 1.1(a):- Block diagram of the Anacom 1/1 DSB/SSB AM transmitter

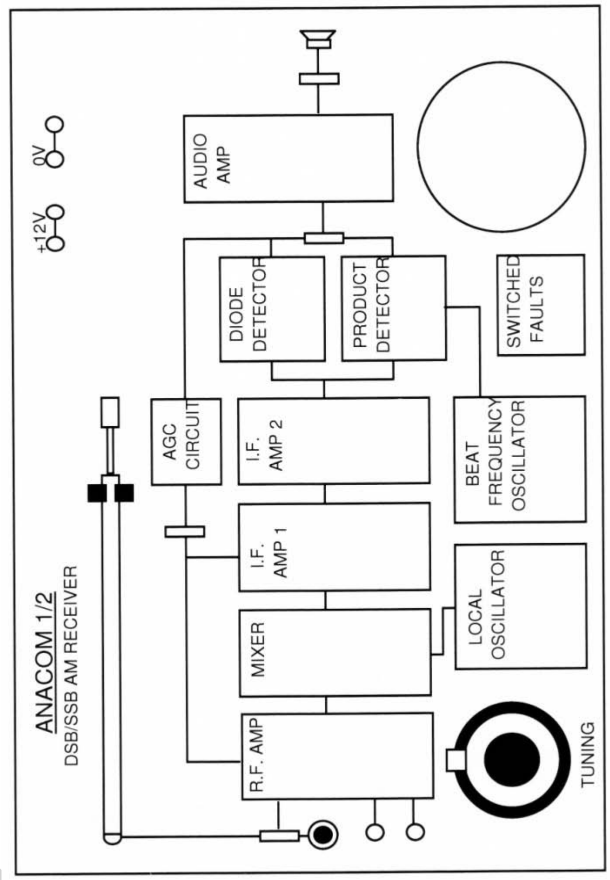


Fig 1.1(b):- Block diagram of the Anacom 1/2 DSB/SSB AM receiver

(d) Experimental Procedure:

GENERATION OF DOUBLE SIDEBAND AM WAVEFORMS

1. Connect the ANACOM 1/1 module to the power supply.
2. Ensure that the following initial conditions exist on the board:
   (a) AUDIO INPUT SELECT switch in INT position.
   (b) MODE switch in DSB position.
   (c) OUTPUT AMPLIFIER'S GAIN preset in fully clockwise position.
   (d) SPEAKER switch in OFF position.
3. Turn on power to the ANACOM 1/1 board.
4. Turn the AUDIO OSCILLATOR block's AMPLITUDE preset to its fully clockwise (MAX) position, and examine the block's output (t.p. 14) on an oscilloscope.
   This is the audio frequency sinewave which will be used as our modulating signal. The sinewave's frequency can be adjusted from about 300Hz to approximately 3.4 kHz, by adjusting the AUDIO OSCILLATOR'S FREQUENCY preset. The amplitude of this audio modulating signal can be reduced to zero, by turning the AUDIO OSCILLATOR'S AMPLITUDE preset to its fully counter clockwise (MIN) position.
5. Examine the block's output (t.p. 9) on an oscilloscope. Test point 9 carries a sinewave of frequency 1MHz and amplitude 120mV pk/pk approx.

6. Next, examine the output of the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block (at t.p.3), together with the modulating signal at t.p.1 Trigger the oscilloscope on the

t.p.1 signal. It is this block that we, will use to perform **double-sideband amplitude modulation.**

Check that the waveforms are as shown below:

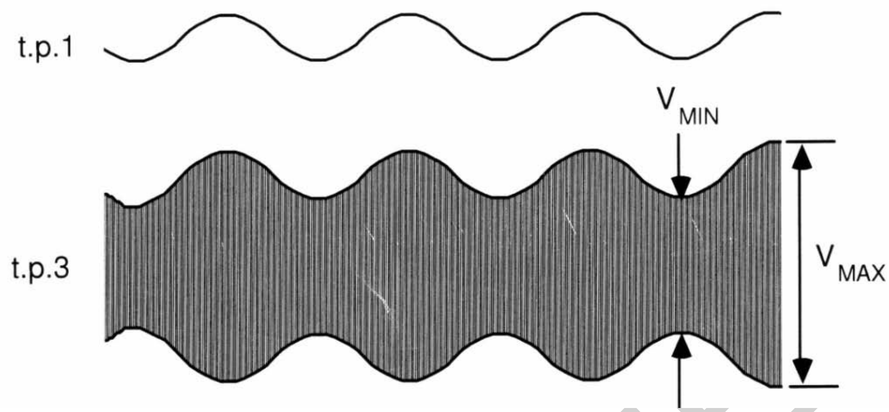


Fig:-1.2

The frequency spectrum of this AM waveform is as shown where fm is the frequency of the audio modulating-signal.

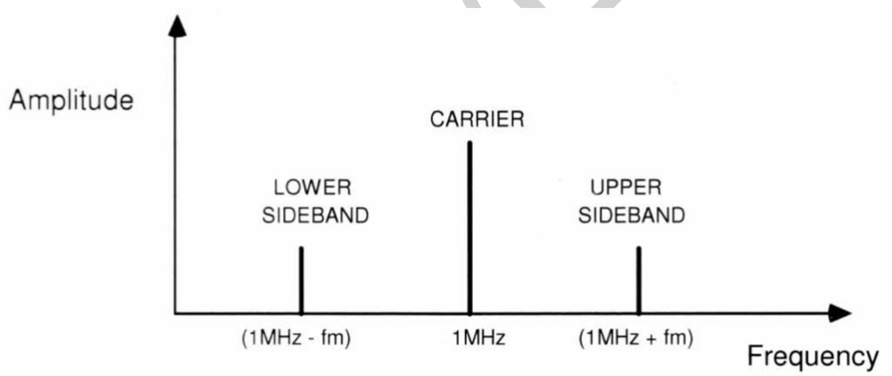


Fig:-1.3

7. To determine the depth of modulation, measure the maximum amplitude (V_{MAX}) and the minimum amplitude (V_{MIN}) of the AM waveform at t.p.3, and use the following formula:

$$Percentage\ Modulation = \frac{V_{MAX} - V_{MIN}}{V_{MAX} + V_{MIN}} \times 100\%$$

8. Now vary the amplitude and frequency of the audio-frequency sinewave, by adjusting the AMPLITUDE and FREQUENCY presets in the AUDIO OSCILLATOR block. Note the effect that varying each preset has on the amplitude modulated waveform.
9. Now turn the BALANCE preset in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, until the signal at t.p.3 is as shown below:

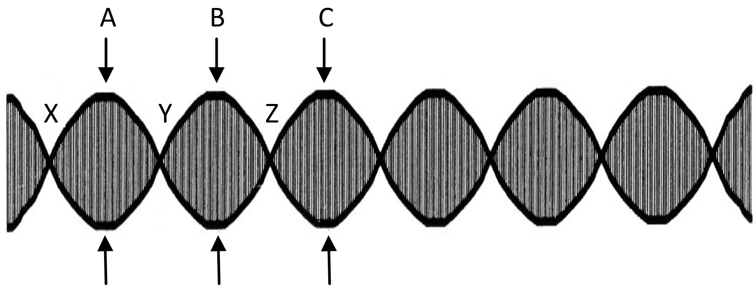


Fig:-1.4

The BALANCE preset varies the amount of the 1 MHz carrier component which is passed through to the modulator's output.
By adjusting the preset until the peaks of the waveform (A,B,C and so on) have the same amplitude, we are removing the carrier component altogether. We say that the carrier has been 'balanced out' (or 'suppressed'), to leave only the two sidebands.
The waveform at t.p.3 is known as a double-sideband suppressed carrier (DSBSC) waveform, and its frequency spectrum is as shown below:

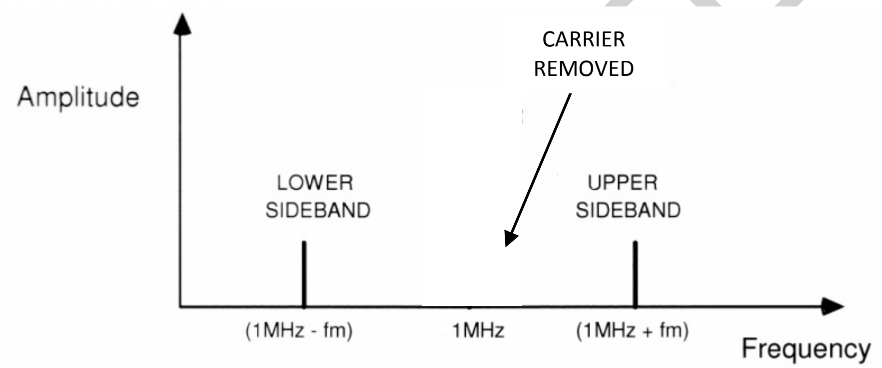


Fig 1.5:- The frequency spectrum of the DSB-SC waveform

10. Examine the output from the OUTPUT AMPLIFIER block (t.p. 13), together with the audio modulating signal (at t.p. 1), triggering the 'scope with the latter. Note that the DSBSC waveform appears amplified slightly, at t.p.13.

RECEPTION OF DOUBLE SIDEBAND AM WAVEFORMS Basic principles of operation of the ANACOM 1/2 receiver:

1. Position the ANACOM 1/1 and 1/2 modules, with the ANACOM 1/1 board on the left, and a gap of about three inches between them. Then connect them to the power supply.
2. Ensure that the following initial conditions exist on both boards:

AM Transmitter	AM Receiver
1. AUDIO OSCILLATOR'S AMPLITUDE preset in fully clockwise position. Amplitude of audio osc. (Max)	1. RX INPUT SELECT switch in ANT. Position.
2. AUDIO INPUT SELECT switch in INT position.	2. R.F. AMPLIFIER'S TUNED CIRCUIT SELECT switch in INT position.
3. MODE switch in DSB position	3. R.F. AMPLIFIER'S GAIN preset in fully clockwise position.
4. BALANCE preset in BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, in fully clockwise position.	4. AGC switch in IN position.

5. OUTPUT AMPLIFIER'S GAIN preset in fully counter-clockwise position.	5. DETECTOR switch in DIODE position.
6. AUDIO AMPLIFIER'S VOLUME preset in fully counter-clockwise position.	6. AUDIO AMPLIFIER'S VOLUME preset in fully counter-clockwise position.
7. TX OUTPUT SELECT switch in ANT. Position.	7. SPEAKER switch in ON position.
8. SPEAKER switch in ON position.	8. BEAT FREQUENCY OSCILLATOR switch in OFF position.
9. On-board antenna in vertical position, and fully extended.	9. On-board antenna in vertical position, and fully extended.

3. Turn on power to the modules.
4. On the ANACOM 1/2 module, slowly turn the AUDIO AMPLIFIER'S VOLUME preset clockwise, until sounds can be heard from the on-board loudspeaker.
   Next, turn the vernier TUNING dial until a broadcast station can be heard clearly, and adjust the VOLUME control to a comfortable level.
5. The first stage, or 'front end', of the ANACOM 1/2 AM Receiver is the R.F. AMPLIFIER stage. This is a wide-bandwidth tuned amplifier stage, which is tuned into the wanted station by means of the TUNING dial.
   Once it has been tuned into the wanted station, the R.F. AMPLIFIER, having little selectivity, will not only amplify the wanted frequency, but also those frequencies which are close to the wanted frequency. As we will see later, these nearby frequencies will be removed by subsequent stages of the receiver, to leave only the wanted signal.
   Examine the envelope of the signal at the R.F. AMPLIFIER'S output (at t.p.12), with an a.c. coupled oscilloscope channel. Note that:
   (a) The amplifier's output signal is very small in amplitude (a few tens of millivolts at the most). This is because one stage of amplification is not sufficient to bring the signal's amplitude up to a reasonable level.
   (b) Only a very small amount of amplitude modulation can be detected, if any. This is because there are many unwanted frequencies getting through to the amplifier's output, which tend to 'drown out' the wanted AM signal.
   You may notice that the waveform itself drifts up and down on the 'scope display, indicating that the waveform's average level is changing. This is due to the operation of the AGC circuit, which will be explained later.
6. The frequency of the LOCAL OSCILLATOR is also tuned by means of the TUNING dial, and is arranged so that its frequency is always 455 kHz above the signal frequency that the R.F. AMPLIFIER is tuned to. This fixed frequency difference is always present, irrespective of the position of the TUNING dial, and is known as the Intermediate Frequency (I.F. for short).
   This frequency relationship is shown below, for some arbitrary position of the TUNING dial:

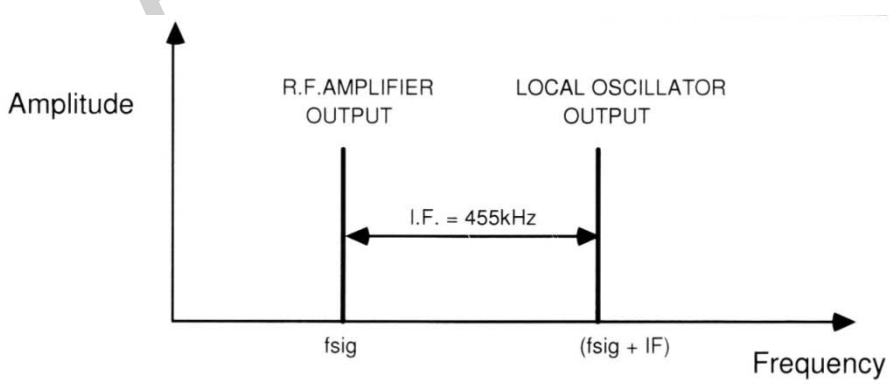


Fig:-1.6

Examine the output of the LOCAL OSCILLATOR block, and check that its frequency varies as the TUNING dial is turned. Re-tune the Receiver to a radio station before continuing.

7. The operation of the MIXER stage is basically to shift the wanted signal down to the I.F. frequency, irrespective of the position of the TUNING dial. This is achieved in two stages:
    (a) by mixing the LOCAL OSCILLATOR'S output sinewave with the output from the R.F. AMPLIFIER block. This produces three frequency components:

    The local oscillator frequency = (fsig + IF)
    The sum of the original two frequencies, f_{sum} = (2fsig + IF)
    The difference between the original two frequencies, f_{diff} = (fsig + IF - fsig) = IF.
    These three frequency components are shown below:

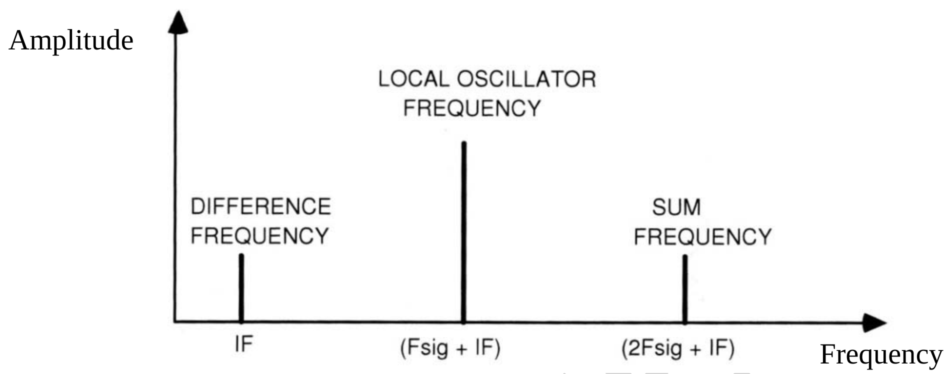


Fig:-1.7

    (b) By strongly attenuating all components except the difference frequency, IF. This is done by putting a narrow-bandwidth bandpass filter on the mixer's output.

    The end result of this process is that the carrier frequency of the selected AM station is shifted down to 455 kHz (the IF. frequency), and the sidebands of the AM signal are now either side of 455 kHz.
8. Note that, since the mixer's band pass filter is not highly selective, it will not completely remove the LOCAL OSCILLATOR and SUM frequency components from the mixer's output. This is the case particularly with the LOCAL OSCILLATOR component, which is much larger in amplitude than the SUM and DIFFERENCE components. What we need to do now is to preferentially amplify frequencies around 455 kHz, without amplifying the higher-frequency LOCAL OSCILLATOR and SUM components.

    This selective amplification is achieved by using two I.F. AMPLIFIER stages, I.F. AMPLIFIER 1 and I.F. AMPLIFIER 2, which are designed to amplify strongly a narrow band of frequencies around 455 kHz, without amplifying frequencies on either side of this narrow band.

    Examine the output of I.F. AMPLIFIER 1 (at t.p.24) and I.F. AMPLIFIER 2 (t.p.28) with an a.c. coupled oscilloscope channel, and note that:
    (a) The overall amplitude of the signal is much larger than the signal amplitude at the MIXER's output, indicating that voltage amplification has occurred.
    (b) The dominant component of the signal is now at 455 kHz, irrespective of which station you have tuned into. This implies that the wanted signal, at the I.F. frequency, has been amplified to a level where it dominates over the unwanted components.

9. The next step is to extract this audio information from the amplitude variations of the signal at the output of I.F. AMPLIFIER 2. This operation is performed by the DIODE DETECTOR block, whose output follows the changes in the amplitude of the signal at its input.

    To see how this works, examine the output of the DIODE DETECTOR block (t.p.31), together with the output from I.F. AMPLIFIER 2 (at t.p.28). Note that the signal at the DIODE DETECTOR'S output:
    (a) Follows the amplitude variations of the incoming signal, as required;
    (b) Contains some ripple at the I.F. frequency of 455 kHz, and
    (c) Has a positive D.C. offset, equal to half the average peak-to-peak amplitude of the incoming signal. We will see how we make use of this offset later on, when we look at Automatic Gain Control (AGC).

10. The final stage of the receiver is the AUDIO AMPLIFIER block. The block contains a simple low-pass filter which passes only audio frequencies and removes the high-frequency ripple from the DIODE DETECTOR'S output signal. This filtered audio signal is applied to the input of an audio power amplifier which drives the on-board loudspeaker (and the headphones, if these are used). The final result is the sound you are listening to.
    The audio signal which drives the loudspeaker can be monitored at t.p.39 (providing that the AUDIO AMPLIFIER block's VOLUME preset is not in its minimum volume position). Compare this signal with that at the DIODE DETECTOR'S output (t.p.31), and note how the AUDIO AMPLIFIER block's low-pass filter has 'cleaned up' the audio signal.

Reception of A.M. signal from the ANACOM 1/1 Transmitter

11. When, the gain of ANACOM 1/1's OUTPUT AMPLIFIER block is-zero, so that there is no output from the Transmitter. Now turn the GAIN preset in ANACOM 1/1's OUTPUT AMPLIFIER block to its fully clockwise (maximum gain) position, so that the Transmitter generates an AM signal.
    On the ANACOM 1/1 module, examine the Transmitter's output signal (t.p. 13), together with the audio modulating signal (t.p.1), triggering the 'scope with the audio signal.
    Since ANACOM 1/1's TX OUTPUT SELECT switch is in the ANT. position, the A.M. signal at t.p.13 is fed to the transmitter's antenna. It can be proven by touching ANACOM 1/1's antenna, and noting that the loading caused by hand reduces the amplitude of the A.M. waveform at t.p. 13.The antenna will propagate this AM signal over a maximum distance of about 4 feet.
12. On the ANACOM 1/1 module, turn the VOLUME preset (in the AUDIO AMPLIFIER block) clockwise, until you can hear the tone of the AUDIO OSCILLATOR'S output signal, from the on-board loudspeaker. Turn the VOLUME preset to the fully counter-clockwise (minimum volume) position before continuing.
13. On the ANACOM 1/2 Receiver, adjust the VOLUME preset so that the receiver's output can be clearly heard. Then adjust the receiver's TUNING dial until the tone generated at the Transmitter is also clearly audible at the Receiver (This should be when the TUNING dial is set to about 55-65), and adjust the Receiver's VOLUME preset until the tone is at a comfortable level.
    Check that you are tuned into the Transmitters output signal, by varying ANACOM 1/1's FREQUENCY preset (in the AUDIO OSCILLATOR block), and noting that the tone generated by the Receiver changes.
    The ANACOM 1/2 Receiver is now tuned into the AM signal generated by the ANACOM 1/1 Transmitter. Briefly check that the waveforms, at the outputs of the following Receiver blocks, are as expected:

R.F. AMPLIFIER	(t.p.12)
MIXER	(t.p.20)
I.F. AMPLIFIER 1	(t.p.24)
I.F. AMPLIFIER 2	(t.p.28)
DIODE DETECTOR	(t.p.31)
AUDIO AMPLIFIER	(t.p.39)

14. We will now investigate the operation of the Receiver's AGC (Automatic Gain Control) Circuit. The AGC circuit prevents the receiver from overloading when it is tuned into a strong A.M. -broadcast signal, and provide the d.c. level at diode detector's output about 0.7 volts when it is below about 0.7 volts by monitoring the d.c. bias voltage at the output of the DIODE DETECTOR.
    The overall result is that the average amplitude of the AM signal at the input to the DIODE DETECTOR is maintained at a constant level. This has two advantages:
    (a) The receiver cannot overload, even when the incoming AM signal is very strong;
    (b) Providing the incoming signal strength is sufficient to bring the receiver's AGC CIRCUIT into operation, then the AGC CIRCUIT will compensate for slow fluctuations in signal strength at the receiver's input, by maintaining the audio output from the DIODE DETECTOR at a constant level.
    To examine its behavior, monitor the output of the DIODE DETECTOR (at t.p.31). Note that the d.c. offset at the DIODE DETECTOR'S output is about +0.7 volts, indicating that the incoming signal to the receiver is a strong one. Check that neither of the monitored signals shows any sign of overloading.

Now turn the GAIN preset, in ANACOM 1/1's OUTPUT AMPLIFIER block, slowly counter-clockwise, thereby reducing the strength of the transmitted signal. Note that the monitored signals do not decrease in amplitude, until the GAIN preset is almost in its fully counter-clockwise (minimum gain) position. This is because the AGC CIRCUIT will compensate for changes in the strength of the incoming signal, so long as the signal strength is large enough to keep the AGC CIRCUIT in operation.

15. We will now prevent the AGC CIRCUIT from controlling the gain of the Receiver, by disconnecting it from the R.F. AMPLIFIER and I.F. AMPLIFIER 1 blocks. Do this by putting the Receiver's AGC switch in the OUT position, and note the effect on the two monitored waveforms.

    Providing the Receiver has been tuned in correctly, the results of removing AGC should be:

    (a) Overloading of the signal at the DIODE DETECTOR'S input, so that all amplitude variations are removed.
    (b) No audio signal at the DIODE DETECTOR'S output.
    (c) A d.c. level of greater than 0.7 volts at the DIODE DETECTOR'S output.

(e) Report:

Should include

1. Block diagram of transmitter and receiver block.
2. Comment on the function of various blocks.
3. Draw the wave shapes at various test points and explain.
4. Calculation of percentage modulation.
5. Comment on the effect of varying amplitude and frequency of audio input on the DSB.
6. Anything else instructed by teacher.
7. Discussion expressing originality of the student.

Experiment no: **2**

Name of the Experiment: **Study of Amplitude Modulation (AM) and Demodulation. [Single Side Band (SSB) Transmission and Reception]**

(a) Objective

1. To understand the theory of SSB modulation.
2. To understand the waveforms and frequency spectrum of SSB modulators.
3. To design and implement the SSB modulators.
4. To understand the measurement and adjustment of SSB modulators.
5. To understand the theory of SSB demodulator.
6. To design and implement the SSB demodulators.
7. To understand the measurement and adjustment of SSB demodulators.

(b) Equipments

1. AM Trainer board (ACS5-1, ASC6-1)
2. Oscilloscope
3. Power supply

(c) Theory

AM-SSB (Single Side Band) means the transmission of only one sideband with the exclusion of other side band and the carrier. It utilizes the fact that the intelligence or message is contained in each sideband and not in the carrier. However for demodulation purpose, carrier is necessary at the receiver. Hence in SSB system carrier is reinserted at the receiving end in proper phase, frequency and amplitude.

Advantages of SSB system

Following are the main advantages of SSB system over double side band (DSB) system:

1. 9 to 12 dB improvement in signal to noise ration.
2. Half bandwidth per channel.
3. Elimination of distortion due to selective fading (fading means reduction in strength of the received signal).
4. Reduction of carrier interference with other stations.
5. Reduction in operating cost.

The Operating Principle of SSB Modulator

We can use the DSB modulation to obtain the SSB modulation. We utilize two DSB-SC modulators and let the phase difference between the two audio signals and carrier signal be 90 degree, i.e. $(\text{DSB-SC})_Q$ and $(\text{DSB-SC})_I$, as shown in equation (1) and (2)

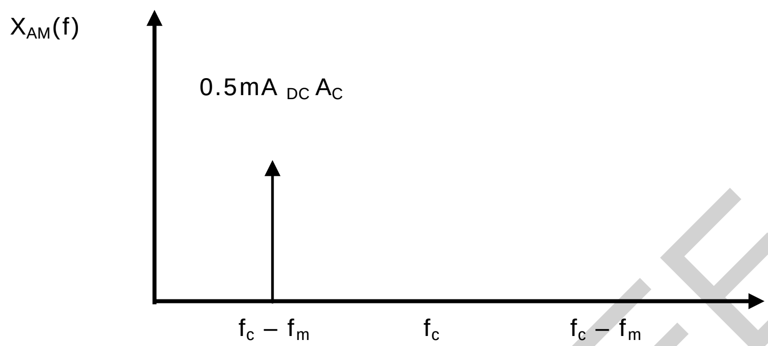
$$(DSB\_SC)_I = \cos 2\pi(f_c - f_m)t + \cos 2\pi(f_c + f_m)t \qquad (1)$$

$$(DSB\_SC)_Q = \cos 2\pi(f_c - f_m)t - \cos 2\pi(f_c + f_m)t \qquad (2)$$

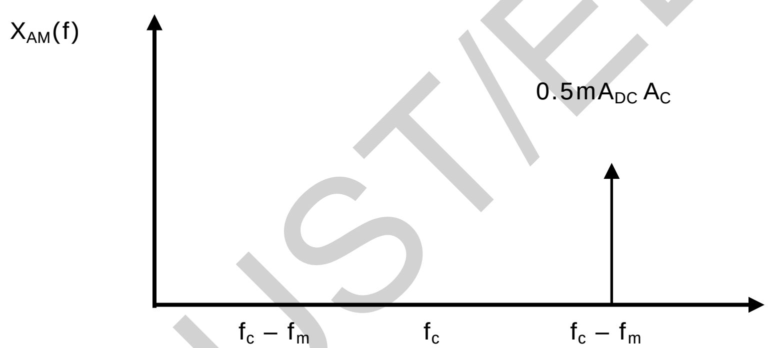
Equation (1) and (2) show that both $(\text{DSB-SC})_Q$ and $(\text{DSB-SC})_I$ signals connect to the adder, then we can obtain USSB or LSSB signals at the output port.

$$X_{LSSB} = (DSB\_SC)_I + (DSB - SC)_Q \qquad (3)$$
$$= \cos 2\pi\,(f_c - f_m)t$$
$$X_{USSB} = (DSB\_SC)_I - (DSB\_SC)_Q \qquad (4)$$
$$= \cos 2\pi\,(f_c + f_m)t$$

Fig 1.1(a) and Fig 1.1(b) are the frequency spectrum of SSB signal. We can see that the frequency spectrum consists of either $f_c - f_m$ signal $f_c + f_m$ signal.



(a) Frequency spectrum of SSB



(b) Frequency spectrum of SSB

Fig 2.1:- Different frequency spectrum of AM modulation

Therefore, during transmission, the power consumption of SSB modulation is less than DSB-SC modulation. From the above-mentioned discussion, we know that the sequence of power consumption of the three different types of modulation is AM>DSB-SC>SSB

The Operating Principle of SSB *Demodulator*

As for the SSB signal, it can be divided into upper and lower SSB signal. The expressions are as follow

$$X_{USSB} = DSB_P\text{-}DSB_Q \qquad (5)$$
$$=\cos 2\pi(f_c+f_m)t$$
$$X_{LSSB} = DSB_P + DSB_Q \qquad (6)$$
$$=\cos 2\pi(f_c- f_m)t$$

Where:

$DSB_P = k \cos(2\pi f_m t) \cos(2\pi f_c t)$

$DSB_Q = k \sin(2\pi f_m t) \sin(2\pi f_c t)$

f_c=frequency of carrier signal, f_m=frequency of audio signal, k=gain of the multiplier or mixer.

If equations (5) and (6) is multiplied $2\cos(2\pi f_c t)$, then we get

$$y_{DU}(t) = kx_{USSB}(t)[2\cos(2\pi f_c t)]$$

$$=0.5[2\cos 2\pi(f_c + f_m)t][2\cos(2\pi f_c t)] \qquad (7)$$

$$= \cos(2\pi f_m t) + \cos 2\pi(2f_c + f_m)t)$$

or

$$y_{DL}(t) = kx_{LSSB}(t)[2\cos(2\pi f_c t)]$$

$$=0.5[2\cos 2\pi(f_c - f_m)t][2\cos(2\pi f_c t)] \qquad (8)$$

$$= \cos(2\pi f_m t) + \cos 2\pi(2f_C - f_m)t)$$

When $y_{DU}(t)$ and $y_{DL}(t)$ through a low pass filter, whose frequency bandwidth equals or greater than the frequency bandwidth of m(t), but smaller than $2f_c$, then we get

$$X_D(t) = \cos(2\pi f_m t) \qquad (9)$$

From the equation (5) to (9) we know that the synchronous demodulator in fig:-2.2 can recover the m(t) signal from the SSB signal.

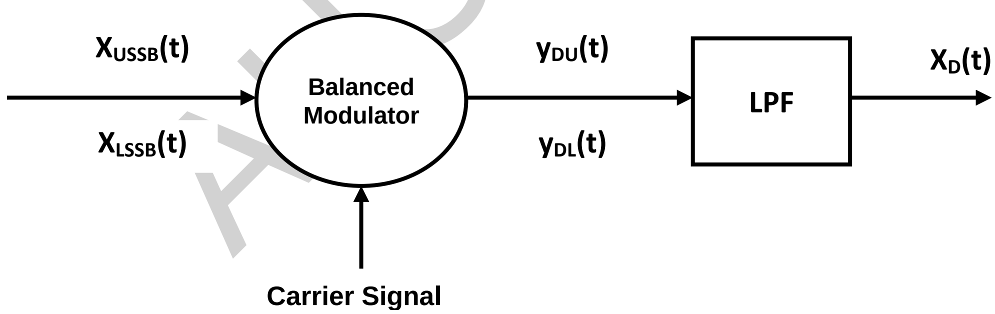


Fig 2.2:- Block diagram of synchronous demodulator

On the other hand, if we consider the phase difference θ(t) between the carrier signals of the carrier signals of the demodulator and modulator, then this situation will cause the signal distortion and the demodulator is unable to recover the original audio signal.

$$y_{DU}(t) = kx_{USSB}(t)[2\cos(2\pi f_c t + \theta(t))]$$

$$=0.5[2\cos 2\pi(f_c + f_m)t][2\cos(2\pi f_c t + \theta(t)] \qquad (10)$$

$$= \cos(2\pi f_m t - \theta(t)) + \cos[2\pi(2f_c + f_m)t + \theta(t)]$$

or

$$y_{DL}(t) = kx_{LSSB}\,(t)[\,2\cos\,(2\pi f_c t+\theta(t))]$$

$$=0.5[2\cos\,2\pi(f_c\text{-}f_m)t]\,[2\cos\,(2\pi f_c t+\theta(t)] \qquad (11)$$

$$= \cos\,[2\pi f_m t + \theta(t)]+ \cos\,[2\pi(2f_c\,\text{-}f_m)t+\theta(t)]$$

Therefore, when you(t) pass through the low-pass filter, then we get

$$x_D(t) = \cos\,[2\pi f_m t - \theta(t)] \qquad (12)$$

$$= \cos\,(2\pi f_m t)\,\cos\theta(t) + \sin\,(2\pi f_m t)\,\sin\theta(t)$$

Or, when $y_{DL}(t)$ pass through the low-pass filter, then we get

$$x_D(t) = \cos\,[2\pi f_m t + \theta(t)]$$

$$= \cos\,(2\pi f_m t)\,\cos\theta(t) - \sin\,(2\pi f_m t)\,\sin\theta(t) \qquad (13)$$

From equations (12) and (13), we know that if the phase difference between the carrier signals of the demodulator and modulator equals to each other, then $x_D(t) = \cos\,(2\pi f_m t)$.This situation indicates that the audio signal can be recovered. If the phase difference is not zero, then we notice that the demodulated signal will distort and unable to recover to the original audio signal.

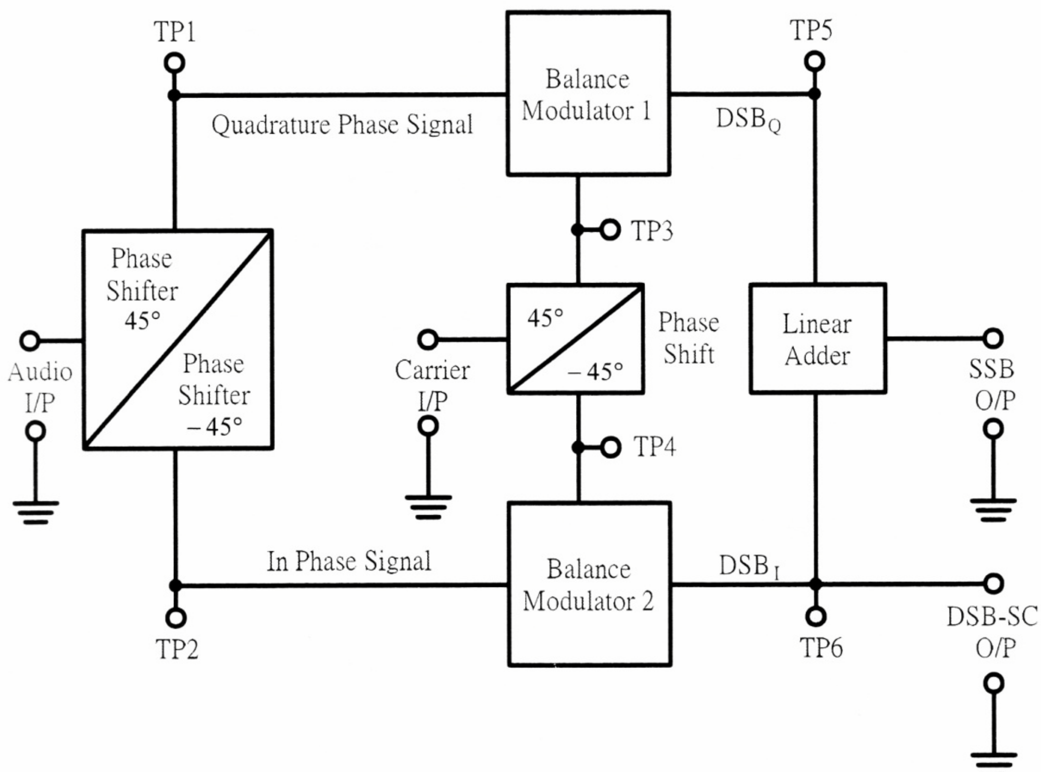


Fig 2.3:-Block Diagram of SSB modulator (ACS 5-1)

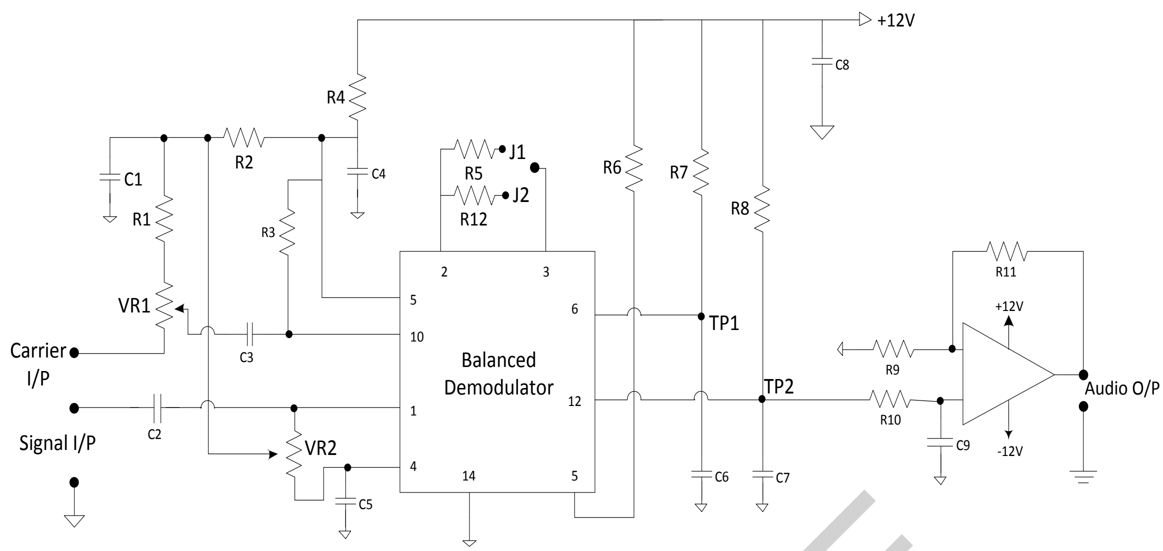


Fig 2.4:-Circuit Diagram of SSB Demodulator (ASC 6-1)

(d) Experimental Procedure

GENERATION OF SINGLE SIDEBAND AM WAVEFORMS

1. To implement a SSB-SC modulator refers to ACS5-1 module (ETEK trainer board).
2. At the audio signal input port (Audio I/P), input a signal of 300 mV amplitude and 2 kHz sine wave frequency. Then at the carrier signal input port (Carrier I/P), input a signal of 300 mV amplitude and 200 kHz sine wave frequency
3. By using oscilloscope, observe on both the audio signal output ports TP1 and TP2 at the same time. Next adjust variable resistor "QPS" so that the phase difference between TP1 and TP2 is 90°. Then by using oscilloscope, observe both the carrier signal output ports TP3 and TP4 in at the same time. Next adjust variable resistor "Phase Adjust" so that the phase difference between TP3 and TP4 is 90°.
4. By using oscilloscope, observe the output signal waveforms of DSB-SC_Q modulation output port (TP5). Next adjust variable resistor VR1 (gain adjustment) so that the output amplitude of the carrier signal is maximum without distortion and also adjust variable resistor VR3 (modulation index adjustment) so that the center level of upper peak and lower peak are 0 V or the modulation index is 100%.
5. By using oscilloscope again, observe on the output signal waveform of DSB-SC_I modulation output port (TP6). Next adjust variable resistor VR2 (gain adjustment) so that the output amplitude of the carrier signal is maximum without distortion, and also adjust variable resistor VR4 (modulation index adjustment) so that the center level of the upper peak and lower peak are 0V or the modulation index is 100%.
6. By using oscilloscope, observe on the output signal of SSB modulation output port (SSB O/P), then record the measured results.

RECEPTION OF SINGLE SIDEBAND AM WAVEFORMMS

1. Connect the modulated SSB signal (SSB O/P) in Fig:-2.3 (ACS 5-1) to the input terminal (DSB-SC/SSB I/P) of the product detector in Fig:-2.4 (ASC 6-1) at the same time , input the carrier signal in Fig-2.3 (ACS 5-1) to the carrier signal input port (Carrier I/P) in Fig:-2.4 (ASC 6-1)

2. By using oscilloscope, observe the output signal waveform of the product detector (Audio O/P) in Fig:-2.4 (ASC6-1). Next adjust variable resistor VR1 and VR2, so that the output amplitude is maximum without distortion. Finally, record the output signal waveforms of the product detector TP1, TP2 and the demodulator signal (Audio O/P).While J1 be short circuit and J2 be open circuit
3. Let J1 be open circuit and J2 be short circuit. Then repeat step 2 and record the result.

(e) Report

1. Block diagram of transmitter and receiver block.
2. Comment on the function of various blocks.
3. Draw the wave shapes at various test points and explain.
4. Anything else instructed by teacher.
5. Discussion expressing originality of the student.

Experiment No: **03**
Name of the Experiment: **Study of Frequency Modulation**

(a) Objective:

1. To understand the Theory of Frequency Modulation(FM).
2. To understand the FM Transmission/Reception processes.
3. To implement frequency modulation and demodulation technique.

(b) Equipments:

1. ANACOM 2 FM trainer module
2. Oscilloscope
3. Power Supply
4. Multimeter

(c) Theory:

Frequency modulation is a type of modulation where the frequency of the carrier is varied in accordance with the modulating signal. The amplitude of the carrier remains constant. The information-bearing signal (the modulating signal) changes the instantaneous frequency of the carrier. Since the amplitude is kept constant, frequency modulation is a low-noise process and is commonly used at VHF radio frequencies for high-fidelity broadcasts of music and speech. Normal (analog) TV also use FM to broadcast the sound. There are several devices that are capable of generating FM signals, such as a reactance modulator.

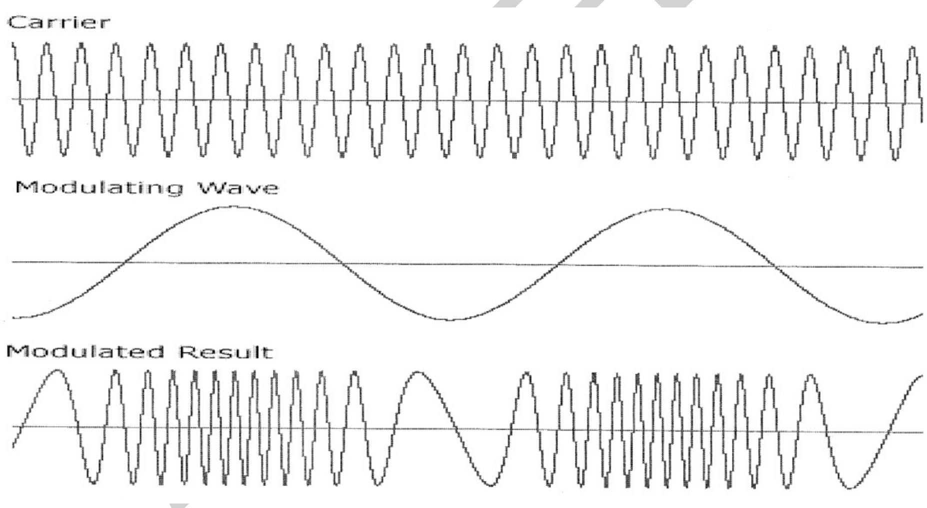


Fig 3.1:- Frequency modulation.

ANACOM 2 has two modulator circuits- the reactance modulator and the varactor modulator circuit and five demodulator circuits - Detuned Resonant circuit, Quadrature detector, Foster-Seeley detector, Ratio detector and Phase-Locked Loop detector.

1. Operation of REACTANCE MODULATOR:

The oscillator generates a radio-frequency sinusoidal output, the frequency of which is determined by the inductance and capacitance of the tuning circuit. The circuit to the right of the tank circuit operates a voltage-variable capacitance.

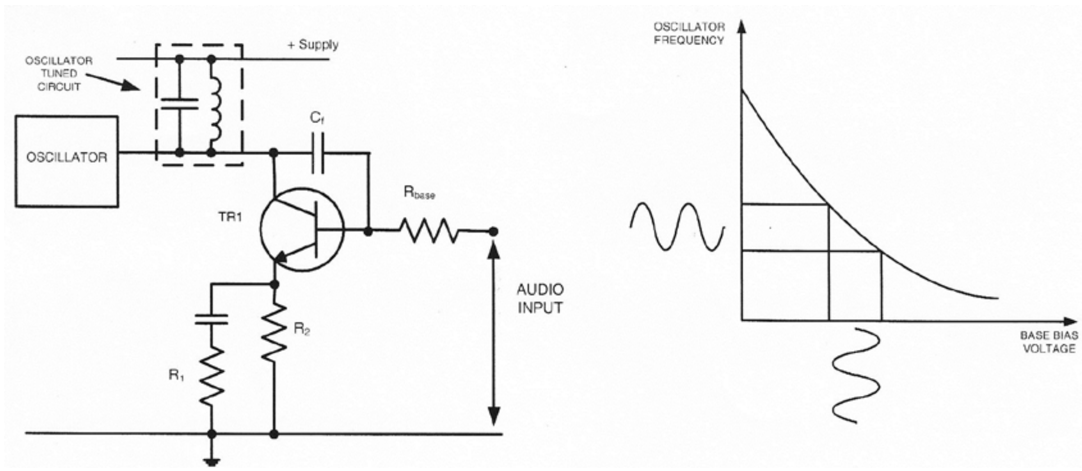


Fig:-3.2

The output of the oscillator appears at the collector of TR1 and is feedback by the capacitor C_f to the base of TR1. As C_f is small, X_{c_f} is large compared to R so that the voltage at the transistor's base is a small sine wave that leads the voltage at the collector by 90°. This base signal results in a collector current, which also leads the collector voltage by 90°. Thus the transistor circuit appears to the oscillator as a capacitance, which adds to the capacitance of the tank circuit.
The size of this capacitance depends on the transconductance (g_m) of the TR1, which depends on R1, R2 and also on the d.c. bias voltage at the base of TR1. The larger is the base bias voltage, the larger is the value of g_m and thus the larger is the value of the capacitance which adds to the tank circuit.
When the sinusoidal audio signal is superimposed onto the d.c. bias voltage, the result will be a sinusoidal change in the frequency of the oscillator. This is how frequency modulation is performed with the reactance modulator.

2. PHASE-LOCKED LOOP DETECTOR:

The PLL detector uses PLL technology to demodulate FM signals. The diagram below shows a simple PLL detector. The phase detector compares the phase of the FM input and the VCO output. Frequency deviation of the carrier results in a phase difference between the two and the phase detector sends an error voltage to the low pass filter. The filtered error signal is used to change the VCO output frequency in order to reduce the phase error. The output of the low pass filter has amplitude that is proportional to the deviation of the FM input, so it is actually a replica of the original modulating signal. FM is converted directly to audio.

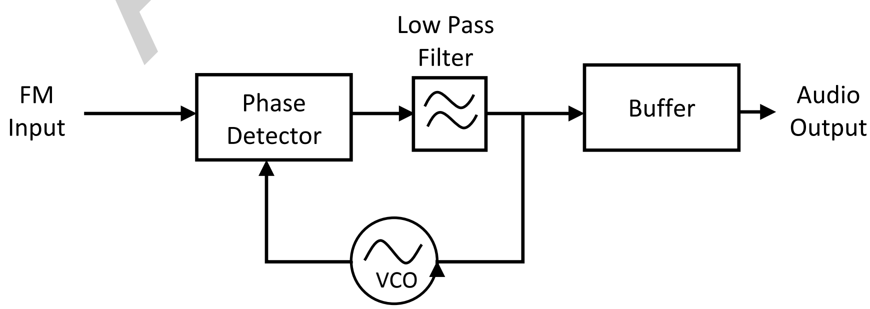


Fig:-3.3

A PLL can operate in three different modes:

1. Free running
2. Capture
3. Tracking

In the free running mode, the input frequency is not close enough to the VCO frequency and the PLL runs at the free running frequency determined by the timing circuits of the VCO. The error voltage is outside the range of the VCO. As the input frequency gets closer to the VCO frequency, the error voltage reaches a value at which it can begin to change the VCO frequency. This is the capture mode. The error voltage will continue to decrease as the VCO frequency gets closer to the input frequency. Finally, when the VCO is operating at the same frequency as the input, the PLL is in the tracking mode. The VCO will track changes in the input frequency as long as the input frequency remains in a range of frequencies known as the hold-in range.

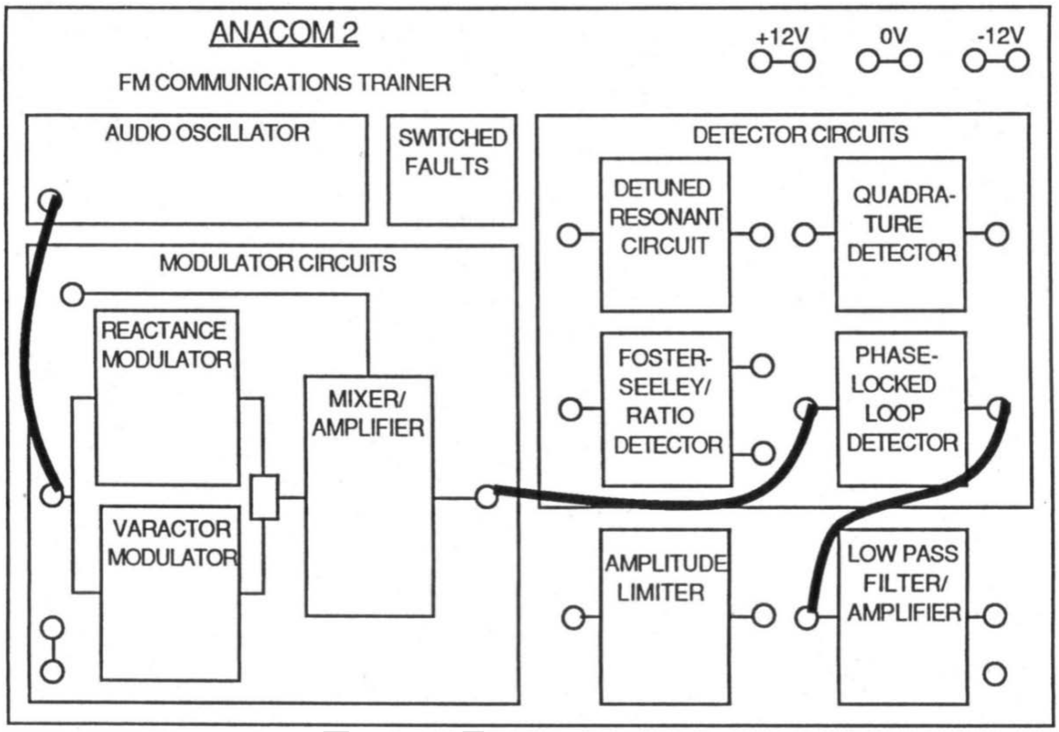


Fig-3.4

(d)Experimental Procedure:

A. Modulation by REACTANCE MODULATOR

1. Connect the ANACOM 2 module to the power supply as shown in Fig:- 3.5

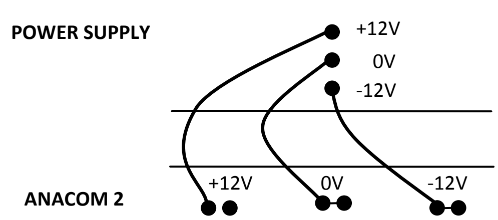


Fig: - 3.5

2. Ensure that the following conditions exist on the ANACOM 2 module:
   a. All switched faults OFF.
   b. AMPLITUDE preset (in the MIXER/AMPLIFIER block) in the fully **clockwise** position.
   c. VCO switch (in the PLL DETECTOR block) in OFF position.

3. Turn on power to the ANACOM 2 module.
4. Turn the AUDIO OSCILLATOR block's AMPLITUDE preset to its fully clockwise (MAX) position, and observe the block's output at t.p.1 on an oscilloscope. This is the audio frequency sine wave (frequency: 300Hz~3.4 KHz) that will be used as the modulating wave.
5. Turn the AMPLITUDE preset to its MIN position.
6. Connect the output of the AUDIO OSCILLATOR block to AUDIO INPUT socket of the MODULATOR CIRCUITS block, as shown in Fig:- 3.4.
7. Put the REACTANCE/VARACTOR switch in the REACTANCE position.
8. Vary the REACTANCE MODULATOR block's CARRIER FREQUENCY preset and observe t.p.34. The observed signal is the unmodulated FM carrier, as the audio input signal has zero amplitude at present. The carrier frequency preset can be varied from 453 KHz to 460 KHz. The amplitude of the FM carrier (at t.p.34) is adjustable by means of the MIXER/AMPLIFIER block's AMPLITUDE preset.
9. The CARRIER FREQUENCY preset actually controls the bias voltage applied to the base of the transistor in the reactance modulator circuit. To make a plot of **oscillator output frequency vs. base bias voltage,** follow the steps below:
    a. Turn the CARRIER FREQUENCY preset to its fully anticlockwise position-corresponding to minimum base bias voltage.
    b. Observe t.p. 34 (oscillator output frequency) and t.p. 11 (base bias voltage).
    c. Slowly turn the CARRIER FREQUENCY preset to its fully clockwise position and record the frequency at t.p. 34 as the d.c. voltage at t.p. 11 increases in 0.1 volt intervals.
10. Put the REACTANCE MODULATOR block's CARRIER FREQUENCY preset in its fully clockwise position. Turn the AUDIO OSCILLATOR'S AMPLITUDE preset to its MAX position. Observe the MIXER/AMPLIFIER block's FM output at t.p.34
11. Vary the amplitude of the modulating signal by turning the AMPLITUDE PRESET slowly counter clockwise and observe t.p.34 to see how the **amplitude** of the modulating signal affects the frequency deviation in the FM signal.
12. Return the AUDIO OSCILLATOR'S AMPLITUDE preset to its fully clockwise position, then vary the **frequency** of the modulating sine wave by adjusting the AUDIO OSCILLATOR'S, FREQUENCY preset throughout its range. Again observe the effect of varying the modulating frequency on the monitored FM waveform.
13. Return the CARRIER FREQUENCY preset to its midway position, and monitor the AUDIO INPUT (at t.p.6) and the FM OUTPUT (at t.p.34).
14. Turn the AUDIO OSCILLATOR'S AMPLITUDE preset throughout its range of adjustment, and note that the amplitude of the FM OUTPUT signal does not change. This is because the audio information is contained entirely in the signal's frequency, and not in its amplitude.

N.B. The same procedure can be followed to observe frequency modulation by ANACOM 2's VARACTOR MODULATOR CIRCUIT when REACTANCE/VARACTOR switch is in the VARACTOR position.

B. Demodulation by PHASE-LOCKED LOOP DETECTOR

N.B. Demodulation can be done by any one of the four blocks by following the same procedure given below.

1. Connect the ANACOM 2 module to the power supply.
2. Ensure that the following initial conditions exist on the ANACOM 2 module:
    a) All switched faults OFF.
    b) AUDIO AMPLIFIER block's AMLITUDE preset in fully **clockwise** position.
    c) AUDIO AMPLIFIER block's FREQUENCY preset in fully **counter-clockwise** position.
    d) AMPLITUDE preset (in the MIXER/AMPLIFIER block) in the fully **clockwise** position.
    e) VCO switch (in the PLL DETECTOR block) in ON position.
3. Turn on power to the ANACOM 2 module.
4. Make the necessary connections according to the Fig:-3.4.
5. Ensure that the CARRIER FREQUENCY preset of the modulator circuit is in the midway position so as to have a carrier frequency of 455Hz.
6. The AUDIO OSCILLATOR'S output signal (at t.p. 1) is now used to frequency modulate the 455 KHz carrier sine wave. Observe the FM output at t.p. 34.

7. Now monitor the audio input to the modulator block (t.p.14), together with the output of the PLL detector block (at t.p.60). The signal at t.p.60 should contain three components:
   - A positive d.c. offset voltage;
   - A sine wave at the same frequency as the audio signal;
   - A high-frequency ripple component.
8. You will note that the amplitude of the high-frequency ripple component at t.p. 60 is actually higher than the required sine wave component. This indicates that the simple, passive low pass filter circuit within the detector is not sufficient to remove this unwanted high-frequency component. Use the LOW PASS FILTER/AMPLIFIER block to overcome this problem.
9. Observe the signals at t.p. 14 and at t.p.73 and note how the output of the demodulated signal has been improved by the low-pass filtering.
10. Adjust the *GAIN* preset in the AMPLIFIER block until the signals at t.p. 14 and t.p.73 are the same.
11. To investigate the **effect of noise** on the system:
    - Adjust a signal generator to have a sinusoidal output of l00 mV p-p at 2 KHz. This will be the noise input.
    - Connect the output of the signal generator to the NOISE INPUT socket of the modulator block.
    - Observe the noise input at t.p.5 and the FM output at t.p.34. Note that the FM signal is now being amplitude-modulated by the noise signal as well as being frequency-modulated by the audio frequency signal.
    - The amplitude modulations simulate the effect that the transmission path noise would have on the FM waveform reaching the receiver.
    - Observe the signal at t.p. 14 and t.p.73. You will notice an additional component at t.p. 73 - a considerable amount of ripple at the frequency of the noise input. This is because the PHASE-LOCKED LOOP is not completely immune to the amplitude variations.
    - Turn the AUDIO OSC. Block’s AMPLITUDE preset to its MIN position, so that no frequency modulation takes place.
    - Now connect an AMPLITUDE LIMITER between the FM output and the input to the PLL Detector. Observe the AMPLITUDE LIMITER's output at t.p.68 as well as the noise at t.p.5. Note that the amplitude modulations due to the noise input have been removed.
    - Observe the output of the LOW PASS FILTER block at t.p.73, and note that the noise component is now minimal.
    - Return the AUDIO OSC. Block’s AMPLITUDE preset to its MAX position, and monitor t.p.73. Note that amplitude variations in the FM waveform now have no effect on the final demodulated output.

(e)Report:

Report on this experiment should include the following:

1. Block Diagram.
2. All observations including waveforms at various test points.
3. A plot of oscillator frequency vs. base bias voltage for the modulator block.
4. Question answers:
   a) Compare between AM and FM.
   b) Discuss where FM is used in the modern communication system.
   c) What is the effect of varying the frequency of the modulating signal on the FM wave?
   d) What is the effect of varying the amplitude of the modulating signal on the FM wave?
5. Anything else instructed by the teacher.

Experiment no: **4**
Name of the Experiment: **Study of Signal Sampling and Reconstruction**

a) Objective:

1. To observe the sampling and reconstruction of signal.
2. To observe the effect of second/fourth order filter on the signal recovery process.

b) Equipments:

1) Trainer board
2) Oscilloscope
3) Power supply.

c) Theory:

Sampling process:

The sampling process is usually described in the time domain. As such, it is an operation is basic to digital signal processing and digital communications. Through use of the sampling process, an analog signal is converted into a corresponding sequence of samples that are usually spaced uniformly in time. Clearly, for such a procedure to have practical utility, it is necessary that we choose the sampling rate properly, so that the sequence of samples uniquely defines the original analog signal.

Consider an arbitrary signal g (t) of finite energy, which is specified for all time. A segment of the signal g (t) is shown in Fig 4.1(a). Suppose that we sampled the signal g (t) instantaneously and at a uniform rate, once every T_s seconds. Consequently, we obtain an infinite sequence of samples spaced T_s seconds apart. We refer to T_s as the **sampling period,** and to its reciprocal $f_s = 1/T_s$ as the **sampling rate.** This idle form of sampling called **instantaneous sampling.**

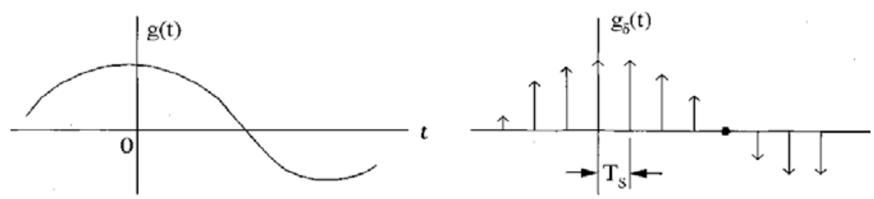


Fig 4.1 the sampling process (a) Analog signal, (b) Instantaneously sampled version of the analog signal.

The signal g (t) is strictly band limited; with no frequency components higher than W Hz. that is the Fourier, transform G (f) of the signal g (t) has the property that G (f) is zero for $|f| \geq W$, as illustrated in Fig 4.2(a). Suppose also that we choose the sampling period $T_s = 1/2W$. Then the corresponding spectrum G_δ (f) of sampled signal g_δ(t) is shown in Fig 4.2(b).

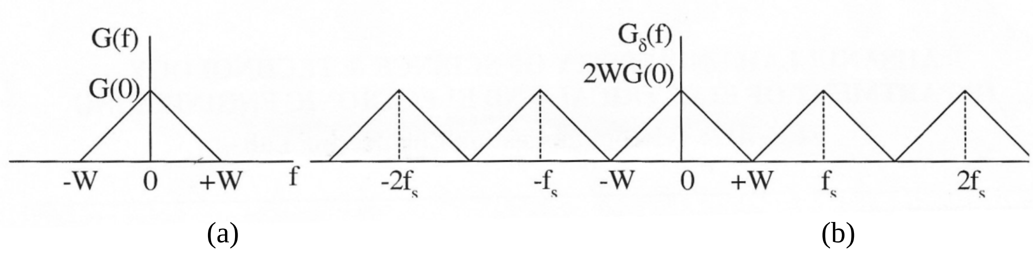


Fig 4.2 (a) spectrum of strictly band limited signal g (t). (b) Spectrum of sampled version of g (t) for a sampling period T_s =1/2W.

The sampling rate of 2W samples per second, for a signal bandwidth of W Hz, is called the Nyquist rate; its reciprocal 1/2W is called the Nyquist interval.

The sampling theorem is based on the assumption that the signal g (t) is strictly band limited. In practice, however, am information bearing signal is not strictly band limited, with the result, some degree of under sampling is encountered. Consequently, the sampling process produces some aliasing.

Aliasing refers to the phenomena of a high frequency component in the spectrum of the signal seemingly taking on the identity of a lower frequency in the spectrum of its sampled version.

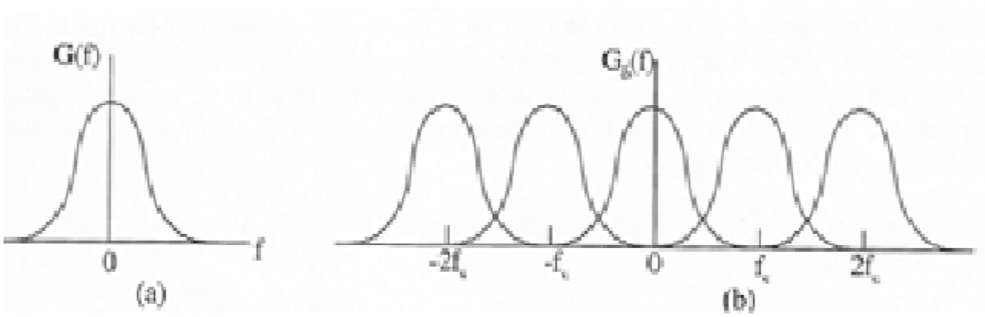


Fig 4.3. (a) spectrum of strictly band limited signal g(t). (b) spectrum of sampled version of g(t) for a sampling period $T_s = 1/2W$

To combat the effect of aliasing in practice, we may use two corrective measures, as described here:

1. Prior to sampling, a low pass ant-aliasing filter is used to attenuate those high frequency components of the signal that are not essential to the information being conveyed by the signal.
2. The filtered signal is sampled at a rate higher than the Nyquist rate.

The use of sampling rate higher than the Nyquist rate also has the beneficial effect of easing the design of the **reconstruction filter** used to recover the original signal from its sampled version.

A message signal that has been anti-alias filtered, resulting in the spectrum shown in Fig.4.4a. The corresponding spectrum of the instantaneously sampled version of the signal is shown in Fig.4.4(b), assuming a sampling rate is higher than the Nyquist rate.

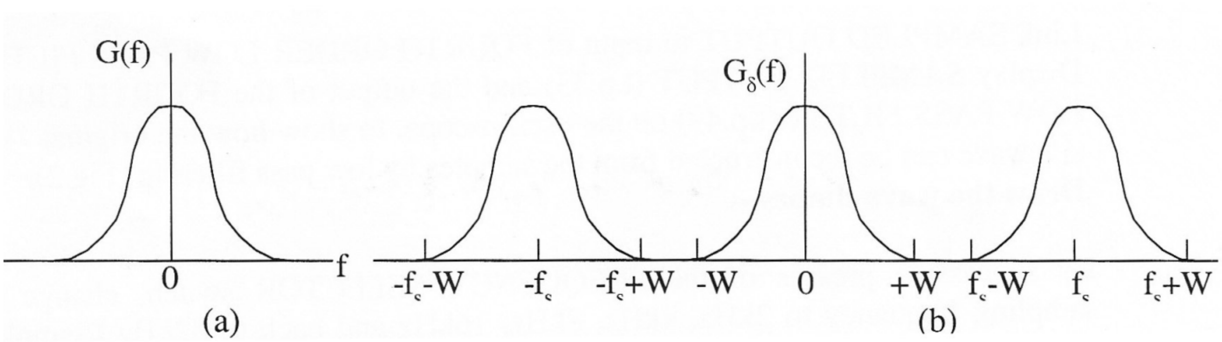


Fig 4.4. (a) anti-aliasing filtered spectrum of a signal, (b) Spectrum of an under sampled version of the signal, assuming the use of a sampling rate greater than the Nyquist rate

d) Procedure:

1. Connect supplies to board. The d.c. power requirements are; +5V, 0V, +12V, -12V.

2. Ensure SAMPLING CONTROL switch is in 'INTERNAL' position.
3. Put DUTY CYCLE SELECTOR switch in position '5'
4. Link 1kHz sinewave output to ANALOG INPUT.
5. Turn on power to board.
6. Display 1KHz sinewave (t.p.7) and SAMPLED OUTPUT (t.p.33) on an oscilloscope. This shows the 1kHz sinewave being sampled at 32kHz; so that there are 32 samples for every cycle of the sinewave (see Fig. 4.5 at the end of this chapter).
7. Link SAMPLED OUTPUT to input of FOURTH ORDER LOW PASS FILTER. Display SAMPLED OUTPUT (t.p.33) and the output of the FOURTH ORDER LOW PASS FILTER (t.p.49) on the oscilloscope, to show how the original 1kHz sinewave can be reconstructed from the samples by low pass filtering (Fig.4.6).
8. By successive presses of the FREQUENCY SELECTOR switch, change the sampling frequency to 2kHz, 4kHz, 8kHz, 16kHz and back to 32kHz (Sampling frequency is always 1/ 10 of the frequency indicated by the illuminated LED). Observe how the SAMPLED OUTPUT changes in each case, and how the lower sampling rates introduce distortion into the filter's output waveform. The sampled waveform contains components at f_x, f_s-f_x, and f_s+f_x, $2f_s$-f_x etc., where f_x = input sinewave frequency, f_s= sampling frequency. Considering the characteristic of the filter circuit, **can you explain why the distortion occurs?**
9. The present position of the DUTY CYCLE SELECTOR switch ('5') indicates that the duration of each sample is 50% of the sampling period (the time between the start of adjacent samples). Variation of the switch setting allows this proportion (called the 'sampling duty cycle') to be changed from 0% to 90% in 10% steps. Using a 32kHz sampling frequency, vary the position of the DUTY CYCLE SELECTOR switch, observing how the SAMPLED OUTPUT changes and how the amplitude of the filter's output waveform changes. This amplitude increases linearly as the sampling duty cycle increases from 10% to 90%.
Vary the position of the DUTY CYCLE CONTROL switch, noting how SAMPLED OUTPUT changes and how the amplitude of the filter's output waveform increases as the sampling duty cycle increases.

DUTY CYCLE	Vp
10%	
30%	
50%	
70%	
90%	

10. Add a link from the SAMPLE OUTPUT to the input of the SECOND ORDER LOW PASS FILTER. Display the outputs of the SECOND ORDER and FOURTH ORDER low pass filters (t.p.44 and t.p.49) on the oscilloscope (Fig.4.7). With a sampling duty cycle of 50% (DUTY CYCLE SELECTOR switch position '5') step through the 8kHz, 16kHz and 32kHz sampling frequencies, comparing the filter outputs after each step. Note that, for each sampling frequency, the fourth order output exhibits less distortion than the second order output. This is because the fourth order filter has a sharper 'roll-off than the second order filter, and thus rejects more frequency components caused by the sampling signal.

11. Remove the links between the SAMPLED OUTPUT and the inputs to the two filters. With a sampling frequency of 32 kHz and a sampling duty cycle of 50%, compare the SAMPLED OUTPUT (t.p.33) and the SAMPLE/HOLD OUTPUT (t.p.35) on the oscilloscope (Fig.4.8).
Vary the sampling duty cycle to illustrate how each sample is held at the sample/hold output. **Note how increasing the sampling duty cycle increases the proportion of time for which sampling occurs, and reduces the time for which these samples are held at the SAMPLE / HOLD OUTPUT.**

12. Link from SAMPLE/HOLD OUTPUT to input of FOURTH ORDER LOW PASS FILTER. Using a sampling frequency of 32kHz and a duty cycle of 50%, display the SAMPLE/HOLD OUTPUT (t.p.35) and the output of the FOURTH ORDER LOW PASS FILTER (t.p.49) on the oscilloscope, to show once again that the original 1 kHz sinewave can be reconstructed by low pass filtering (Fig. 4.9).

Using a 32kHz sampling frequency, vary the sampling duty cycle, and note that, in contrast with the results of Step 9 above, the filter's output amplitude is now independent of the sampling duty cycle, and is equal to the amplitude of the original analog input. This is an important result - with the SAMPLE/HOLD OUTPUT, the proportion of sampling time to holding time has no effect on.

13. By making successive presses of the FREQUENCY SELECTOR switch, note how the filter's output waveform changes with sampling frequency.

e) Report:

Should include

1. Comment on the function of various blocks.
2. Draw the wave shapes at various test points and explain.
3. Comment on the effect of varying sampling frequency.
4. Comment on the effect of varying duty cycle.
5. Discuss about the outputs of the Second Order and Fourth Order low pass filter blocks.
6. Discuss about Sample and Hold block.
7. Anything else instructed by teacher.
8. Discussion expressing originality of the student.

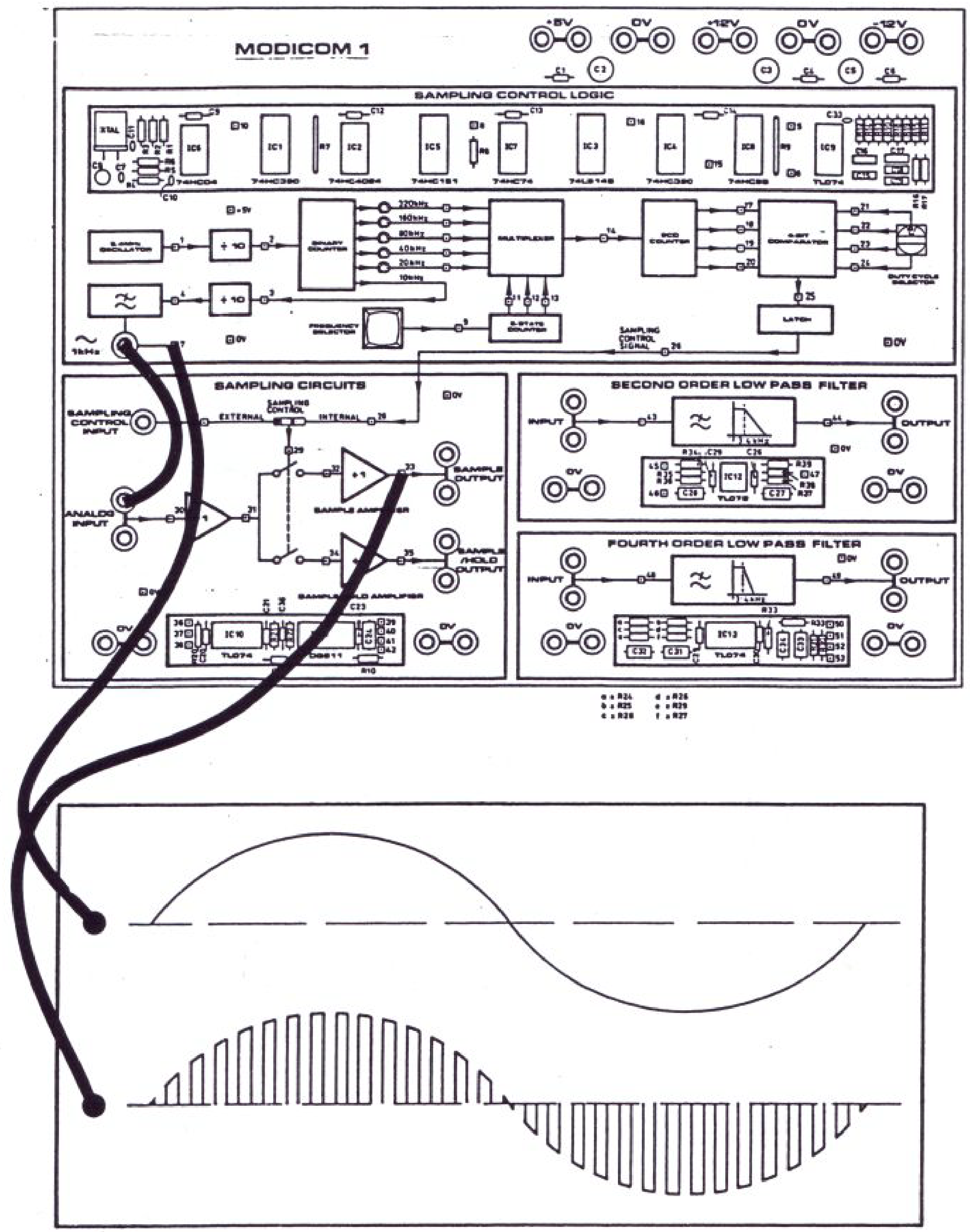


Figure 4.5 : SAMPLED OUTPUT (t.p.33) on an Oscilloscope.

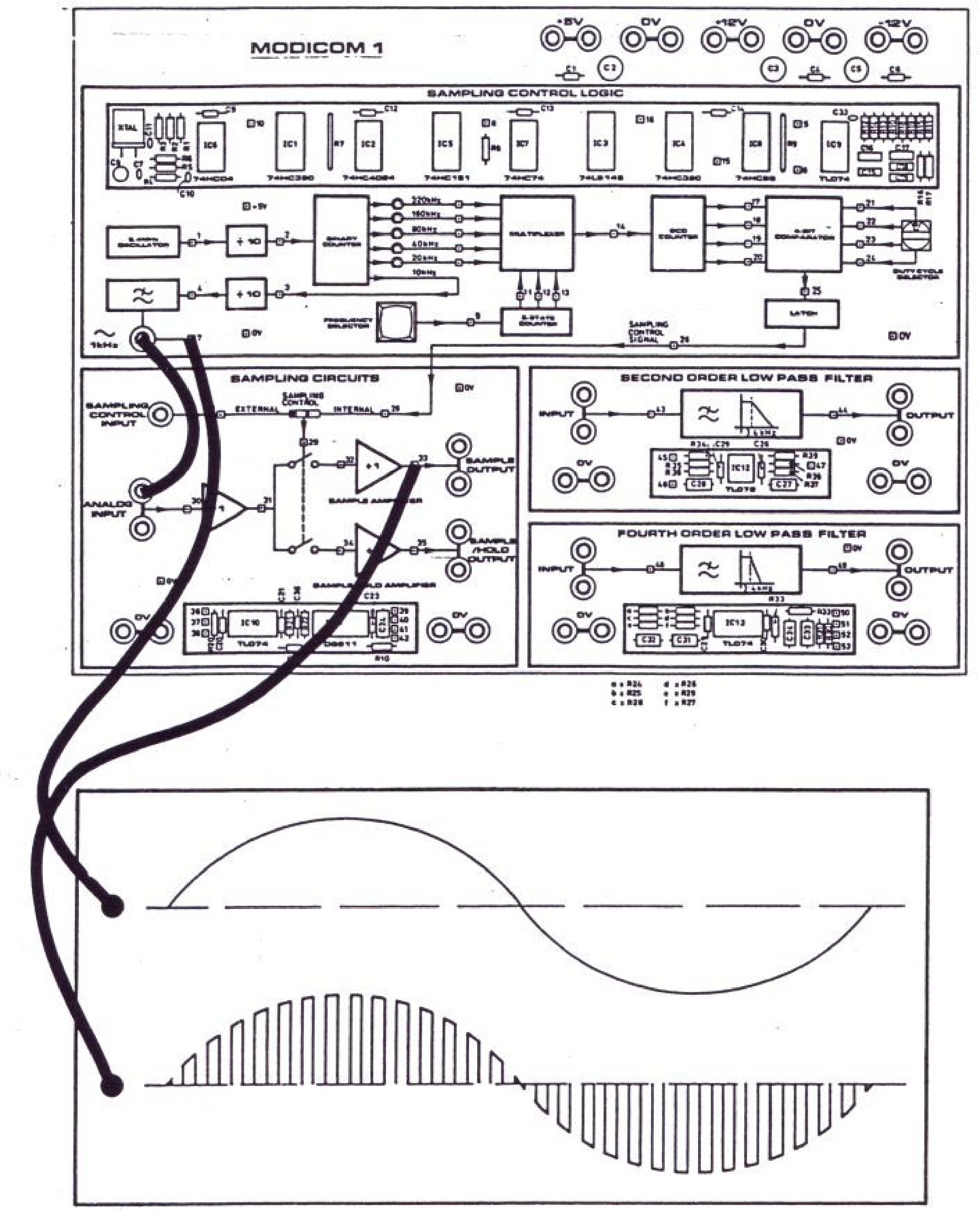


Figure 4.6: Output of the FOURTH ORDER LOW PASS FILTER (t.p.49) on the Oscilloscope

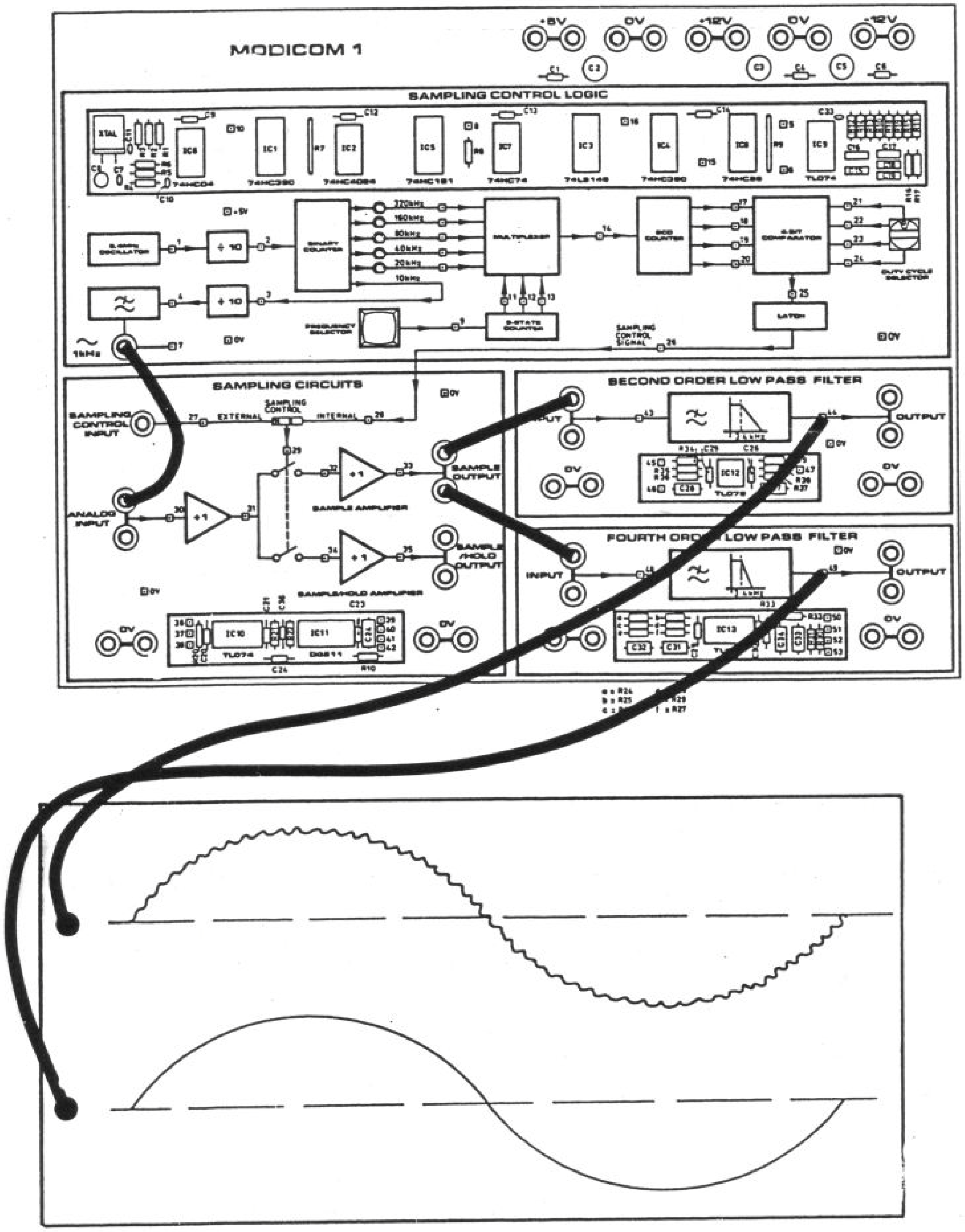


Figure 4.7: Outputs of the SECOND ORDER and FOURTH ORDER low pass filters (t.p.44 and t.p.49) on the Oscilloscope.

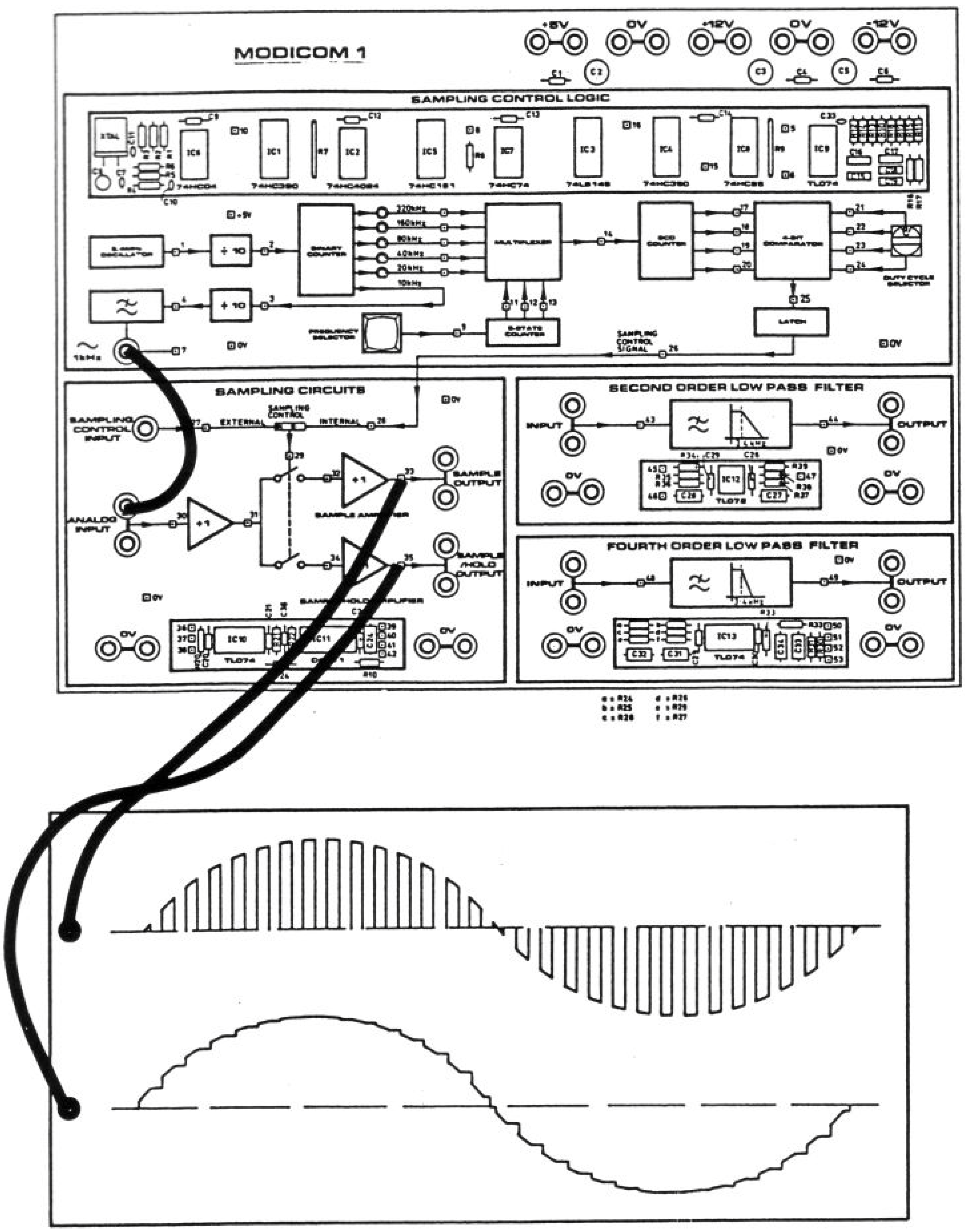


Figure 4.8 : SAMPLE/HOLD OUTPUT (t.p.35) on the Oscilloscope.

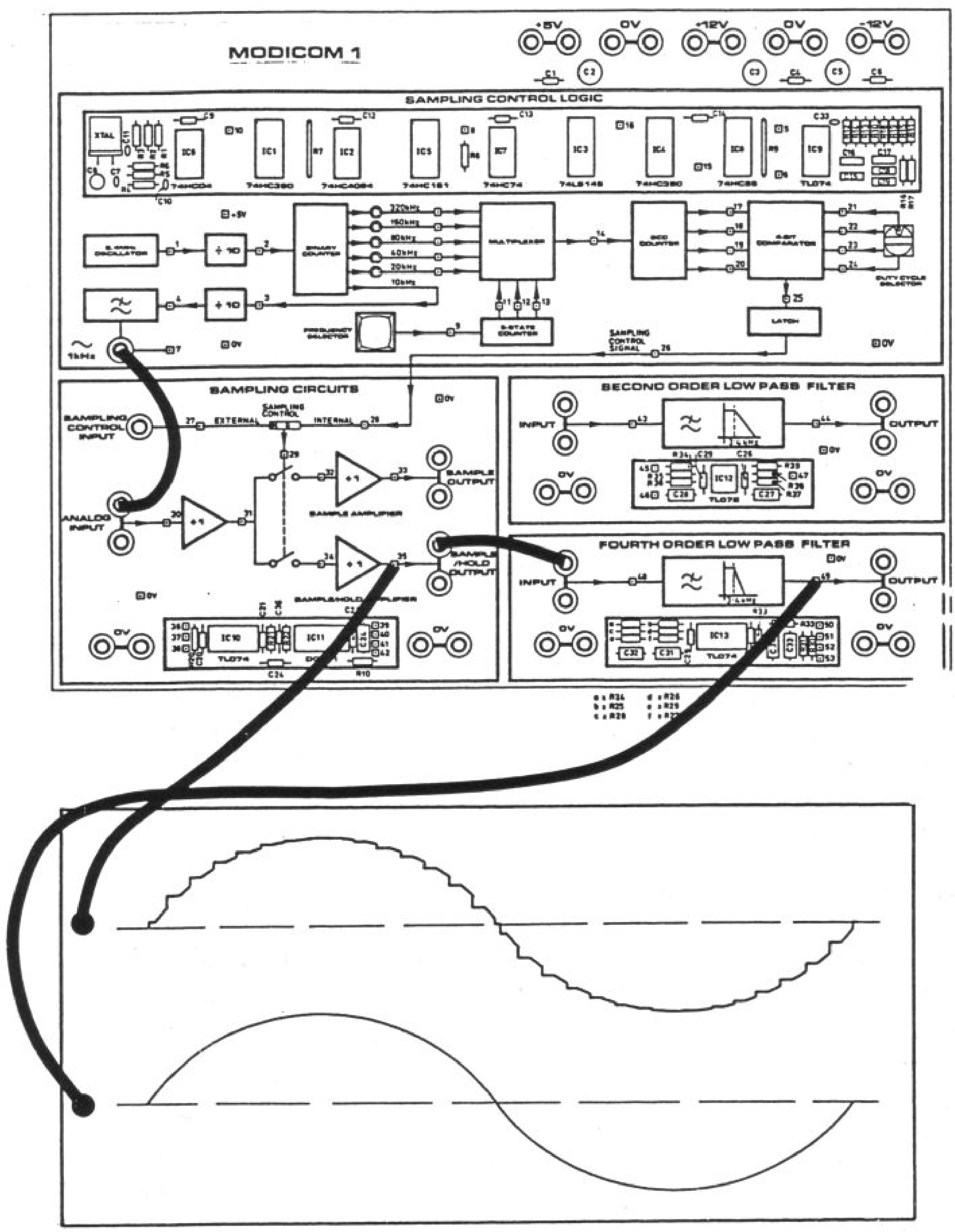


Figure 4.9: Output of the FOURTH ORDER LOW PASS FILTER (t.p.49) on the Oscilloscope.

Experiment no: **5**
Name of the Experiment: **Study of Time Division Multiplexing System.**

a) Objective:

1. To understand the concept of TDM.
2. To observe the waveshapes of multiplexed and demultiplexed signal in a 4-Ch TDM System.
3. To understand the importance of synchronization in TDM system.

b) Equipments:

1. Trainer board
2. Oscilloscope
3. Power supply

c) Theory:

Multiplexing (also known as **muxing**) is a method by which multiple analog message signals or digital data streams are combined into one signal over a shared medium. In **pulse-amplitude modulation** (PAM), the amplitudes of regularly spaced pulses are varied in proportion to the corresponding sample values of a continuous message signal; the pulses can be of a rectangular form or some other appropriate shape. Pulse-amplitude modulation as defined here is somewhat similar to natural sampling, where the message signal is multiplied by a periodic train of rectangular pulses.

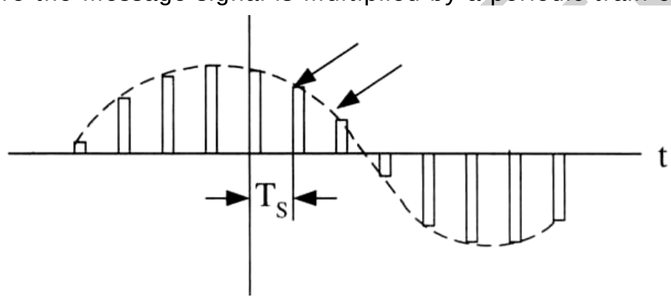


Fig 5.1. Flat-top samples, representing an analog signal.

An important feature of the sampling process is a conservation of time. That is, the transmission of the message samples engages the communication channel for only a fraction of the sampling interval on a periodic basis, and in this way some of the time interval between adjacent samples are cleared for use by other independent message sources on a time-shared basis. We thereby obtain a time-division multiplex (TDM) system, which enables the joint utilization of a common communication channel by a plurality of independent message sources without mutual interference among them.

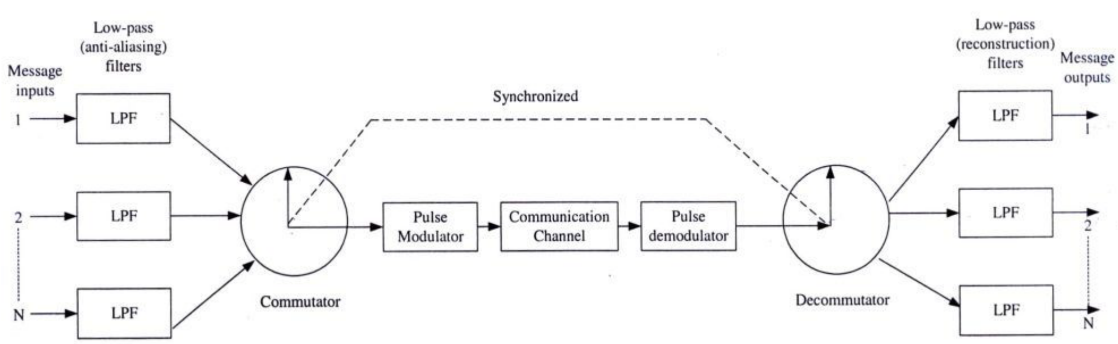


Fig 5.2. Block Diagram of TDM system

The concept of TDM is illustrated by the block diagram shown in Figure 5.1 Each input message signal is first restricted in bandwidth by a low-pass anti-aliasing filter to remove the frequencies that are nonessential to an adequate signal representation.
The low-pass filter outputs are then applied to a commutator, which is usually, implemented using electronic switching circuitry. The function of the commutator is twofold: (1) to take a narrow sample of each of the N input messages at a rate f_s**,** that is slightly higher than 2W, where W is the cutoff frequency of the anti-aliasing filter, and (2) to sequentially interleave these samples inside the sampling interval T_s.
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 modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over the common channel. It is clear that the use of time-division multiplexing introduces a bandwidth expansion factor N, because the scheme must squeeze N samples derived from N independent message sources into a time slot equal to one sampling interval.
At the receiving end of the system, the received signal is applied to a pulse demodulator, which performs the reverse operation of the pulse modulator. The narrow samples produced at the pulse demodulator output are distributed to the appropriate low-pass reconstruction filters by means of a decommutator, which operates in synchronism with the commutator in the transmitter.
This **synchronization** is essential for a satisfactory operation of the system. The way this synchronization is implemented depends naturally on the method of pulse modulation used to transmit the multiplexed sequence of samples.

The Phase Locked Loop (PLL)

The PLL contains a phase detector, a dc amplifier, a low pass filter and a voltage controlled oscillator (VCO). When PLL has an input signal with a frequency of f_{in}. Its VCO will produce an output frequency that equals f_{in}. Fig 5.3. is a block diagram of PLL.

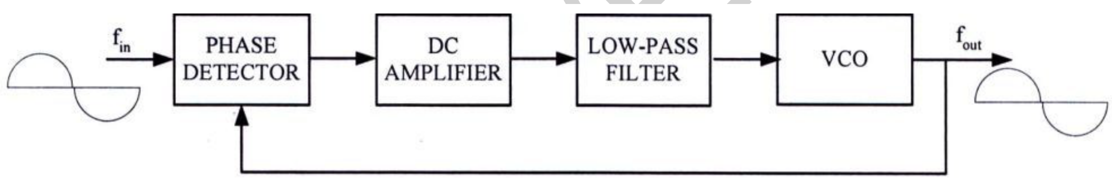


Fig 5.3:- Block Diagram of a PLL.

The phase detector produces an output voltage proportional to the phase difference between two input signals. The input voltage to the VCO determines the output frequency.
Let us assume that the input frequency is 11 kHz and VCO frequency is 10 kHz. This will appear to be a phase difference $\Delta\phi$. In this case the phase detector produces an output voltage. After being amplified and filtered, this voltage increases the VCO frequency. The VCO frequency will increase until it equals 11 kHz, the signal frequency of the input signal. When both frequencies are equal the VCO is locked on to the input signal.

d) Experimental Procedure:

Mode 1: Three connections between Transmitter and Receiver, four analog channels, minimal* Receiver complexity.

1. Connect supplies to the board. The D.C. supply requirements are +5V, 1 A, ± 12V @ 1 A/rail
2. Put DUTY CYCLE CONTROL switch in position '5'
3. Ensure the following presets are turned fully CLOCKWISE: SYNC LEVEL, 250Hz, 500Hz, 1kHz, ~2kHz and COMPARATOR THRESHOLD LEVEL

4. Turn the CLOCK TIMING CONTROL preset fully anticlockwise.

5. Make the following connections:
~250Hz to CH.0 input of TRANSMITTER block
~500Hz to CH.1 input of TRANSMITTER block
~ 1 kHz to CH.2 input of TRANSMITTER block
~ 2 kHz to CH.3 input of TRANSMITTER block

6. Turn on power to board.

7. Draw the wave shapes of CHO, CH1, CH2, and CH3 in a graph paper (minimum two cycles has to be shown)

8. Display TX OUTPUT (t.p.20) together with TRANSMITTER CH.0 input (t.p11 -for scope triggering purposes). The signal on TX OUTPUT is made up of samples of the four sine wave inputs. The operation of converting a signal into a series or samples is called PULSE AMPLITUDE MODULATION (P.A.M.). The TRANSMITTER circuit samples Input Channel 0 (CH.O). 'Then Channe1, Channel 2. and Channel 3; this sampling procedure is repeated continuously, and is known as Time-Division Multiplexing (T.D.M), since samples from different channels are transmitted at different moments in time. The time-division multiplexed samples appear at TX OUTPUT (t.p.20).

Draw the wave shape of TX OUTPUT (t,p.20) together with CHO, CH1 CH2, and CH3 in same graph paper (continuous sampling procedure has to be shown)

9. Vary the "250Hz, ~500Hz, ~1 kHz and ~2 kHz presets in turn, to make it clearer which samples are from which input channel. These presets simply change the amplitudes of the sine wave inputs.

10. Return these presets to their fully clockwise positions before continuing.

11. Make the following connections:

TX CH.0	to RX CH.0
TX CLOCK	to RXCLOCK
TX OUTPUT	to RX INPUT

This is **Mode 1** of operation. In this mode, TX **CH.0 is used to tell the Receiver which transmitted samples belong to Channel 0, and** TX **CLOCK is used to clock the Receiver in synchronism with the Transmitter.**

The board interconnections are shown in Fig. 5.4 at the end of this chapter.

12. Display TX OUTPUT (t.p.20) and the Receiver's CH.0 low pass filter input (t.p.42) on the oscilloscope, to show that the samples corresponding to Analog Channel 0 have been extracted from the transmitted time-division multiplexed stream of samples.

13. Display the input of the Receiver's CH.0 low pass filter (t.p.42) and the Receiver's CH-0 output (t.p.43) on the oscilloscope, to show that the 250Hz sine wave has been reconstructed by low pass filtering.

14. The present position of the DUTY CYCLE CONTROL switch ('5') indicates that the duration of each sample is 50% of the 'time slot' allocated to each channel. Variation of the DUTY CYCLE CONTROL switch setting allows this proportion to be changed from 0% to 90% in 10% steps. Display TX OUTPUT (t.p.20) and Receiver CH.0 output (t.p.43) on the oscilloscope.

15. Return the DUTY CYCLE CONTROL switch to the position '5 '. Remove the probe on Receiver CH.0 output (t.p.43), and put it on each of the other Receiver outputs in turn (t.p.45, 47, 49), to show that each of the original sine waves has been correctly reconstructed.

Mode 2: Two connections between Transmitter and Receiver, four analog channels, phase-locked loop used for generation of Receiver clock.

16. Mode 1 of operation requires three links to be made between Transmitter and Receiver. This can be reduced to two links by removing the TX CLOCK-RX CLOCK link, and deriving the Receiver's clock from the TX CH.O signal. This new mode of operation is MODE 2. To demonstrate this mode, remove the TX CH.O-RX CH.0 and TX CLOCK-RX CLOCK links, and add the following new links between the Transmitter Timing Logic, Phase Locked Timing Circuit and Receiver Timing Logic blocks:

TX CH.O to PLLI/P
SYNC to RX CH.0
CLK to RX CLOCK

In addition, ensure that the slider of the switch in the Phase Locked Timing Circuit block is in the Right-Hand position.

The board interconnections are shown in Figure 5.5.

17. The Phase Locked Loop is now locked onto the TX CH.O signal, and produces two outputs:

SYNC: This is of the same frequency as the TX CH.O signal and is used to tell the Receiver which transmitted samples belong to Channel 0.

CLK: This signal has four times the frequency of the TX CH.O signal, and is used to clock the samples into the Receiver

SYNC and CLK can be examined on test points 29 and 27 respectively.

18. Show that all four sine waves can still be reconstructed in Mode 2, by displaying Receiver CH.O, CH.1, CH.2 and CH.3 outputs in turn on the oscilloscope (t.p. 43, 45, 47, 49).

Mode 3: One connection between Transmitter and Receiver, three analog channels, one sync channel, phase-locked loop used for generation of Receiver clock and for channel synchronization.

19. The number of links between Transmitter and Receiver can be further reduced to a single link by dedicating analog Channel 0 to carry synchronization ('sync') pulses. This is Mode **3** of operation. To demonstrate this mode, REMOVE the following links:

~250H to the TRANSMITTER block's CH.O input.
TX CH.O to PLL I/P.

Then MAKE the following connection:

SYNC LEVEL to the TRANSMITTER block's CH.O input.

In addition, put the slider of the switch in the Phase Locked Timing Circuit block into the Left-Hand position.

The board interconnections are shown in Fig. 5.6.

20. Display TX OUTPUT (t.p.20) and Transmitter CH. 1 input (t.p. 13) on the oscilloscope, using the latter for scope triggering purposes. Vary the SYNC LEVEL preset, and note how the sync pulses change in amplitude. Return this preset to its fully clockwise position. At the Receiver, these pulses are detected by a voltage comparator whose threshold level is selected to distinguish between signal samples and the higher amplitude sync pulses. The output of the comparator is the stream of extracted sync pulses (t.p.22). These sync pulses are input to the Phase-Locked Loop, which generates SYNC and CLK signals as for Mode 2.

21. Examination of Receiver CH. 1, CH.2 and CH.3 outputs (t.p. 45, 47 and 49) shows that the three original sine waves have once again been reconstructed. Receiver CH.O output (t.p.43) is a D.C. level, related to the level of the transmitted sync pulses.

e) Report:

Should include

1. Block diagram of transmitter and receiver block
2. Comment on the function of various blocks.
3. Discuss about Synchronization.
4. Anything else instructed by teacher.
5. Discussion expressing originality of the student

CH0

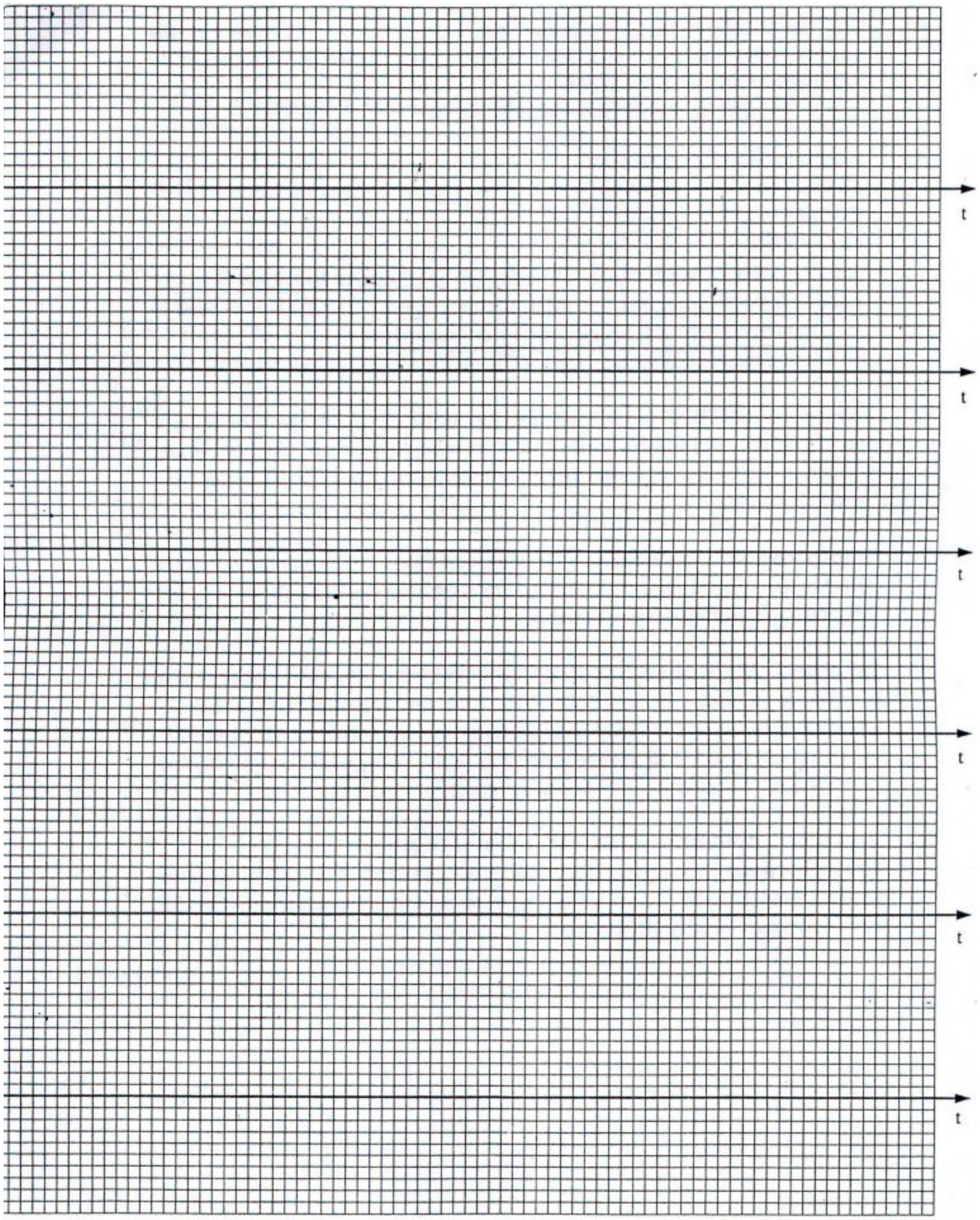
CH1

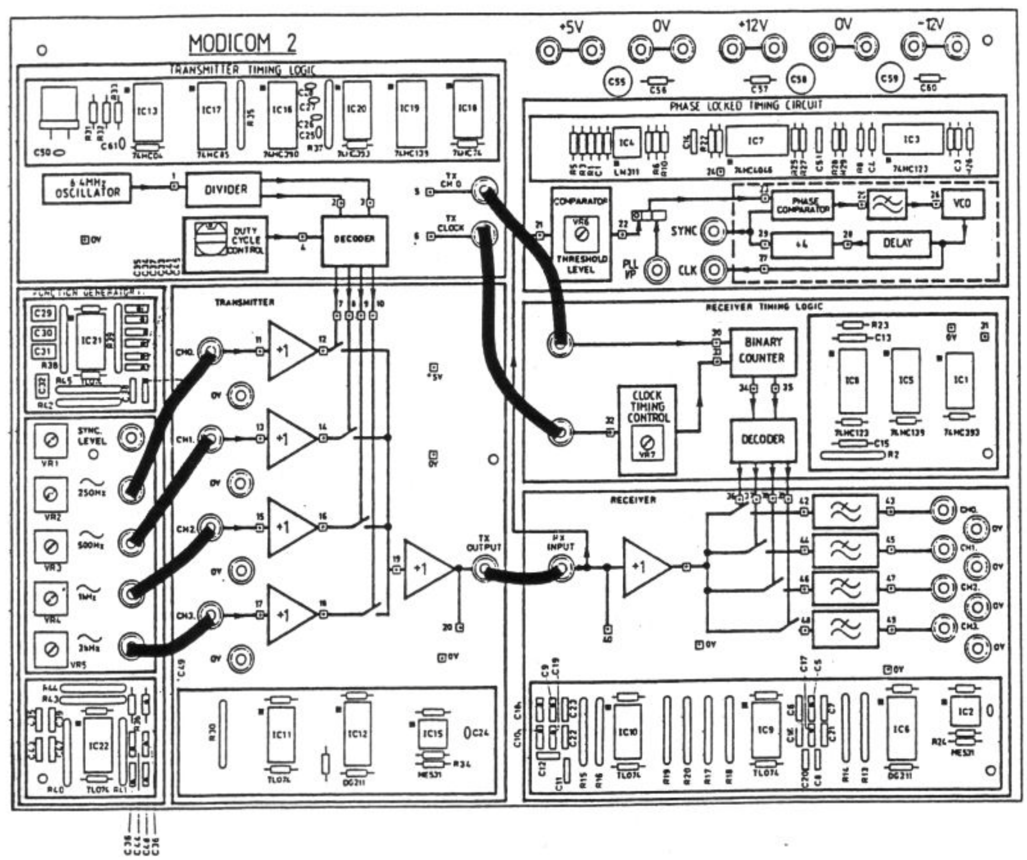
CH2

CH3

CLK

TX O/P





Mode 1 Connections

Figure 5.4: Mode 1 Connections.

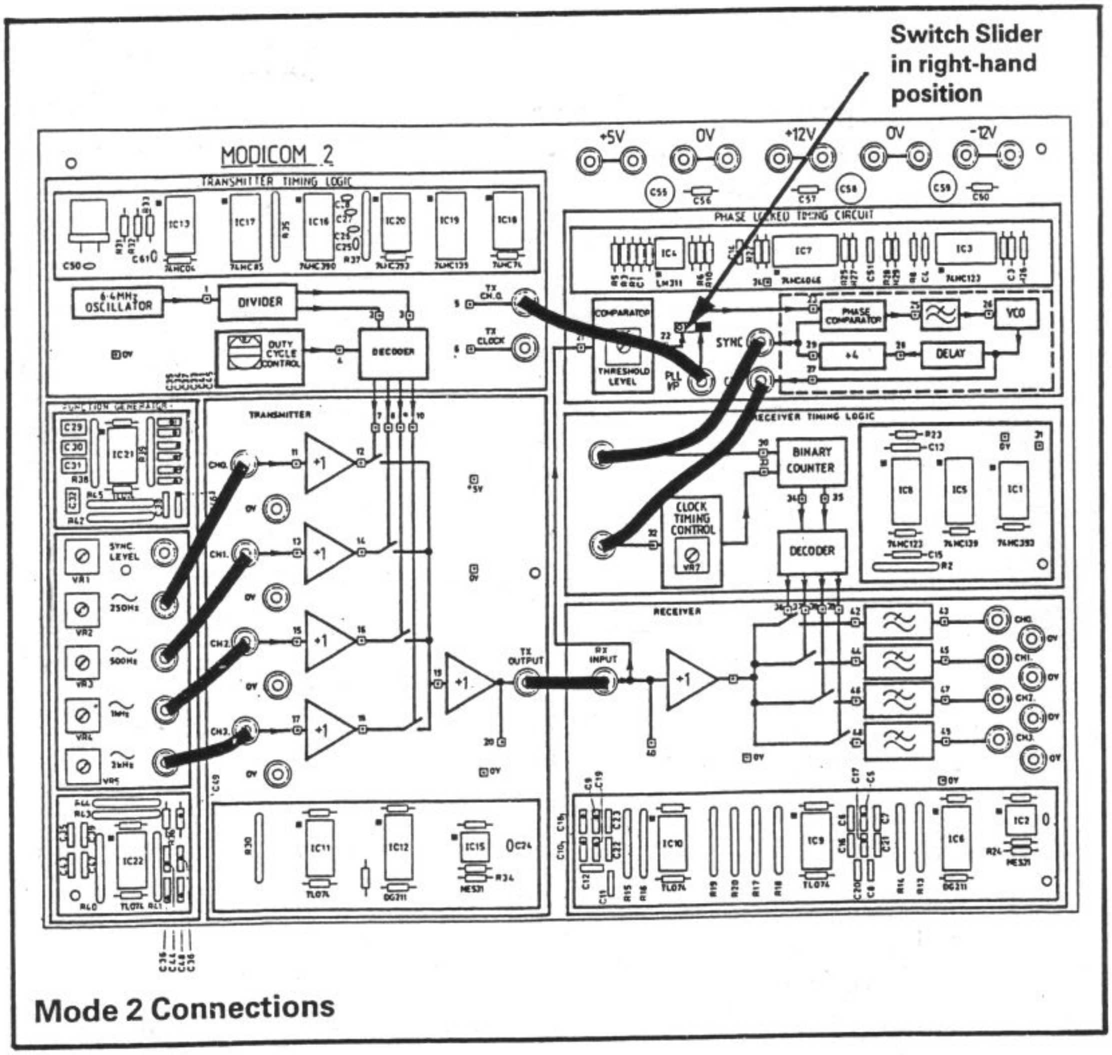


Figure 5.5: Mode 2 Connections.

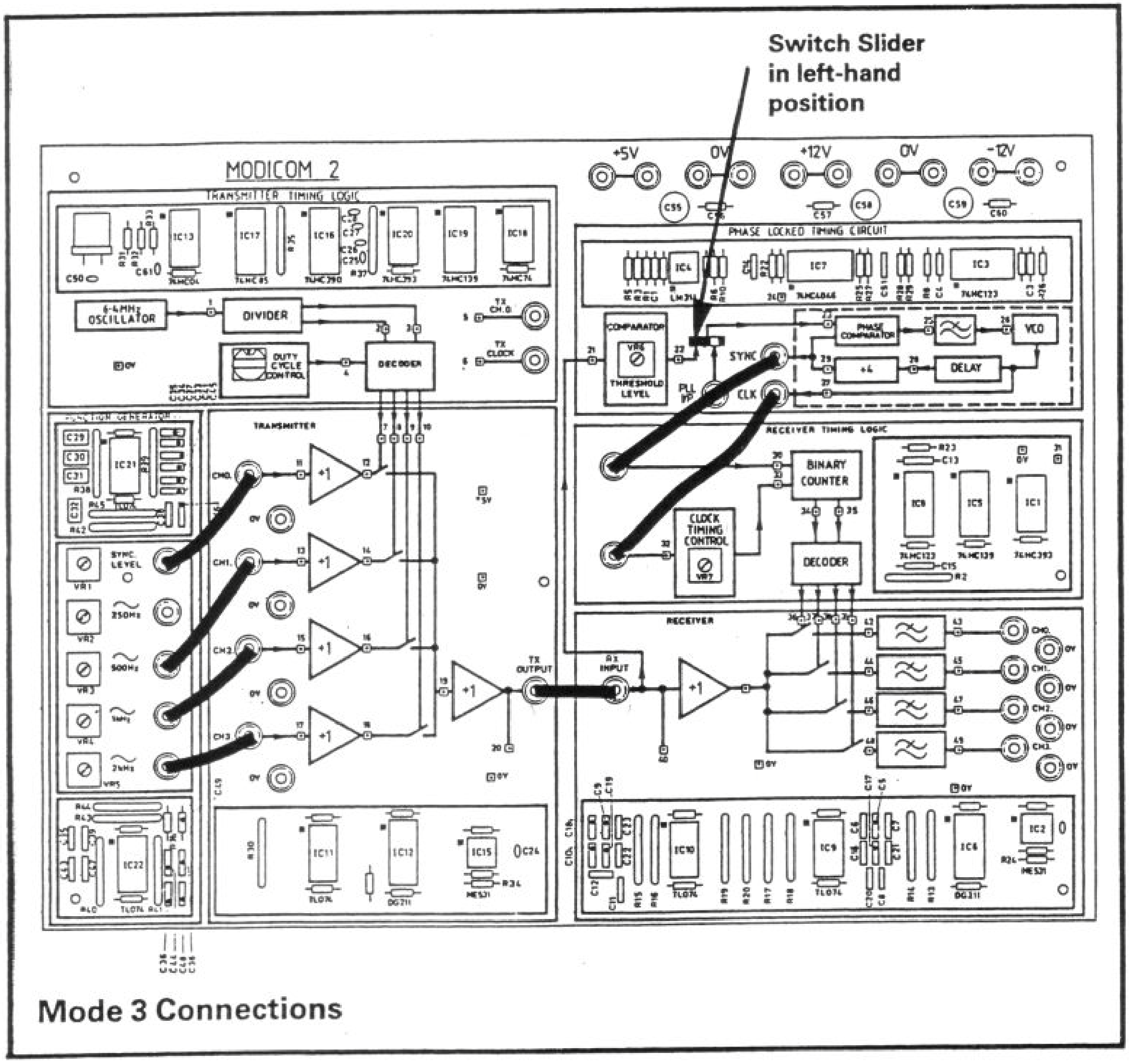


Figure 5.6: Mode 3 Connections.

Experiment no: **6**

Name of the Experiment: **Study of FDM (Multiplexer and Demultiplexer)**

a) Objectives:

1. To understand the operation theory of frequency-division multiplexing (FDM) and Demultiplexing.
2. To design and implement the FDM multiplexer and demultiplexer.

b) Equipments:

1. Trainer board
2. Oscilloscope
3. Power supply

c) Theory:

1. Frequency Division Multiplexing:

If the transmission channel only consists of one modulated signal, then the usage of channel is very low and the efficiency is also not good. Therefore, in order to comfort with the economic benefit, the channel must be able to transmit multiple signals, such as in the telephone system, the frequency range of the sound is 300 Hz to 3 kHz. In order to transmit this kind of signal via a single channel, we must divide the signal into several slots to prevent the interference, then we can obtain the original signal at the receiver. Generally, there are two types of signal division, which are frequency division multiplexing (FDM) and time division multiplexing (TDM).

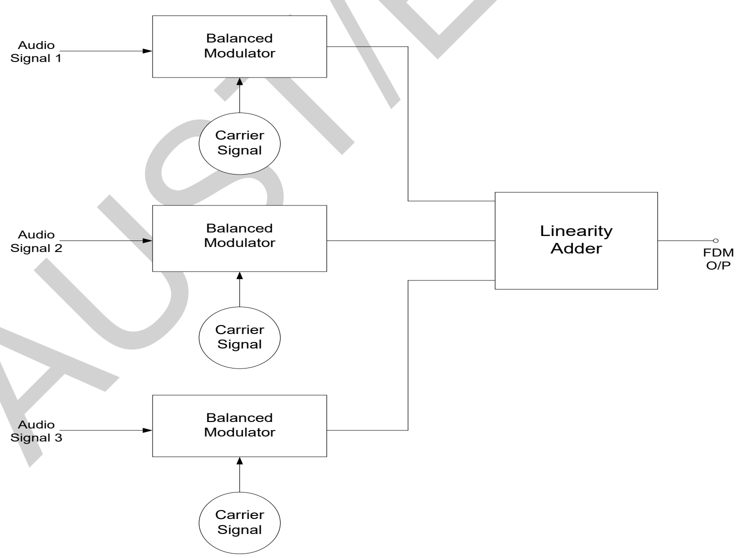


Fig. 6.1: Block diagram of FDM multiplexer.

Fig. 6.1 is the system block diagram of FDM. Like TDM, FDM is used to transmit multiple signals over the same communications channel simultaneously. However, unlike TDM, FDM does not use pulse modulation. In Fig. 6.1, assume that all the input audio signals are low-pass pattern and after each input signal, there will be a low-pass filter. The objective is to remove all the unwanted signals except the audio signals. Then the audio signals will be sent into the modulator so that the frequency range of the signals will shift to different region. The conversion of the frequency is controlled by the carrier signal, therefore, we utilize the simplest technique, which is the AM modulation to implement the modulator. Then the modulated signals will pass through the band pass filter, which can limit the signal bandwidth to prevent the interference between each signal, Finally, all, the signals will be added by the linearity adder. As compare to TDM, we, utilize sampling to implement the TDM system und AM modulation to implement the FDM system.

2. Generation of Audio Signal (MUX block)

Fig. 6.2 is the circuit diagram of adjustable audio signal generator. We use ICL8038 to design the audio generator which can produce sine wave, triangle wave and square wave. The range of the output frequency and output amplitude are 1 Hz to 200 kHz and 0 V to 2 V, respectively. In this circuit, we only use sine wave with 2 V output amplitude and 300 Hz to 1.5 kHz output frequency. In Fig. 6.2, the VR1 is to adjust the output frequency, which can change the time of charge and discharge. The faster the time of charge and discharge (the smaller the 'value of resistor), the higher the output frequency; nevertheless, the slower the time of charge and discharge (the larger the value of resistor), the lower the output frequency. VR2 is used to adjust the output amplitude where the output amplitude can be varied from 0V to 2 V.

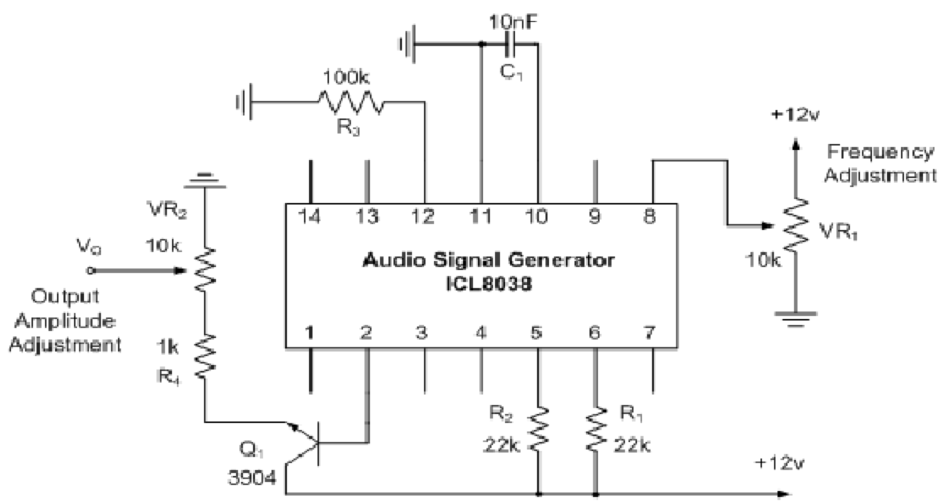


Fig. 6.2: Audio signal generator.

3. Generation of Carrier Signal (MUX block)

In this experiment, we utilize the diodes amplitude-limited Wien Bridge oscillator to implement the carrier signal generator. VR_1 can be adjusted to satisfy the condition for oscillation (Barkhausen Principle). This circuit is suitable to prevent non-linear distortion produced by larger output voltages.

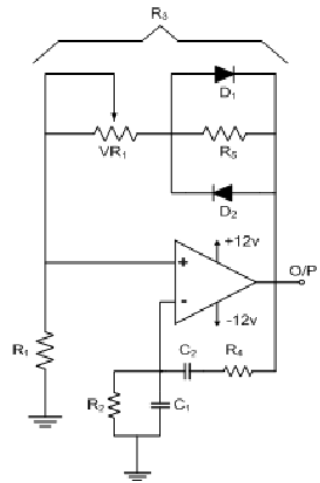


Fig. 6.3: Circuit diagram of diodes amplitude-limited Wien Bridge oscillator.

4. Generation of DSB-SC Signal (MUX block)

DSB-SC modulation is a kind of AM modulation, therefore, we can utilize the structure of AM modulator to implement the DSB-SC modulator.

Fig. 6.4 is the circuit diagram of AM modulator. We can see that the carrier signal and audio signal belong to single ended input. The carrier signal is inputted from pin 10 and the audio signal is inputted from pin 1. Therefore R_8 determine the gain of the whole circuit and R_9 determine the magnitude of bias current. If we adjust the variable resistor VR1 or change the input amplitude' of audio signal, then we can control the percentage modulation of amplitude modulation, which means we can adjust the output become the DSB-SC modulation. By adjusting variable resistor VR_2, we can control the magnitude of the output amplitude, which is also the gain

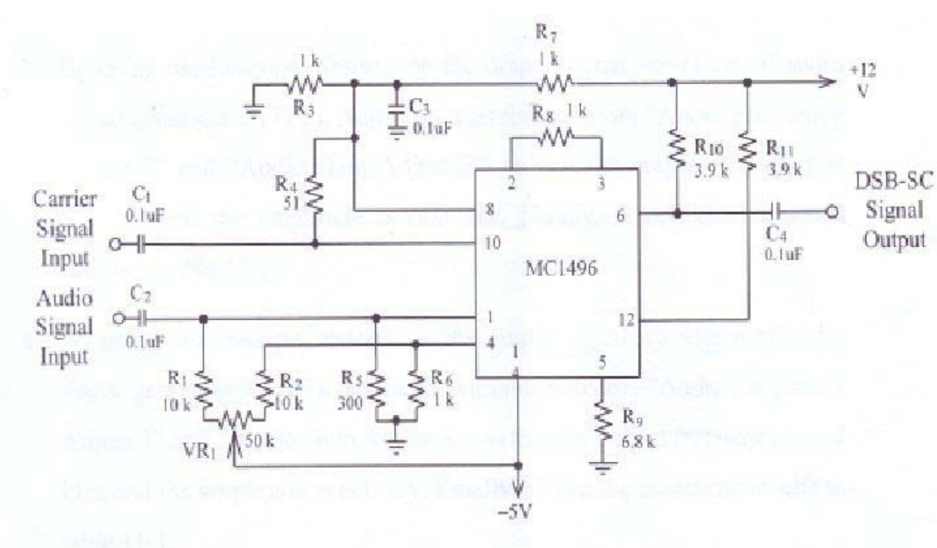


Fig. 6.4 : Circuit diagram of DSB-SC modulation by utilizing MC 1496.

5. Generation of FDM signal using Linearity Adder (MUX block)

The main objective of linearity adder is to add the three DSB-SC modulated signals to become the FDM signal.

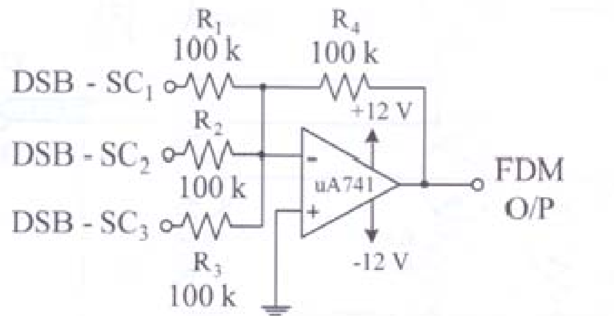


Fig. 6.5: Circuit Diagram of Linearity Adder.

6. Demultiplexing of FDM signal:

There are two ways to implement the FDM demultiplexer. The first way is shown in Fig. 6.6(a). Let the FDM -signals pass through a band pass filter and a low-pass filter. The objective of band pass filter is to remove the signal, which its frequency is larger. and lower than f_o, then only left a single DSB-SC modulated signal. After that this signal will pass through a low-pass filter, then we can recover the modulated signal and obtain the original audio signal. The second way to implement the FDM demultiplexer is shown in Fig. 6.6(b). Fig. 6.6(b) is the block diagram of synchronous product detector. After the signal passed through the synchronous product detector, we will add a low-pass filter to remove all the unwanted signal and recover the original audio signal. In this chapter, we will discuss the operation theory and the design of synchronous product detector.

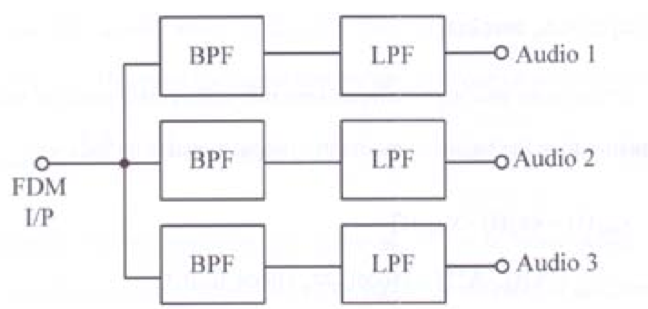


Fig. 6.6(a): Block diagram of FDM demultiplexer.

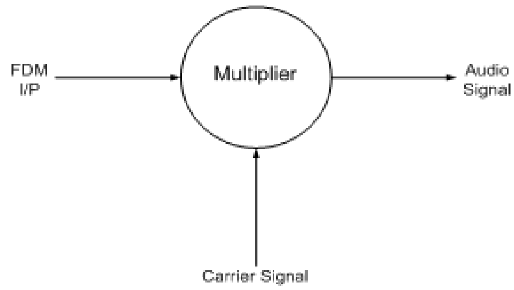


Fig. 6.6(b): Block diagram of synchronous product detector.

7. Use of Synchronous Product Dectector(DeMUX block)

Fig. 6.7 is the circuit diagram of synchronous product detector. The variable resistor VR_1 controls the input magnitude of carrier signal; variable resistor VR_2 controls the input magnitude of amplitude modulated signal; then the output signal of MC 1496 is located at pin 12. . C_7, C_9 and R_9 comprise the LPF. The DC signal is blocked by C_{10}.

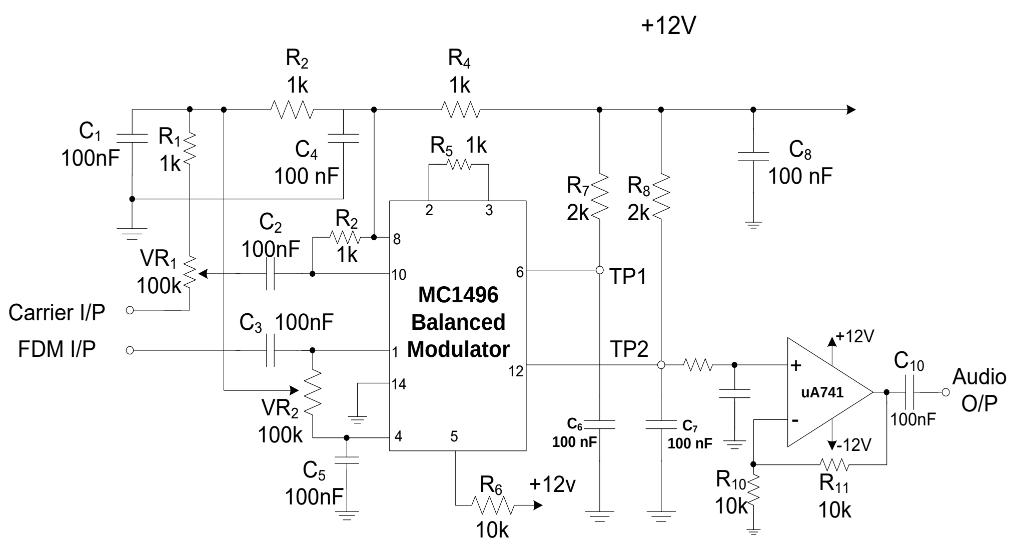


Fig. 6.7: Circuit diagram of synchronous product detector.

8. Low-pass Filter

Fig. 6.8 is the circuit diagram of second order active low-pass filter.

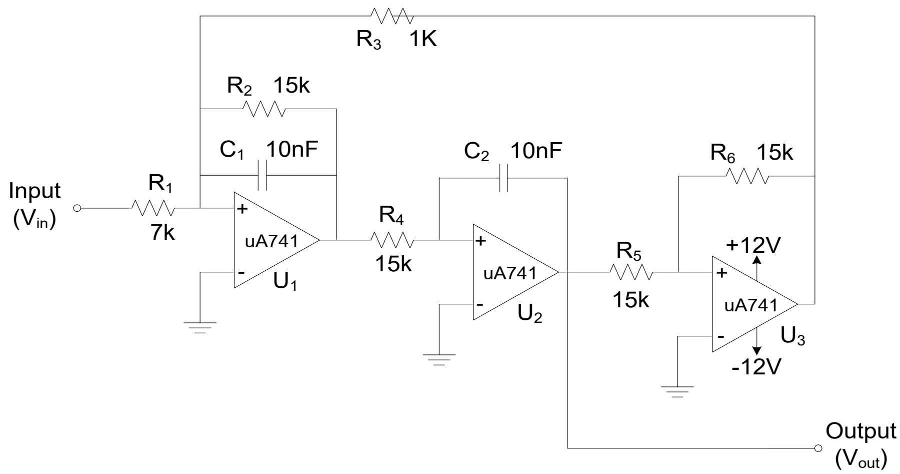


Fig. 6.8: Circuit diagram of second order active low-pass filter.

(d)Experimental PROCEDURE:

FDM Multiplexing:

1. Observe the output signal waveform of audio signal generator 1 (TP 1). Adjust the variable resistors “Audio Frequency, Adjust 1” and “Audio Gain Adjust I”, so that the output frequency is 500 Hz and the amplitude is 600 mV.

2. Repeat the same for audio signal generator 2 & 3 (TP3,7). Adjust the variable resistors “Audio Frequency Adjust 2 & 3” and “Audio Gain Adjust 2 & 3” respectively, so that the output frequency and amplitude is 800 Hz & 600 mV for signal generator 2 and 1.2 kHz and 600 mV for sig. gen.3 respectively.

3. Observe the o/p waveform of balanced modulator 1-3 (TP, 6,9). Adjust the variable resistor “Modulator Adjust 1-3” respectively, so that the output signal is DSB-SC modulated signal.

4. Observe on the output signal waveform of FDM output port (FDM O/P).

FDM Demultiplexing:

1. Follow step 1-3 of FDM multiplexing experiment.

2. Observe the output waveform of TP3, TP6 and TP9. Adjust variable resistors "Mod Adjust 1", "Mod Adjust 2" and "Mod Adjust 3" `so that the o/p is the modulated DSB-SC signal.

3. Connect the modulated FDM signal (FDM O/P) of Tx to the input terminal (FDM I/P) of Rx. Connect the corresponding three carrier signals of Tx to the three input terminals of the carrier signal (Carrier I/P) of Rx.

4. Observe the output signal' waveforms of the audio signal 1-3 (Audio O/P 1). Then adjust variable resistors "Carrier Adjust, 1-3" and "Gain Adjust 1-3" respectively, so that the output amplitude is maximum without distortion. Finally, record the measured results.

(e) Report:

1. Draw the circuit diagram of audio signal generator, DSB-SC modulator by utilizing MC1496 and linearity adder. Comment on the operating principle of each circuit.
2. Anything else instructed by the teacher.
3. Discussion expressing originality of the student.

Experiment No: **7**
Name of the Experiment: **Study of Pulse Code Modulation**

(a) Objective:

1. To understand the theory of Pulse Code Modulation (PCM).
2. To understand the advantages of using a Digital Signal in data TX/RX process.
3. To understand the different stages required to generate a PCM signal.
4. The implementation process of transmission and reception of a PCM signal.

(b) Equipments:

1. MODICOM 3/1 and MODICOM 3/2 PCM trainer
2. Oscilloscope
3. Power Supply (IC Power 60 Module)
4. Multimeter

(c) Theory:

Pulse code modulation (PCM) is a process in which analog signals are converted to digital form. The analog signal is represented by a series of pulses and non-pulses (1 or 0 respectively). The advantage of PCM is that the stream of pulses and non-pulse streams of 1's and 0's are not easily affected by interference and noise. Even in the presence of noise, the presence or absence of a pulse can be easily determined. Moreover, digital signals are easy to process by cheap standard techniques. This makes it easier to implement complicated communication systems such as telephone networks. The block diagram of a general PCM system is given below:

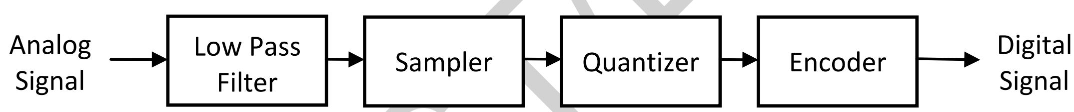


Fig 7.1:-Block Diagram of PCM System

The **filtering** stage removes frequencies above the highest signal frequency. These frequencies if not removed, may cause aliasing problems when the signal is processed through the stage of sampling. **Sampling** of a waveform means determining instantaneous amplitudes of a signal at fixed intervals. **Quantization** is the process of allocating levels to the infinite number of possible amplitudes of sample values of the analog signal.

In this example (Fig:-7.2) the maximum amplitude value of the sampled signal is +8V and the minimum is -8V.

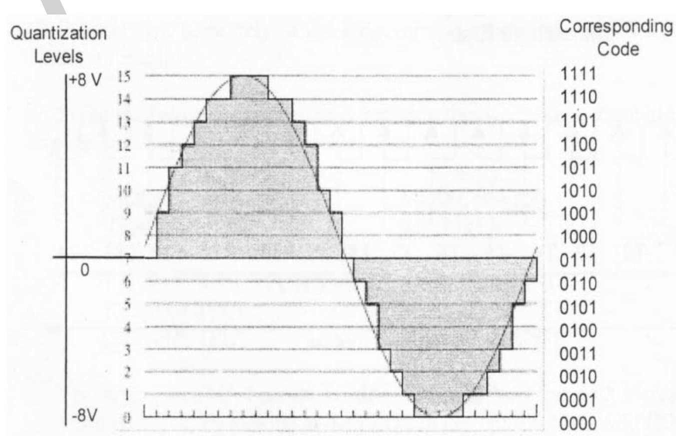


Fig 7.2:-Quantization levels and corresponding code of PCM

The voltage full range (from -8 to +8) of values is 16 volts. For 4-bit PCM system, this 16 volt range is divided into 2^4 or 16 quantization levels. In the **encoding** process each step level is assigned a 4-bit binary code. The code starts with zero at the lowest level. This will be the last part of the conversion and the PCM signal will then be transmitted/sent. One disadvantage of PCM is that the signal accuracy is reduced because of the quantizing of the samples.

The MODICOM 3 PCM has a transmitter and a receiver module. Two analog input channels are provided; the transmitter samples each channel in turn, and converts each sample into a 7-bit binary code. The binary codes from the two channels are time-division multiplexed together and transmitted serially to the receiver where they are demultiplexed, decoded and low pass filtered to obtain the original analog channel signal. PCM system can be operated at two speeds, FAST and SLOW. In FAST mode each analog channel is sampled at a rate of 16 KHz, so that real-time operation can be observed. In SLOW mode, all the signals and data are slowed by a factor of approx. 250000 resulting in a sampling period of approx. 15 seconds.
The sequence of operations at the Transmitter is synchronized to the Transmitter's clock, which appears at the TX CLOCK output on the Transmitter board. Each clock cycle is known as a 'timeslot'. Operations repeat every 15 timeslots, numbered 0 to 14, and one complete cycle of 15 timeslots is known as one Transmitter 'timing frame'. The Transmitter's TX T0 output (which stands for 'Transmitter timeslot 0') goes high during timeslot 0. The data appearing at the Transmitter's TX DATA OUTPUT is as follows:

Timeslot 0: This timeslot is reserved for outputs from the SYNC CODE GENARATOR block. If the SYNC CODE GENERATOR is OFF, a '0' is transmitted.

Timeslots 1 to 7: These timeslots carry a 7-bit data word corresponding to the last sample taken of analog input Channel 0. The least significant bit is transmitted first.

Timeslots 8 to 14: These timeslots carry a 7-bit data word corresponding to the last sample taken of analog input Channel 1. The least significant bit is transmitted first.

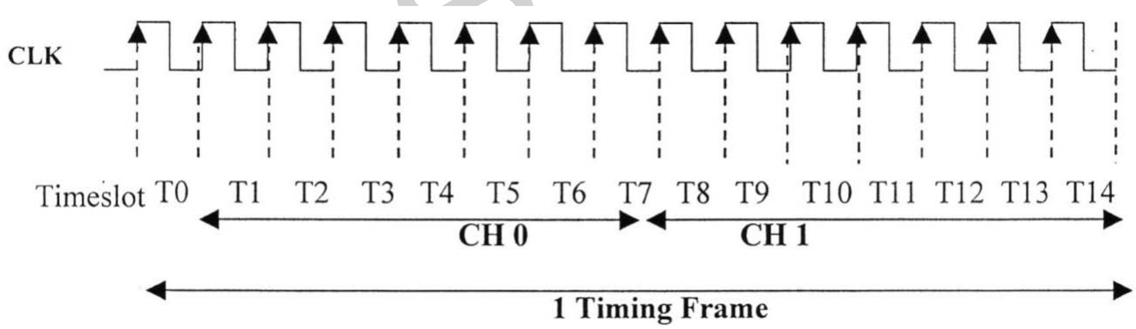


Fig 7.3:-TDM Timing Frame

(d)Experimental Procedure:

A. Mode 1 Connection

1. Connect the D.C. supplies to the boards as illustrated in Fig-7.4

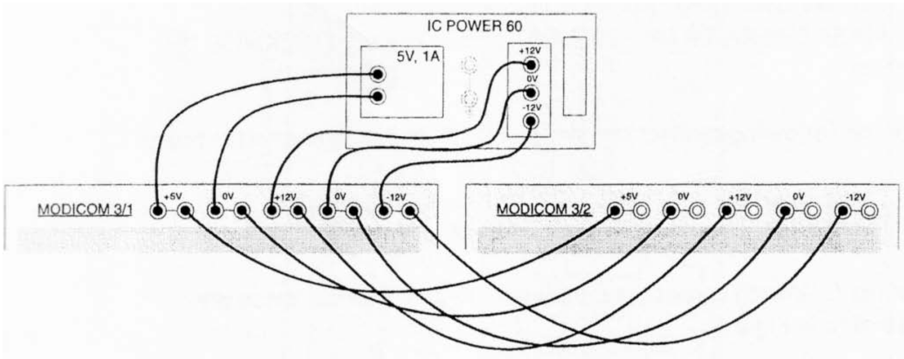


Fig 7.4:-Power Supply Connection

2. Ensure the following initial conditions exist on the modicum 3/1 Transmitter board:
    a. MODE switch in FAST position.
    b. SYNC CODE GENERATOR 'off/on' switch in OFF position.
    c. ERROR CHECK CODE SELECTOR switches in 00 (off) position.
    d. All four SWITCHED FAULTS off.

3. To observe the quantization, use a DC signal as transmitted signal. Make the following links on the Transmitter board:
    - D.C. 1 to CH.O input
    - CH.O input to CH. 1 input.

    The same D.C. level is now present at the Transmitter's CH.0 and CH.1 inputs. By adjusting the Transmitter's D.C. 1 preset position, this D.C. level can be adjusted between approx. -5.8V and +5.8V. Varying the input voltage over this range takes The Transmitter's A/D output through its full range of 00 to 7F hex.

4. In order to correctly interpret the data stream, the Receiver must be clocked at the same rate as the Transmitter and it must be able to tell which timeslot is which i.e. the receiver must be synchronized with transmitter. In Mode 1 connection, the TX CLOCK is sent separately to the Receiver and another signal TX TO tell the Receiver which is timeslot 0.

5. Make the following connections between the two boards, these are illustrated in Fig:-7.5

MODICOM 3/1 TRANSMITTER		MODICOM 3/2 RECEIVER
TX clock output	to	RX clock input
TX to output	to	RX sync input
TX Data output	to	RX data input

6. Try varying the D.C. 1 preset position, and note how the A/D converter's 7-bit output word changes, by looking at the LED's in the Transmitter's A/D CONVERTER block.
    Also observe that the same 7-bit data word appears at the Receiver's D/A input (as indicated on the LED's in the Receiver's D/A CONVERTER block), showing that the TX and the RX are synchronized.

7. Display transmitter CH.0 input (t.p. 10) and Transmitter TX DATA OUTPUT (t.p.44) on the oscilloscope, using the former for scope triggering purposes. This clearly shows the data stream being sent from Transmitter to Receiver.

8. To observe the digitization of sine waves, make the following connections on the MODICOM 3/1 Transmitter board:

~ 1 kHz to CH.0 input
~ 2 kHz to CH.1 input

9. Display Transmitter CH.0 and CH. 1 inputs (t.p. 10 and 12) on an oscilloscope. Note that the 1 kHz input appears on input Channel 0, and the 2kHz sine wave appears on input Channel 1.

10. Display Transmitter CH.0 input (t.p. 10) and Transmitter TX DATA OUTPUT (t.p.44) on the oscilloscope, using the former for 'scope triggering purposes. This clearly shows the data stream being sent from Transmitter to Receiver. Vary the Transmitter's ~1 kHz and ~2 kHz presets (these vary the amplitude of the two sine waves), and note that the transmitted data changes.
11. To examine the system's timing in depth, switch both transmitter and receiver to SLOW mode and observe the data flows on the LED's.

12. Observe Receiver CH.0 and CH. 1 outputs (t.p.33 and 36) on an oscilloscope. Note that the two sine waves have been successfully sent from the Transmitter to the Receiver, and that each appears on the correct output channel.

B. Mode 2 Connection

The number of connections between Transmitter and Receiver can be reduced to two by sending a pseudo-random synchronization code sequence' within the transmitted data stream. This is a sequence of 15 bits, which are produced by the Transmitter's SYNC CODE GENERATOR. The actual sequence is:

000100110101111 repeating

One bit of this sequence is transmitted in each Transmitter timing frame, during Transmitter timeslot 0. At the Receiver, this sequence can be detected by the SYNC CODE DETECTOR block, and used to decide which incoming data bit corresponds to which Transmitter timeslot.

1. To observe Mode 2 operation, make the following connections on the MODICOM 3/1 Transmitter board:
   - D.C. 1 to CH.0 input
   - CH.0 input to CH. 1 input.

2. Make the following connections between the two boards; these are illustrated in Fig-7.6

MODICOM 3/1 TRANSMITTER		MODICOM 3/2 RECEIVER
TX CLOCK OUTPUT	to	RX clock input
DATA output	to	RX data input

3. Return both Transmitter and Receiver to FAST mode. Put the Transmitter's SYNC CODE GENERATOR switch in the ON position. Put the Receiver's SYNC CODE DETECTOR switch in the ON position.

4. Vary D.C. 1 and note that the Transmitter's A/D converter output and the Receiver's D/A CONVERTER input are always the same as each other. Note also that the 'SYNC BIT COUNTER' LED in the Receiver's SYNC CODE DETECTOR block is on; indicating that the Receiver knows which Transmitter timeslot is which. We say that the Receiver is 'frame synchronized' to the Transmitter. Once frame synchronization has been achieved, the TX.T0 and RX.T0 signals will be coincident with each other.

5. Now briefly switch the Transmitter's SYNC CODE GENERATOR off, and note that the Transmitter's A/D output and the Receiver's D/A input no longer match each other. Note also that the 'SYNC BIT COUNTER LED is now off; indicating that frame synchronization

has been lost. Switch the Transmitter's SYNC CODE GENERATOR on again, and note that the two boards once again become frame synchronized.

C. Mode 3 Connection

The number of connections between Transmitter and Receiver can be further reduced to only one by removing the clock connection between Transmitter and Receiver. The Receiver clock can then be generated by the Receiver's CLOCK REGENERATION CIRCUIT. This block contains a Phase- Locked Loop which generates a clock signal synchronized to rising transitions of the incoming data stream. This single-connection mode is known as 'Connection Mode 3' and is illustrated in Fig:-7.7.

To demonstrate this mode, take the following steps:

1. Ensure that both boards are in FAST mode.

2. REMOVE the TX CLOCK OUTPUT-RX CLOCK INPUT link between Transmitter and Receiver.

3. At the Receiver, MAKE the following links:

   RX DATA INPUT to INPUT of CLOCK REGENERATION circuit.
   OUTPUT of CLOCK REGENERATION circuit to RX CLOCK INPUT.

4. Before connection mode 3 is used, it may be necessary to trim the frequency of the Voltage-Controlled Oscillator (VCO) in the receiver CLOCK REGENERATION CIRCUIT, to ensure that the generated clock signal remains synchronized to the incoming data signal.
   To ensure the VCO frequency is correctly adjusted, follow the steps below:
   a) Turn the D.C. preset fully clockwise.
   b) Turn the VCO FREQUENCY ADJUST preset, until a position is found where the SYNC BIT COUNTER LED, in the SYNC CODE DETECTOR block is ON.
   c) Turn the D.C. 1 preset fully counter-clockwise and check that the SYNC BIT COUNTER LED is still ON. If the LED switches off, retire the VCO FREQUENCY ADJUST preset until the LED stays on for both extreme positions of the D.C. 1 preset.

5. At the transmitter, remove the current connections to the two analog inputs, and make the following connections;

   ~ 1 kHz to CH0
   ~ 2kHz to CH.1

   Adjust the Transmitter's ~1 kHz and ~2 kHz presets until the Transmitter's input signals on t.p. 10 and t.p. 12 are both 8V pk/pk.

6. By examining the Receiver's analog outputs (t.p. 33 & 36), verify that the Transmitter's sinewave inputs have been correctly reconstructed at the Receiver.

(e) Report:

Should include

1. Block diagram of transmitter and receiver block
2. Comment on the function of various blocks.
3. Discuss about Synchronization.
4. Anything else instructed by teacher.
5. Discussion expressing originality of the student

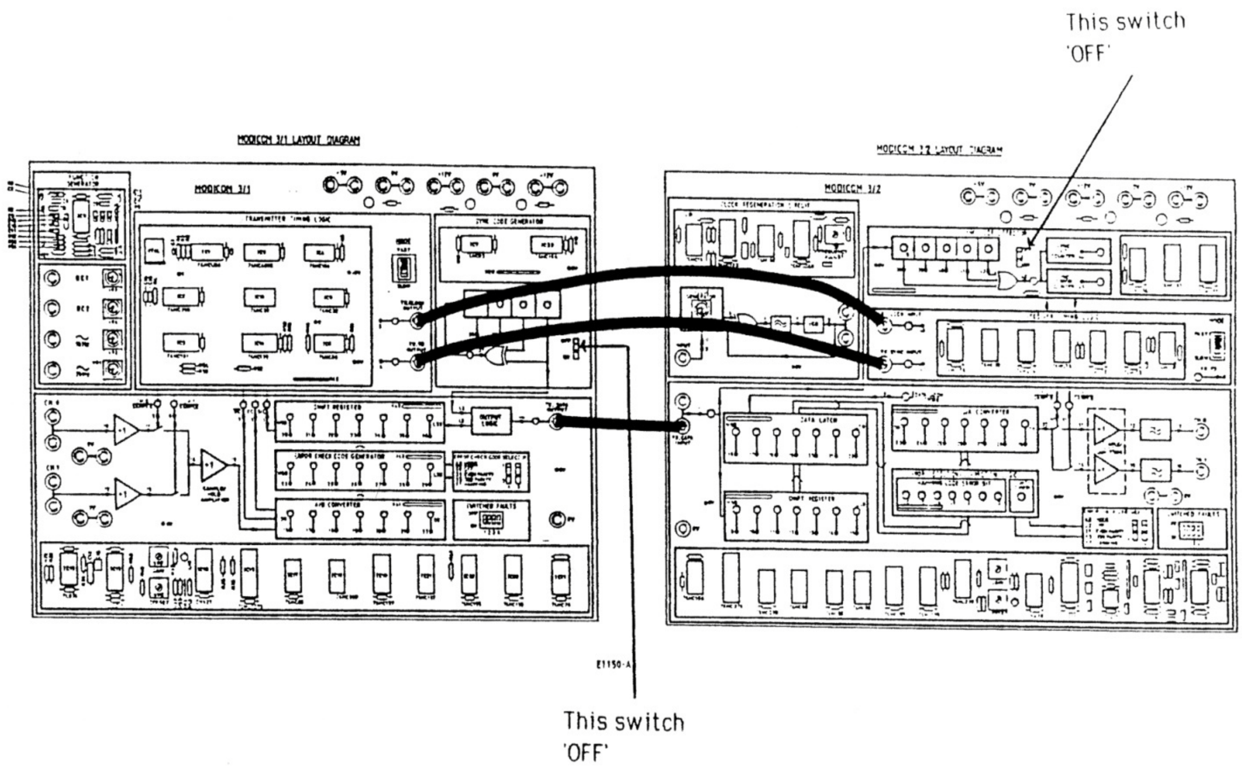


Fig 7.5:- Mode 1 Connection

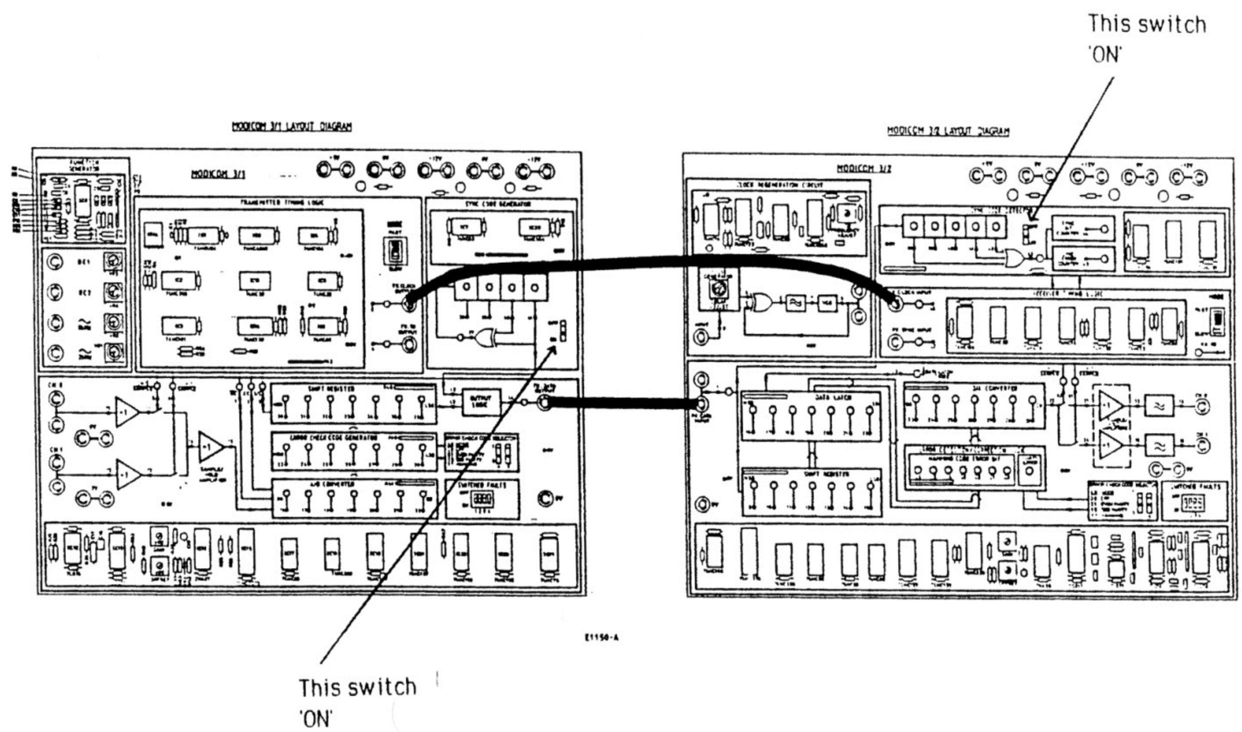


Fig 7.6:- Mode 2 Connection

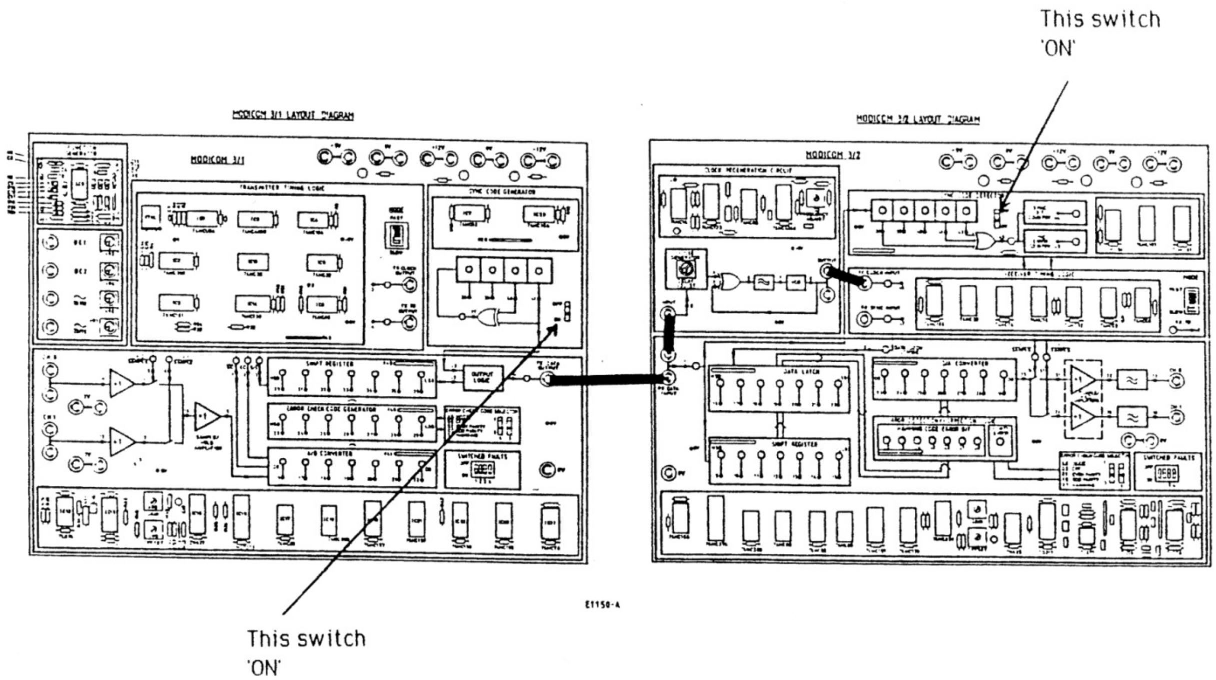


Fig 7.7:- Mode 3 Connection

Experiment No.: **08**
Name of the Experiment**: Study of Communication Using Optical Link**

a) Objective:

1. To understand the theoretical aspects of communication using optical link.
2. To review the amplitude modulation & frequency modulation techniques.
3. To examine the amplitude modulation & frequency modulation techniques using optical link.
4. To observe die effect of analog and digital data on modulated signal. 4. To know the real life applications of communication using optical link.

b) Equipment:

1. Modicom board (6).
2. Power supply.
3. Set of connection leads.
4. Oscilloscope.

c) Theory:

I. Optical communication link & their application

An optical fibre (Fig. 8.1) is a strand of glass or plastic with special optical properties. It enables light to travel a long distance down its length. By converting electrical signal into light (at transmitter), sending this light down a length of optical fiber (communication channel) and reconstructing the electrical signal (in a receiver) an optical communication link is formed (Fig.8.2). Usually, LED, LASER etc. are used as optical sources as they can convert electrical signals into light. They can be used both for analog modulation (when light from LED vary in continuous manner) and digital modulation (when discrete change of light intensity is obtained corresponds to digital signals) purposes. Again, p-i-n diodes, photodiodes. Avalanche photodiode etc. are used as optical detectors to convert light signals into electrical signals.
AC amplifier in receiver circuit increases the amplitude of received signal and removes dc component which is present: at detector output.
Because of enormous bandwidth, smaller size-weight, immunity to interferences, lower transmission losses etc. optical fiber based communication system are widely used in telephone network, computer communication, railway link, military communication, satellite communication etc.

II. Amplitude modulation & its benefits

It's a kind of analog modulation technique in which the amplitude of message signal are used to vary amplitude of analog carrier (Fig.8.3). The detail of AM was studied in Experiment 1,2 of EEE-3208. Here, the optical communication board is used to demonstrate the amplitude modulation of a light source with both analog and digital signal, the transmission of the modulated signal via fiber-optic cable and the recovery of original signal at the receiver. Modulation of the light beam by digital data is advantageous because small change in the amplitude of detector's O/P have no effect on the receiver's final O/P. Also non-linearity's within EMITTER LED and PIN diode cannot distort received signal because detector O/P is squared up at the comparator O/P.

III. Frequency modulation & its benefits

It's also a kind of analog modulation technique in which the amplitude of message signal are used to vary frequency of analog carrier (Fig.8.3). The detail of FM was studied in Expt.3 of EEE 3208. Here, the optical communication board is used to demonstrate the change of frequency of a digital signal which is then used to switch a light source on and off. At the receiver, the incoming digital pulses are squared up and the original analog signal extracted from the changing frequency of the digital signal. The digital signal is used to vary some characteristics (frequency) of a digital signal. The digital signal is used to switch light beam on and off. At receiver, incoming pulse train is simply squared up by a voltage comparator and demodulated to extract the original signal. Here, a PLL (phase locked loop detector)is used to produce a signal whose average level is proportional to the" frequency of digital stream. The average level is extracted by a LPF and amplified by AC amplifier.

d) Experimental procedures:

I. Amplitude modulation (during transmission of analog signal)

1. Connect supplies to MODICOM 6 board
2. Ensure that all switched faults are off.
3. Make the connection shown as shown in Fig 8.4. The thick lines indicate connection via 4mm leads, while the thinner lines between Emitter LED's and PIN diodes represent fibre optic links.
4. Switch EMITTER 1's driver to analog mode. This ensures that the current through emitter LED (and hence LED's brightness) is changed linearly with the analog voltage applied to the driver's output.
5. Turn the-1kHz preset in the FUNCTION GENERATOR block to fully CW (maximum amplitude) position.
6. Turn on the power to the board.
7. Monitor the i/p of the EMITTER bock at tp.5. This 1 KHz sine wave is being used to amplitude modulates EMITTER 1's emitter LED.
8. Briefly disconnect the fiber optic cable from DETECTOR1's PIN diode. The transmitted light is clearly seen at the end of the cable. This light beam is actually being amplitude modulated by die 1 KHz sine wave, although the modulation is at too high a frequency to be detected with the eye. Reconnect the fiber optic cable to DETECTOR1's PIN diode before continuing.
9. Examine the O/P of DETECTOR1 at tp.10, check that smaller version of reconstructed electrical signal is detected at the receiver,
10. Monitor O/P of AMPLIFIER 1 (tp.28), adjust the gain adjust preset until monitored O/P has the same amplitude as that applied to EMITTER1 i/p (tp5).
11. While monitoring tp28, change the amplitude of the modulating 1 KHz sine wave by varying 1 KHz preset in the function generator block and observe the effect
12. Carefully flex the fibre optic cable (means bend radius) and note that output amplitude changes. This hapens wnen direct modulation of light is taken place because it causes change in attenuation of the light beam. Also observe the same effect by varying the length of Optical fibre (if possible)

II. Amplitude modulation (during transmission of digital signal)

1. Make the connection shown as shown in Fig 8.5.
2. Switch EMITTER 1's driver to digital mode. This ensures that the fast changing digital signal applied to driver's i/p cause emitter LED to switch quickly between 'on' and 'off state.,
3. Monitor the i/p of the EMITTER1 block at tp.5. This 1 KHz square wave is being used to amplitude modulate EMTTTER1's emitter LED,
4. Examine the O/P of DETECTOR1 at tp.10. check that smaller version of reconverted electrical signal is detected at the receiver.
5. Monitor both i/p to COMPARATOR1 at tp.13,14 and if necessary, slowly adjust the comparator's BIAS preset, until the d.c level on the '-' i/p at tp.13 lies midway between the high and low levels of the signal on the '+' i/p at tp.14. This dc level is the comparator's threshold level.
6. Examine O/P of comparator1 (tp15).
7. Carefully flex the fibre optic cable and note that this time there is no change in the final O/P. the O/P amplitude is independent of bend radius of the cable and of the length of the cable, providing that the detector's O/P signal is large enough to cross the comparator's threshold level.

III. Frequency modulation

1. Connect supplies to MODICOM 6 board
2. Ensure that all switch faults are off.

3. Make the connection shown in Fig 8.6. The thick lines indicate connection via 4mm leads, while the thinner lines between Emitter LED's and PIN diodes represent fibre optic links.
4. Switch EMITTER'S driver to digital mode. This ensures that fast-changing digital signals applied to the driver's i/p cause the EMITTER LED to switch quickly between 'ON' and 'OFF' states.
5. Turn the 1 KHz preset in function generator block to fully CCW position.
6. Turn on the power to the board.
7. Monitor the O/P of VCO (tp.2) in Frequency modulator block (frequency of digital signal will be constant).
8. Examine the O/P of DETECTOR1 (tp.10) - successful detection is checked.
9. Monitor both i/p to COMPARATOR1 at tp.13, 14 and slowly adjust the comparator's BIAS preset, until the d.c level on the '-' i/p at tp.13 lies midway between the high and low levels of the signal on the '+' i/p at tp.14.
10. Examine tp.28 (O/P of amplifier). Then vary 1 kHz preset to observe the same at tp. 28. If required vary gain adjust of amplifier so that two signals are equal.
11. Carefully flex the fibre optic cable and note that once again there is no change in the final output. The O/P amplitude is independent of bend radius of the cable and of the length of the cable, providing that the detector's O/P signal is large enough to cross the comparator's threshold level.
12. Also observe the wave shape at i/p and O/P of various blocks.

e) Report:

1. Comment on the block diagram, merits, demerits and applications of optical communication system.
2. Submit all experimentally observed wave shapes.
3. What is the purpose of EMITTER, Comparator, Detector, PLL block, frequency modulator block, LPF, AC amplifier in this experiment? Discuss.
4. Comment on the merits and demerits of AM and FM for both analog & digital data transmission.
5. Anything else instructed by teacher.
6. Discussion expressing originality of the student.

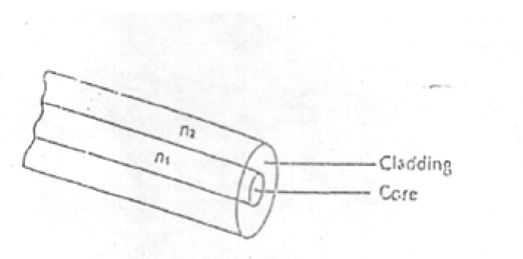


Fig 8.1:-Optical Fiber Core and Cladding

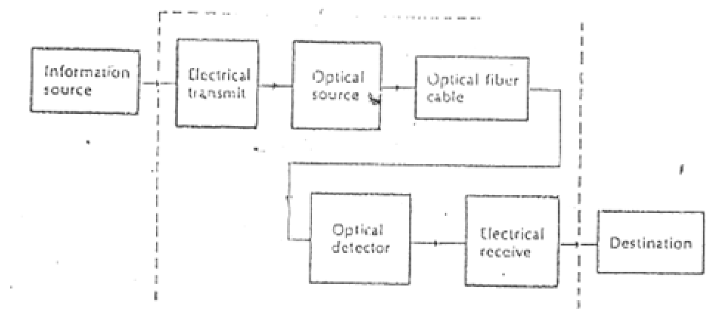


Fig 8.2:-Block Diagram of Communication System

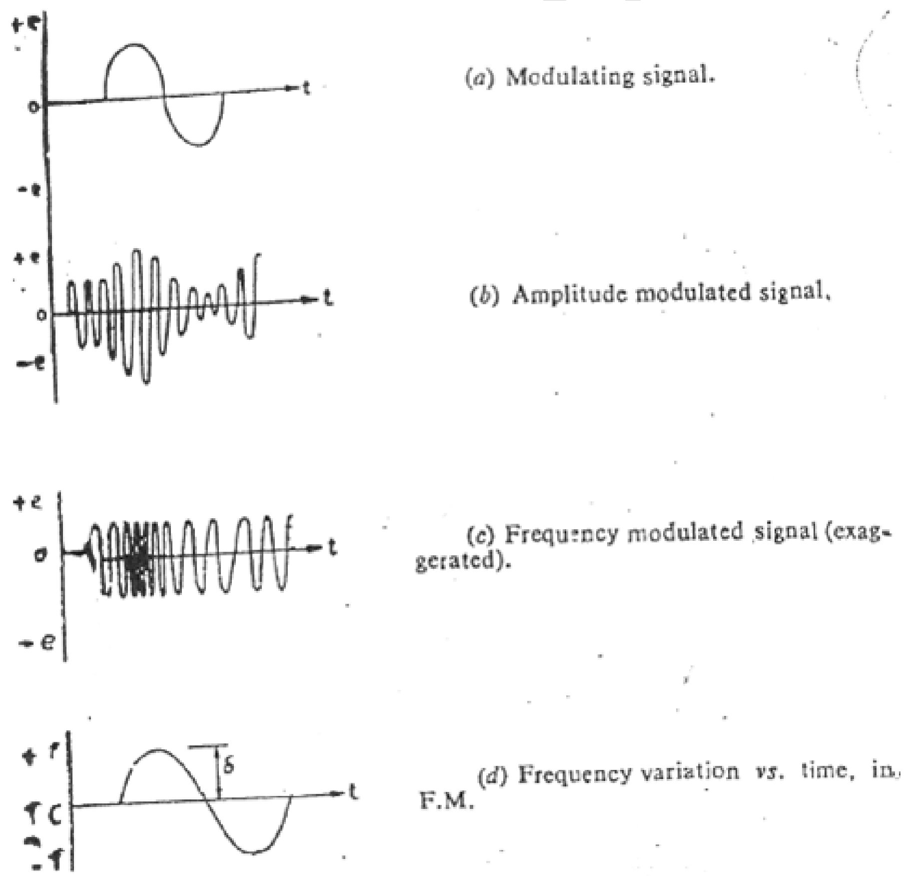


Fig.8. 3- Various modulation waves

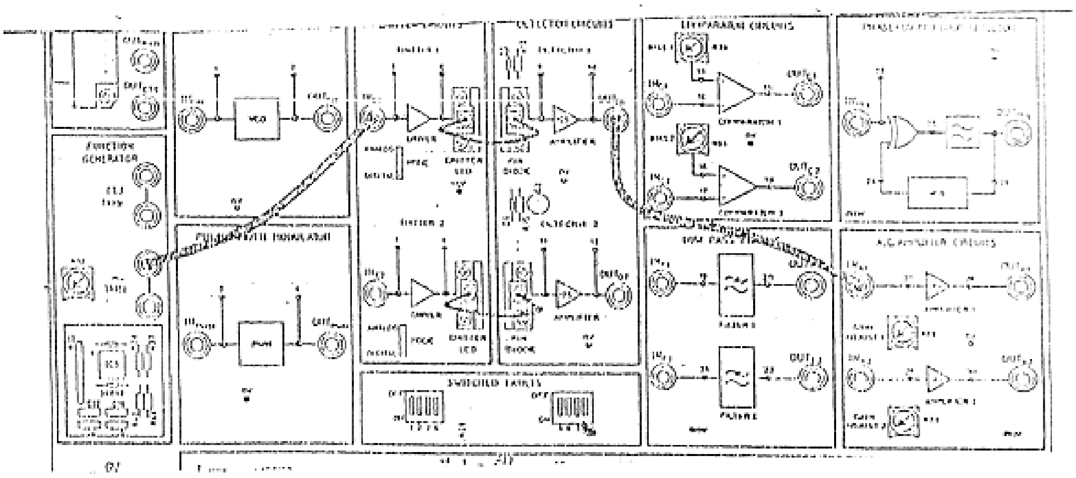


Fig: 8. 4- Amplitude Modulation (i/p analog signal)

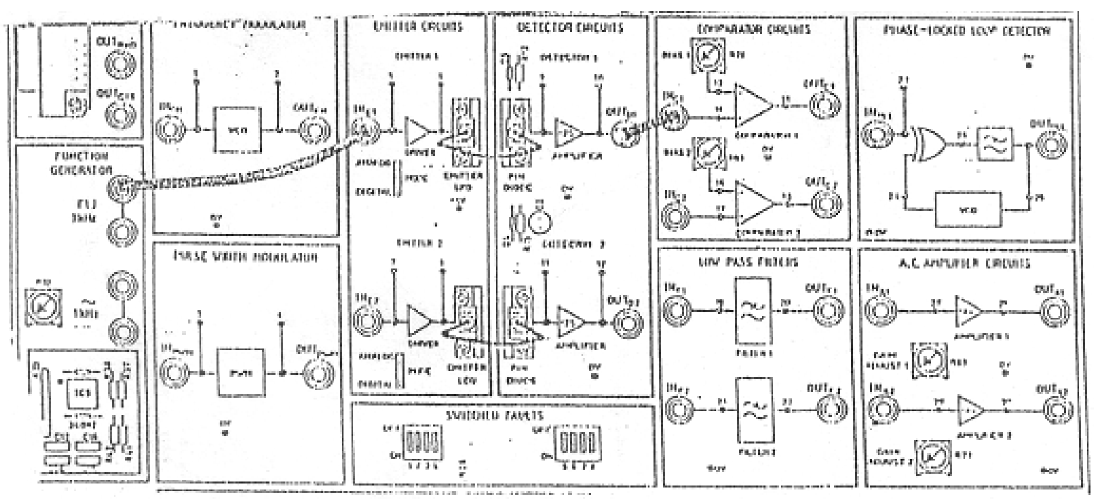


Fig: 8.5- Amplitude Modulation (i/p Digital signal)

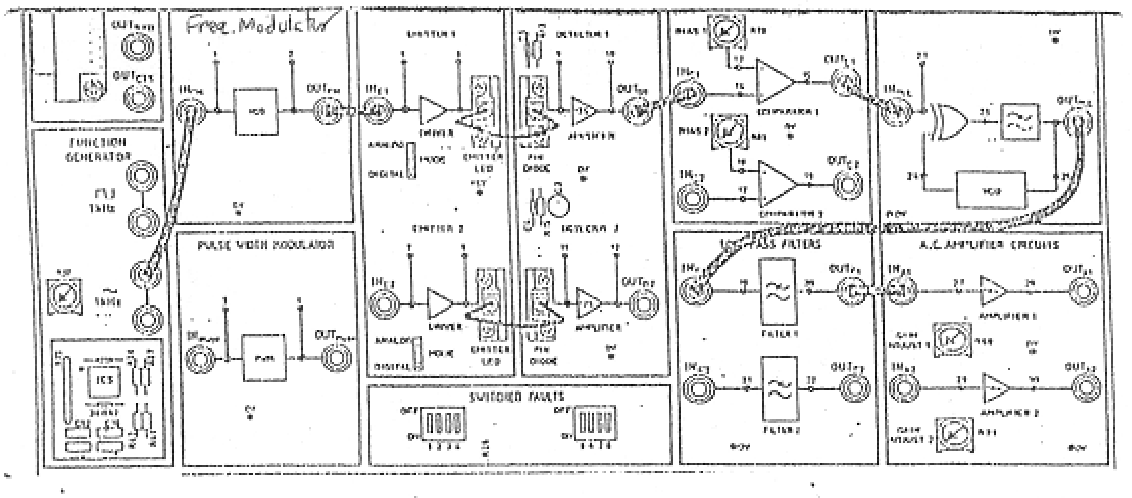


Fig: 8.6- Frequency Modulation (i/p analog signal)

Experiment No.: **09**

Name of the Experiment**: Study of Delta Modulation and Demodulation**

a) Objective:

1. To study the method of signal digitization using Delta Modulator.
2. To study the method of signal reconstruction using Delta Demodulator.
3. To observe the effect of slope Overload problem and its solution.

b) Equipments:

1. Trainer board
2. Oscilloscope
3. Power supply

c) Theory

(I) Delta Modulation

Delta modulation is a modulation technique in which just 1 bit is sent per sample to indicate whether the signal is larger or smaller than the previous sample. This system has the merits of having extremely simple coding and decoding procedures. Further the quantizing process is also very simple. But it has the drawback that it cannot easily handle rapid changes in magnitudes and as a result quantizing noise tends to be quite high.

The block diagram as shown in Fig. 9.1, the subtraction between the low frequency signal x(t) and the signal x_s (t) will produce a difference signal d(t), where x_s (t) is a reference signal, which is the former sampled value. Therefore the expression of the difference signal d(t) is given as $d(t) = x(t) - x_s(t)$, d(t) will be converted to a limiter to obtain a signal $\Delta(t) = \Delta$ or $-\Delta$, if $d(t) \geq 0$ so, $x_s(t)$ is lower than low frequency signal x(t) so estimated value to small, so next estimated value is to be increased by Δ ,

if $d(t) \leq 0$ next estimated value is to be decreased by Δ .

So, every sampling value is related to the former sampling value and therefore can be estimated based on the former sampling value. It can greatly save the transmission bandwidth. For example, if the PCM signal after encoded is 8 bits, then the transmission bandwidth will be

$B_T \geq U_s / 2 = 4f_s = 8$ W, which is 8 times more than the original bandwidth 'W'. However delta modulation can reduce the transmission bandwidth and achieve the quality of transmission as PCM modulation.

Assume the sampling signal is, $s(t) = \sum\delta(t-nT_s)$; Where T_s=Sampling interval

After sampling, the expression of delta modulation signal would be

$$s(t) = \sum\delta(t-nT_s) = \sum\Delta(nT_s)\,\delta(t-nT_s)$$

Finally we integrate the delta modulation signal to take as the next reference signal

$$x_s(t) = \sum\Delta(nT_s) \int \delta(\zeta-nT_s)\, d\zeta$$

Although delta modulation has the advantages of simple structure and small transmission bandwidth, it also has the disadvantage of slope overload. As a result of delta modulation is the capacity of the variation Δ in the T_S region, therefore the maximum slope of the delta modulation is $\Delta / T_S = f_S \Delta$. If the maximum slope of the input signal exceeds $\Delta / T_S = f_S\Delta$, that means the step size is too small, then slope overload will occur in the delta modulation. Consequently the slope of the input signal must satisfy the prerequisite condition of delta modulation, which is given as follow.

Slope overload will cause the modulation signal changes cannot follow closely enough to the input signal, and then the recovery of the original signal will become distorted. In order to prevent slope overload, the slope of the input signal cannot be too high and we can also increase the values of f_s or Δ .

(II) Delta Demodulation

As delta modulation excludes the encoder, therefore the structure of delta modulation is simpler than the structure of PCM. On the other hand, the DM signal only consists of a single bit of estimated error value ($e_q(k)$), so, the required transmitted bandwidth of DM signal is smaller than the PCM system. DM signal ($x_q(t)$) is a series diversity signal ($\Delta(t)$), therefore, the structure of the delta demodulator will be easier to achieve. Fig. 9.4 is the block diagram of delta demodulation. As a result of DM signal is a series diversity signal, so we use the integrator to accumulate the series signal, then we get

$y_q(t) = \Delta(t) + \Delta(t - T_s) + \Delta(t - 2T_s) + \Delta(t - 3T_s) + \ldots$

Where $\Delta(t)$: The diversity signal, i.e. the magnitude of step value.

However, the accumulated series signal consists of high frequency harmonics, therefore, we use the low-pass filter to remove the high frequency parts. Then we can demodulate the DM signal and recover the low frequency signal, as shown in equation

$y_D(t) = L_p\{y_q(t)\} = x(t)$

From Fig.9.4, the bipolar square wave will pass through the integrator and obtain a waveform, which is similar to the audio signal. Then the output signal will pass through a low-pass filter and finally, we can obtain the audio signal.

(III) Important blocks of delta modulator-demodulator trainer

1. Delta Modulator

Fig. 9.3 is the basic circuit diagram of delta modulation. The audio signal will pass through a low-pass filter to remove the unwanted signals, which can prevent the interference from noise. The comparator is to compare the audio signal and the output signal of integrator, then the difference will be sampled by the D-type flip-flop and the output signal is a TTL digital signal. After that the output signal will feedback to integrator for integration and the output signal of integrator will again compare to the input signal to obtain the value of Δ or- Δ .

The modified circuit diagram of delta modulation is in Fig. 9.5 Here a multiplexer is added to control the gain of the integrator. This is because the gain of the integrator will affect the slope of the output signal of integrator; therefore, this method can prevent the occurrence from slope overload. U 1 is the comparator, which can compare the audio signal and the output signal of integrator, then the output square wave signal will be sampled by a D-type flip-flop and finally the output signal is the delta modulation signal. U2 is the conversion of unipolar to bipolar circuit. Since there is no output signal from integrator by inputting the unipolar square wave signal, therefore, we need to convert the unipolar signal to bipolar signal. Analog switch is a structure of multiplexer. The purpose of the analog switch is the selection of the amplified gain of integrator. When AB=00, the signal will pass through R_{14},-R_{11} and send into integrator; when AB=11, the signal will pass through R_{14} to integrator. U3 is an inverse integrator. The expression without R_{16} is given as

$v_o = -v_c = -(1/C)\int i\,dt = -(1/C)\int (v_i/R)\,dt = -(1/RC)\int v_i\,dt$

By adding a shunt resistor R_{16} between integrator U3 and capacitor C_1, we can improve the low frequency response of the integrator.

2) Delta Demodulator

Fig. 9.5 is the basic circuit diagram of delta demodulator. The D-type flip-flop is the sampler. The input CLK signal of the delta demodulator must be synchronized with the CLK signal of the delta modulator, which is the TTL signal. UI: A is the unipolar to bipolar converted circuit. As a result of the unipolar square wave signal is unable to integrate to the original audio signal, therefore, we must convert the unipolar signal to bipolar signal. U1:B is the invert integrator, which can integrate the bipolar square wave. If without adding resistor R_{16}, the output is

$v_o = -v_c = -(1/C) \int i dt = -(1/C) \int (v_i / R) dt = -(1/RC) \int v_i dt$

If the two terminals of the capacitor of the integrator shunt with a resistor, the objective is to improve the low frequency response of the integrator, which utilizes close loop gain of the inverter. Resistor R16 and capacitor C_1 can be assumed as a equivalent impedance. U1:C, R_7, R_{10}, C_1 and C_3 comprise a second order low-pass filter. Resistors R5 and R8 comprise a negative feedback, which its main function is to provide gain.

Fig. 9.6 is the circuit diagram of delta demodulator, which is modified from the basic circuit diagram in Fig. 9.5 in Fig. 9.6, we added an analog switch. It is similar to the delta modulator that we have added a multiplexer. The main function is to control the gain of the integrator. Since the gain will affect the slope of the integrator, therefore, by using the method, w, can improve the problem of slope overloading. The analog switch is similar to the structure of multiplexer. When AB=00, the signal will be sent into the integrator through resistors R_4, R_5, R_6, R_7. When AB=1 1, the signal will he sent into the integrator through resistor R_7.

d) Experimental Procedure:

(I) Delta modulator

1. To implement a delta modulator circuit (Fig.9.3), Let J2 and J3 be short circuit i.e. the connection between X_0 and X is on. At the signal input port (i/p 1), input a 2 V amplitude and 500 Hz sine wave frequency. Next at the CLK input port (i/p 2), input a 5 V amplitude and 32 TTL signal. Then observe the input signal (T1), the output port of comparator (T2), the output port of the conversion from unipolar to bipolar (T3), the output port of tunable gain (T4), the output port of integrator (T5) and the output port of delta modulation signal (O/P) by using oscilloscope.
2. Let J2 and J4 be short circuit, i.e. the connection between X_1 and X is on. At the signal input port (I/P 1), input a 2 V amplitude and 1 kHz sine wave frequency. Next at the CLK input port (I/P 2), input a 5 V amplitude and 64 kHz TTL signal. Then by using oscilloscope, observe on the output signal waveforms of T1, T2, T3, T4, T5 and O/P signal.
3. Let J1 and 3 be short circuit, i.e. the connection between X2 and X is on. At the signal input port (I/P1), input a 2V amplitude and 1.5 kHz sine wave frequency. Next at the CLK input port (I/P2), input 5V amplitude and 128 kHz TTL signal. Then observe the same wave shapes as of step-1.
4. Let J1 and J4 be short circuit, i.e. the connection between X3 and X is on. At the signal port (I/P1), input a 2V amplitude and 2 kHz sine wave frequency. Next at the CLK input port(I/P2), input a 5V amplitude and 256kHz TTL signal. Then observe the same wave shapes as of step-1.

(II) Delta demodulator

1. To implement a delta demodulator circuit (Fig.9.6), let J2 and J3 be short circuit, i.e. the connection between X_0 and X is on. At the signal input port (I/P1), input a 2V amplitude and 500Hz sine wave frequency. Next at the CLK input port(I/P2), input a 5 V amplitude and 32 TTL signal.
2. To implement a delta demodulator circuit (Fig.9.6), let J2 and J3 be short circuit, i.e. the connection between X_0 and X is on. Then connect the modulated delta signal (O/P) to the input terminal (I/Pl) of the delta demodulator. At the CLK input port (I/P2) of the delta demodulator, input 5 V amplitude and 32 kHz TTL signal. Then by using oscilloscope, observe the output signal waveforms of sampling signal output port (T1), unipolar-to-bipolar (T2), tunable gain (T3), low-pass filter (T4), integrator (T5) and signal

output port (O/P).

3. Let J2 and J4 short circuit (for both tx an rx), i.e. the connection between X1 and X is on. Let m(t) is 2V, 1 kHz and c(t) is 5V, 64 kHz TTL. Then connect the modulated delta signal (O/P) to the input terminal (I/P1) of the delta demodulator. Then observe the same wave shape as of step-2.
4. Let J1 and J3 be short circuit (for both tx and rx) i.e. the connection between X2 and X is on. Let m(t) is 2V, 1.5 kHz and c(t) is 5V, 128 kHz TTL. Then connect the modulated delta signal (O/P) to the input terminal (I/P1)of thr delta demodulator. Then observe the same wave shape as of step-2.

e) Report:

1. Comment on the operation and block diagram of delta modulator with necessary Figures
2. Show and explain the results.
3. Anything else instructed by the teacher.
4. Discussion expressing originality of the student.

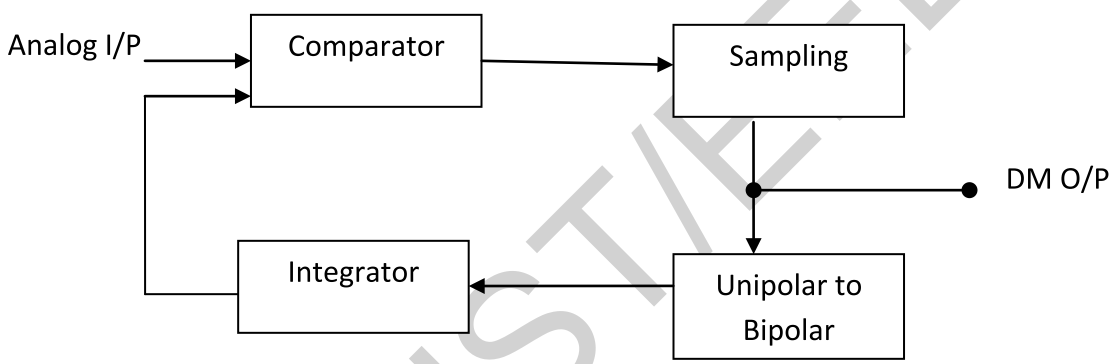


Fig. 9.1: Block Diagram of Delta Modulator.

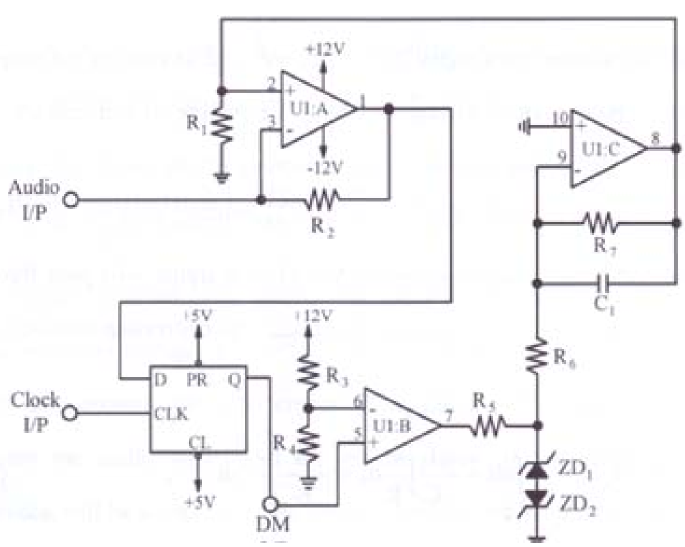


Fig. 9.2: Basic Circuit Diagram of delta Modulator.

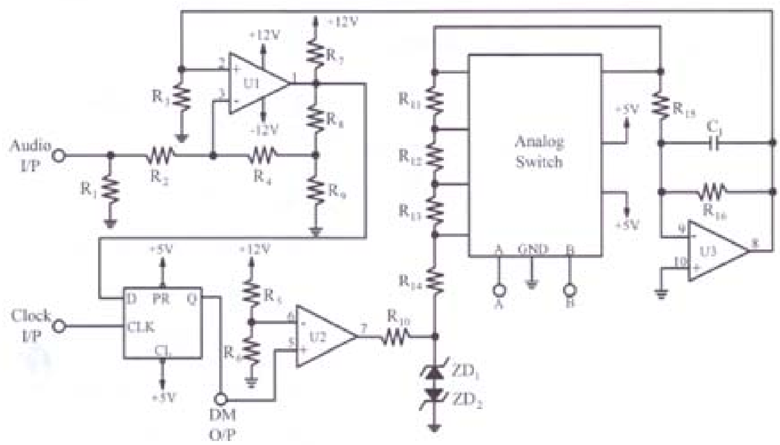


Fig. 9.3: Circuit Diagram of delta Modulator.

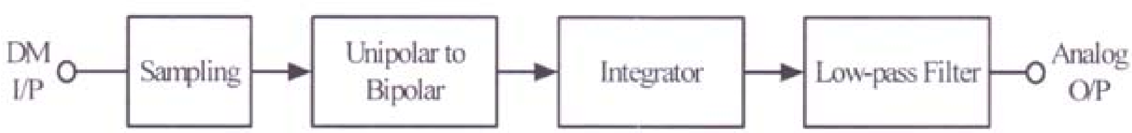


Fig. 9.4: Block Diagram of delta demodulator.

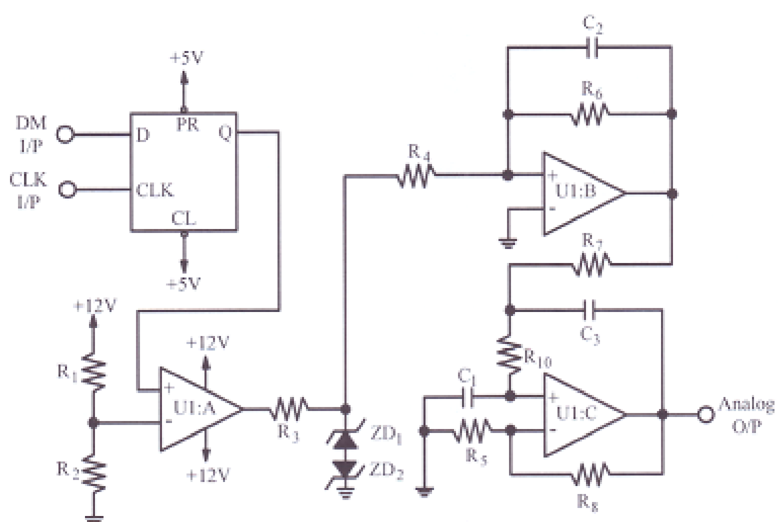


Fig. 9.5: Basic Circuit Diagram of delta demodulator.

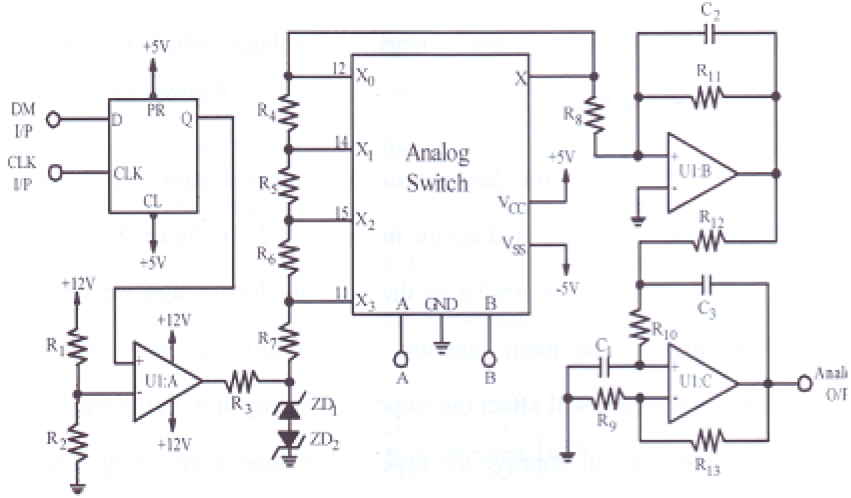


Fig. 9.6: Circuit Diagram of delta demodulator.

Experiment no: **10**
Name of the Experiment: **Study of EPABX Trainer System**

(a) Objective

1. To learn about EPABX trainer system.
2. To study about speech circuit, DTMF and switching mechanism

(b) Equipments

1. EPX-2005 Kit
2. Oscilloscope
3. Four telephone instruments
4. Console programming instrument
5. Telephone chord

(c) Theory

1. Fixed phone communication system

The main feature of PSTN system is that each subscriber is connected with the rest of the system by wired dedicated link (Fig-10.1). Each subscriber is provided with a subscriber apparatus (telephone set) which is located in a place called a Customer Premise (CP) and the telephone switch is located in a building called a Central Office (CO). The telephone set is also termed as Customer Premise Equipment (CPE). The telephone is connected to the switching system with two copper wires, often called a local loop or a subscriber loop. This a dedicated access circuit from the customer premise into the network. Telephone switches are connected with trunks. While subscriber loops are dedicated access circuits, trunks are shared connections between COs. Necessary switching function in CO is performed by a computer controlled switching system.

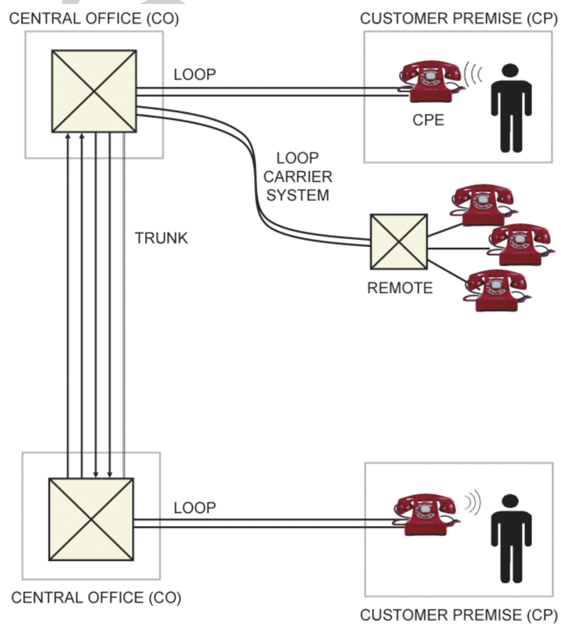


Fig 10.1:- Overview of a PSTN (Public Switched Telephone Network) System

2. The Speech circuit of subscriber apparatus

The main parts of a speech circuit include built-in transmitter, receiver and side tone circuits plus DC loop interface regulator and equalizer circuit. The DC loop interface regulator controls the voltage and current characteristics of the entire speech network depending upon the value of loop current in the subscriber line, sets the operating voltage in the integrated circuit and biases the speech circuit which is connected to the telephone line by a conventional rectifier bridge (dynamically equivalent to a small resistance in series with the signal path and a high resistance in parallel to it). External components are used to adjust transmit, receive and side tone gains and frequency response.

Side tone - It is the reproduction in the receiver of sound picked up by the microphone of the same telephone instrument. To reduce side tone, hybrid circuits are used.
Hybrid circuit - It connects two 2 wire circuits to a single 2 wire circuit. It isolates transmitted signal from the received signal so that side tone is removed (Fig-10.2 and Fig-10.3).

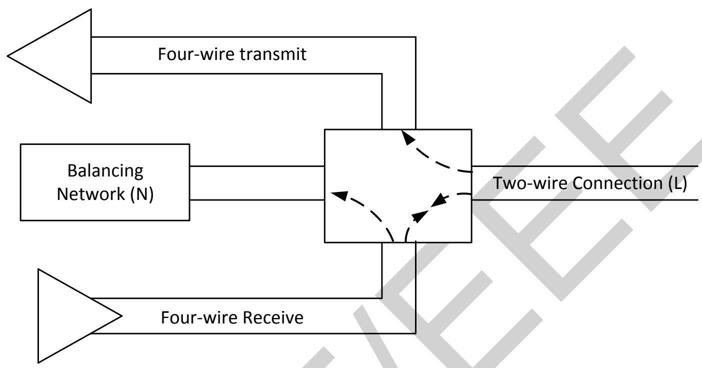


Fig 10.2:- Operation of a hybrid transformer.

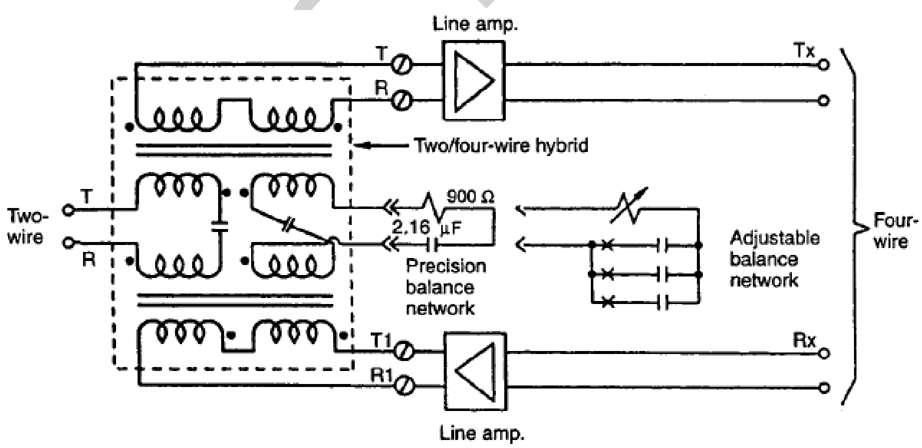


Fig 10.3:- Schematic diagram of two-wire to four-wire conversion using a hybrid.

3. Dual tone multi-frequency (DTMF)

Modern telephone sets are equipped with push button keypad with 12 keys for 0-9,* (asterisk) and # (pound sign). Pressing one of the keys causes an electronic circuit in the key pad to generate two output tones that represents the number. The digit must persist for at least 40 ms and there must be a pause of at least 40 ms between each pulse. The two frequencies are chosen from a set of 4 low frequencies (697 to 941 Hz) and 4 high frequencies (1.209 to 1.633 KHz). Table-10.1 shows the DTMF matrix.

f_b/f_a	1209 Hz	1336 Hz	1477 Hz	1633 Hz
697 Hz	1	2	3	A
770 Hz	4	5	6	B
852 Hz	7	8	9	C
941 Hz	*	0	#	D

Table-10.1: DTMF Matrix

The two frequencies are transformed to a DTMF signal using the following equation

$$f(t) = A_a \sin(2\pi f_a t) + A_b \sin(2\pi f_b t)$$

Where the ratio between the two amplitudes should be:

$$A_b/A_a = k \qquad 0.7 < K < 0.9$$

DTMF using IC: Dialing may be accomplished by sending dual tones onto the line in which a low-frequency and high-frequency sine wave oscillator feed the speech circuit. The IC DTMF generator has a counter and decoder that counts pulses from a crystal-controlled oscillator and provides output codes that corresponds to the low and high frequency tones required. Each of the two outputs from the counter feed into its own D/A converter. Both tones are summed in an operational amplifier and are fed to the speech as combined signal by the output stage. DTMF tone detection is done by DSP, A/D conversion etc.

4. Ringing / Electronic Ringer circuit / RSM

Ringing is the way the called party is signaled about arrival of a call. When the call arrives at the local central office and if the called telephone is on-hook, the **ringing voltage** is sent over the local loop to ring the called telephone. At the same time, the central office serving the called telephone sends back a **ring back signal** back to calling telephone to indicate that the called telephone is ringing. The AC ringing signal at central office is generated by a dc motor driving an ac generator or by a solid inverter (called the ringing machine and powered by -48V dc central office supply). This is done completely by a relay that is energized by the switch. The ringing signal sent to a customer's telephone is 90 Volts AC at a frequency of 20 Hz in North America. In Europe it is around 60-90 Volts AC at a frequency of 25 Hz.

The ringer circuit (at subscriber apparatus) is supplied from the line ahead of the switch hook so that the circuit can be energized by the ringer signal even though the telephone handset is on-hook. Conventionally ringing is done by electromagnetic bell. But, presently available single tone electronic ringer has a fixed frequency self –resonant oscillator which is turned on and off cycles of the AC ringing voltage. Multitone ringers are necessarily more complex electronically than single tone types. The output signal of a multi-tone ringer is produced by switching between two or more frequency at a rate determined by the tone ringer circuitry whereas the AC ringing voltage frequency determines the switching rate of the single tone ringer.

5. Switching mechanism

Switching systems are assemblies of equipment that setup, maintain, and disconnect connections between multiple communication lines. Switching systems are often classified by the type of network they are part of (e.g., packet or circuit switched) and the methods that are used to control the switches. . In PSTN system circuit switching is used. The term “switch” is sometimes used as a short name for switching system. Public telephone switching systems have many switches within their network.

Modern switches use computer systems to dynamically setup, maintain, and disconnect communication paths through one or more switches. True computer-based switching came about through the introduction of the electronic switching systems (ESS’s). ESS COs did not require a physical connection between incoming and outgoing circuits. Paths between the circuits consisted of temporary memory locations that allowed for the temporary storage of traffic. For an ESS system, a computer controls the assignment, storage, and retrieval of memory locations so that a portion of an incoming line (time slot) could be stored in temporary memory and retrieved for insertion to an outgoing line. This is called a time slot interchange (TSI) memory matrix. The switch control system maps specific time slots on an incoming communication line (e.g., E1) to specific time slots on an outgoing communication line.

The public telephone network switching system architecture uses a distributed switching system that has a hierarchy of switching levels. Distributed switching systems connect calls through the nearest switching system. With distributed network architecture, the call processing requirements are distributed to multiple points. Using a multilevel hierarchy structure for switching systems allows switching to occur at lower levels of switching unless the telephone call must pass between multiple switches. At that point, the call is passed up to a higher-level switch for transfer to more distant locations. Each subscriber is connected to ‘CO’ through local loop (subscriber loop).The line side interface consists of two wires T (Tip) and R (Ring) which refers to tip & ring parts (formally known as plugs) of manual switch boards.

6. Call set-up procedure

When the handset of the telephone is resting in its cradle (ON HOOK state), the speech circuit between telephone handset and the central office is open but ringer circuit is connected. When the handset is removed from the cradle, the spring loaded buttons comes up. The switch hook closes (OFF HOOK state). This completes the circuits to the exchange and current flows in the circuit. The OFF HOOK signal tells the exchange that someone wants to make a call. The exchange returns a dial tone to the calling to let the caller know that the exchange is ready to accept a telephone number. Call processing involves several elementary processing sequences in the exchange, each lasting a few tens or hundreds of milliseconds.

7. Interfacing circuits between subscriber line and switching digital circuitry

Each subscriber is provided with a hardware interface at the switching center. It is also known as the line circuit/SLIC (subscriber line interface circuit). The functions of the SLIC are often summarized by the acronym ‘BORSCHT’ (**B**attery feed, **O**ver-voltage, **R**inging, **S**upervision, **C**odec and **H**ybrid and **T**esting)

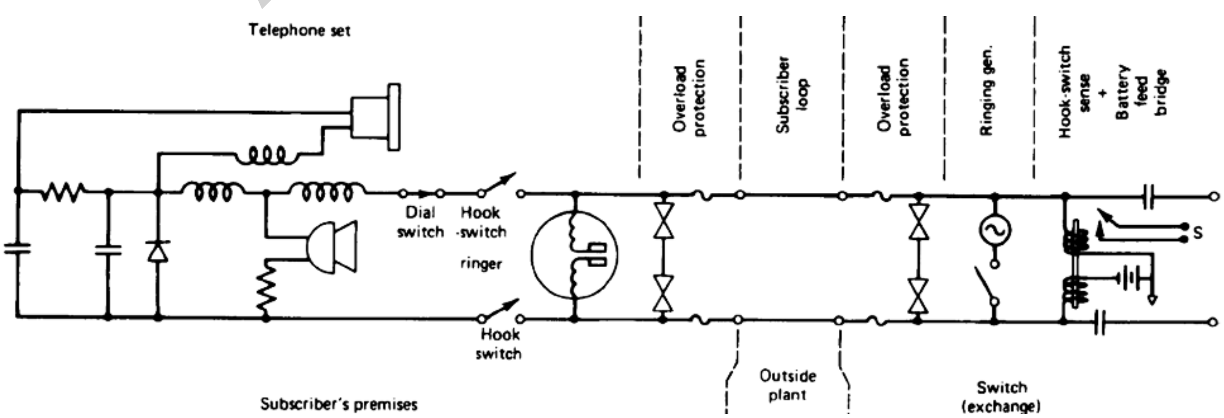


Fig 10.4:- The conventional telephone subset (Subscriber and subscriber loop)

8. **PABX (Private automatic branch exchange) system**

The use of computers to control the switching functions of a central office led to the designation 'electronic switching system (ESS). The most common switching function involves direct connections between subscriber loops at an end office or between station loops at a PBX (private branch exchange). These connections inherently require setting up a path through switch from originating loop to a specific terminating loop. Call distributions are often implemented by same basic equipments as PBXs. In the present PABX, 4 subscribers are connected.

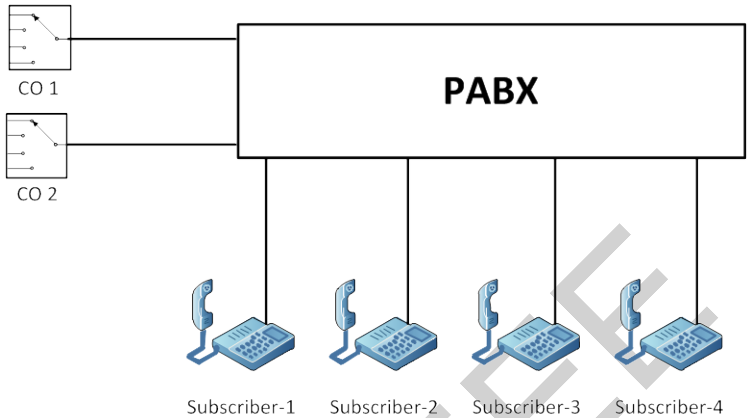


Fig 10.5:- Subscribers connected to PABX trainer

9. **Telecommunication trainer board function**

Dialed Number	Function
0	Extension programmed / denied
9	Access to reserved
24-29	Direct access to trunk line (27-1st trunk, 28-2nd trunk)
30-33	Extension to extension
13/14	Automatic call back on busy trunk line/ don't disturb
12/0	Hot line/ Cancel all present features

(d) Experimental procedure

(I) Speech Circuit (keep switch fault in OFF position)

1. Connect power supply in proper polarity to the kits EPX-2005 and switch it on (as in Fig:-10.6).
2. Connect four telephone instruments to extension 1 to extension 4 respectively.
3. Pick up the hand set of extension 30 and dial 31 (ringing will be heard)
4. Pick up the hand set at extension 31 (loop between the two extensions is completed and speech path is established)
5. Do the same to establish connection between CO line and extension also.
6. Observe the following wave shapes –
   i. CRO at AEX1 for ext. 30 and at AEX2 for ext. 31,
   ii. Ring signal at AEX2 and ring back signal at AEX1 (While the phone is ringing)
   iii. After connection is established observe voice signal

(II) DTMF and Pulse Dialing (keep switch fault in OFF position)

1. Connect power supply in proper polarity to the kits EPX-2005 and switch it on.
2. Keep the switch SW1 for DTMF dialing as on Fig:-10.7(b) and also instrument on DTMF (Tone) position.
3. Pick up the handset from any extension and dial number of extension. IC 8870 is used as integrated DTMF receiver and observe frequency of the number at TR1 which is tone
4. For pulse dialing, keep the switch SW1 (as on Fig:-10.7(c)) and instrument must be on pulse dialing position and observe dial pulses on busy test points of CO1 and CO2 respectively.
5. Access CO1 or CO2 from any extension, observe the switching of relays by dialing outgoing numbers through CO1 & CO2 and hear the pulses of dialed number.

(III) Study of DTMF generator (keep switch fault in OFF position)

1. Connect power supply in proper polarity to the kits EPX-2005 and switch it on (As in Fig 10.8).
2. Pick up the hand set of extension 30 and dial 31. Observe ring signal. Transformer is used to boost up the signal for ring voltage, which is required at ringer circuitry.
3. Observe ring back signal at corresponding test point.
4. Two types of rings can be heard from the telephone instrument connected to the system. In case of internal instrument call- the ring is continuous one with a one sec ON and two sec. OFF. A ring from a CO junction line will ring like a normal telephone.
5. Also, we can observe the ring back signal at test point RBTN.

(IV) Switching mechanism etc. (keep switch fault in OFF position)

1. Connect power supply in proper polarity to the kits EPX-2005 and switch it on (As in Fig:-10.9)
2. Connect subscriber line (DOT) at CO1 or CO2.
3. Incoming call always come through central office. It may ring on any specific extension, like it is factory set extension 30.
4. We can observe LED indication during incoming call. We can observe TTL voltage at test point RING (CO1) & RING (CO2) respectively.
5. We can take these calls by pressing 8 on any extension. Flow of signal is shown in block diagram.
6. For outgoing calls we can connect to central office by pressing ‘0’ or we can select CO1 by dialing 27 & CO2 by 28 through any extension. Then dial outgoing number when dial tone of exchange is heard.
7. These subscriber lines and extension lines are connected to mechanical switching devices. The ring ON signal for digital circuitry (latch) can be seen at RING test points of respective extensions (1-4). Observe CO1/ CO2 as indicated by L1/L2 and TTL signal at test point BUSY for CO1 and CO2. [Study of interface circuit]

(V) Study of Tone generation (keep switch fault in OFF position)

1. **Dial tone:** Lift handset of ext. 30 and hear the dial tone is 8 sec. long continuous sound during which exchange waits for dialing to be initiated and can be observed at DLTN in tone generation section. Otherwise busy tone is issued from exchange.
2. **P & T Dial tone:** On accessing a direct line by dialing ‘0’, normal P & T dial tone is obtained (DOT line must be connected to at CO & CO2). This signal can be observed at TR1.
3. **Busy tone:** Lift handset of ext. 31 first and then also of ext. 30 and dial no. 31 (hear busy tone- a discontinuous sound Du-Du). There are two types of busy tone – High speed busy tone (equal duration of ON/OFF), Call number busy tone (double duration on and single duration off).
4. **Internal Ring back tone:** Lift handset of ext. 30, dial 31, 32 and ring back tone is observed at RBTN for ext. 30. This is a discontinuous sound of two frequencies- one sec. on two sec. off. This tone is heard until called subscriber answers.
5. **Ringing tone:** Two types of rings can be heard from the telephone instrument connected to the system. For internal call- one sec. on two sec. off continuous ringing tone. For external calls- normal telephone like tone.

(d)Report

1. Comment on various parts of a complete PABX system.
2. Show and explain the results.
3. Anything else instructed by the teacher.
4. Discussion expressing originality of the student.

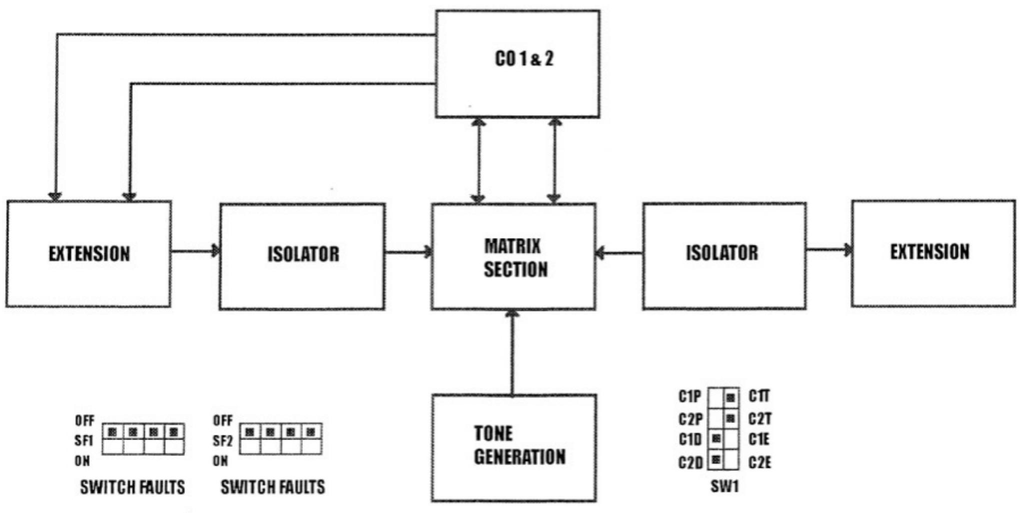


Fig 10.6:-Block diagram for study of Speech Circuit

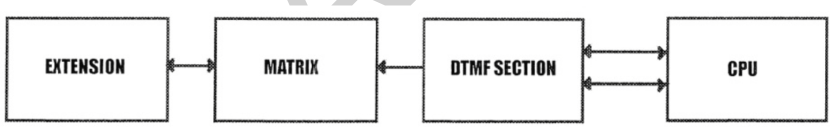


Fig 10.7(a):-Block diagram for DTMF Section

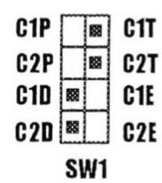


Fig 10.7(b):-Switch setting for DTMF Dialing

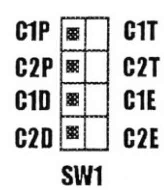


Fig 10.7(c):-Switch setting for Pulse Dialing

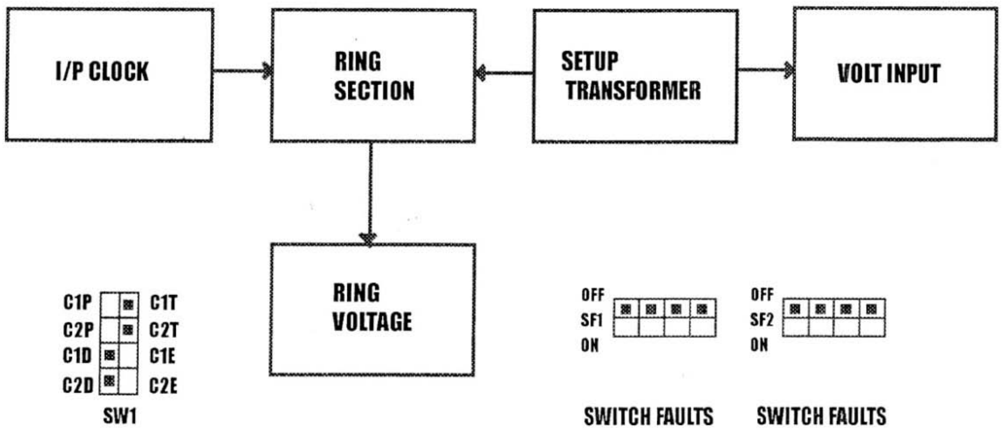


Fig 10.8:- Block diagram for dual tone ring generator

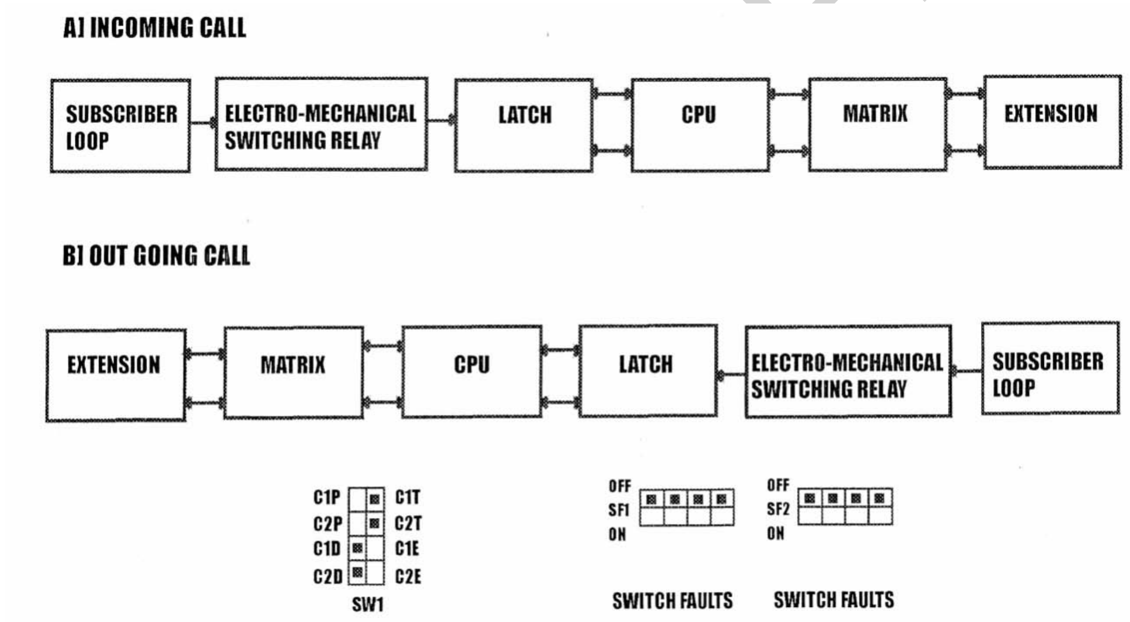


Fig 10.9:- Block Diagram for switching mechanism between Subscriber

Reference & Acknowledgement(s)

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Special Thanks to:

1. Mizanur Rahman
   Lab attendant.

Experiment no	Reference
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2.	• Analog Communication System, Application and Measurement ETEK Technology Co. LTD • Communication Systems , Simon Haykin
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