INSTITUTE OF TECHNOLOGY
& MANAGEMENT
GWALIOR • MP • INDIA

Laboratory Manual

ADC
(IT-404)

For

Second Year Students
Department: Information Technology

Department of Information Technology

Vision of IT Department:

The Department of Information Technology envisions preparing technically competent problem solvers, researchers, innovators, entrepreneurs, and skilled IT professionals for the development of rural and backward areas of the country for the modern computing challenges.

Mission of the IT Department:

- To offer valuable education through an effective pedagogical teaching-learning process.
- To shape technologically strong students for industry, research & higher studies.
- To stimulate the young brain entrenched with ethical values and professional behaviors for the progress of society.

Program Educational Objectives:

Graduates will be able to

- Our graduates will show management skills and teamwork to attain employers' objectives in their careers.
- Our graduates will explore the opportunities to succeed in research and/or higher studies.
- Our graduates will apply technical knowledge of Information Technology for innovation and entrepreneurship.
- Our graduates will evolve ethical and professional practices for the betterment of society.

Program Outcome(POs):

Engineering Graduates will be able to:

1. **Engineering knowledge**: Apply the knowledge of mathematics, science, engineering Fundamentals, and an engineering specialization to the solution of complex engineering problems.
2. **Problem analysis**: Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.
3. **Design/development of solutions**: Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.
4. **Conduct investigations of complex problems**: Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data,and synthesis of the information to provide valid conclusions.
5. **Modern tool usage**: Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modeling to complex engineering activities with an understanding of the limitations.
6. **The engineer and society**: Apply reasoning informed by the contextual knowledge toassess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
7. **Environment and sustainability**: Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.
8. **Ethics**: Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
9. **Individual and team work**: Function effectively as an individual, and as a member orleader in diverse teams, and in multidisciplinary settings.
10. **Communication**: Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, andgive and receive clear instructions.
11. **Project management and finance**: Demonstrate knowledge and understanding of the engineering and management principles and apply these to one's own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.
12. **Life-long learning**: Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

Course Outcomes

Data Structure (IT-404)

CO1:	To understand the classification of signals & systems and apply the basics of signals and systems to study the Linear Time Invariant system behavior
CO2 :	To analyze various continuous wave and angle modulation and demodulation techniques in the presence of noise
CO3 :	To remember modulation and demodulation techniques for digital signal and determine bandwidth requirement and sampling theorem for low pass and Band pass signal,
CO4 :	Design and Comparison of various systems (Differential PCM (DPCM), Delta Modulation (DM) and Adaptive Delta Modulation (ADM),)
CO5 :	Evaluate digital modulation techniques (ASK, FSK, PSK & QAM) with the help of MATLAB tool.

Course	P0 1	P0 2	P0 3	P0 4	P0 5	P0 6	P0 7	P0 8	P0 9	P01 0	P0 11	P0 12	PSO 1	PSO 2	PSO 3
CO1	3	3	2	0	0	0	0	0	0	0	0	0	0	0	0
CO2	2	3	2	0	0	0	0	0	0	0	0	0	0	0	0
CO3	3	2	2	0	0	0	0	0	1	1	0	0	0	0	1
CO4	2	2	0	0	0	0	0	0	1	1	0	0	2	0	1
CO5	0	0	0	2	1	0	0	0	0	0	0	0	2	0	0
Avg	**2.5**	**2.5**	**2**	**2**	**1**	**0**	**0**	**0**	**1**	**1**	**0**	**0**	**2**	**0**	**1**

List of Program

1. OBJECT

(1) To study the DSB-SC AM Generation through Balanced Modulator.

(2) To observe the DSB-SC AM signals on Oscilloscope and measure its depth of modulation.

APPARATUS REQUIRED: -

DBS-SC AM Generation kit, Oscilloscope, Patch chords and BNC Chord

THEORY: -

Consider the diagrams below

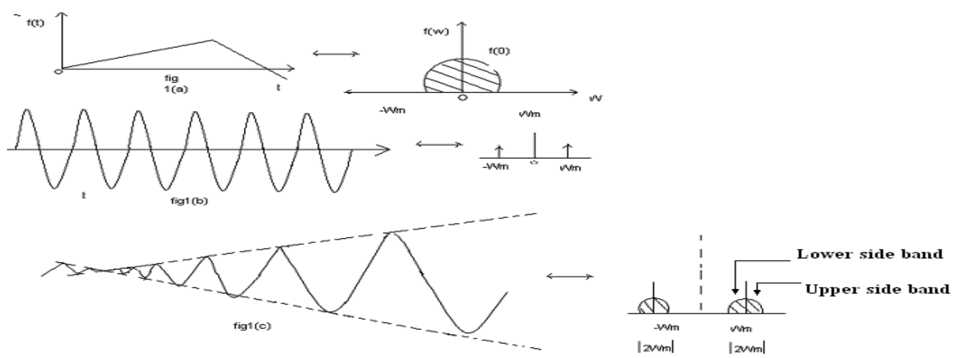


Fig2.1 (c) is a result of the multiplication (t) cosωct $\Leftrightarrow$1/2[F(ω+ωc)+F(ω−ωc)]

The function f(t) cosωct is Amplitude Modulated .The above signal shows phase reversal at the zero crossing of envelop. It also observed that impulses at $\pm\omega_c$ are missing which means

(a) The carrier term ω_c is suppressed in the spectrum.

(b) The base band is present twice in modulated spectrum.The term $\pm$ (ω_c+ω_m) is upper side band and the term $\pm$ (ω_c−ω_m) is lower side band .Hence it is known as Double Side Band- Suppressed Carries Amplitude Modulation system.

BALANCED MODULATORS: -

In a balanced modulator two non-linear devices are connected in the balanced mode so as to suppress the carrier wave. Below is the diagram of balanced modulator using two devices as non-linear elements.

$e_1 = \cos\omega_c t + f(t)$

$e_2 = \cos\omega_c t - f(t)$

For non- linear circuits current is given by

$I = ae + be^2$

Hence

$$I_1 = ae_1 + be_1^2$$
$$= a\,[\cos\omega_c t + f(t)] + b[\cos\omega_c t + f(t)]^2 \ldots\ldots\ldots\ldots\ldots\ldots\ldots(1)$$

and

$$I_2 = ae_2 + be_2^2$$
$$= a[\cos\omega_c t - f(t)] + b[\cos\omega_c t - f(t)]^2 \ldots\ldots\ldots\ldots\ldots\ldots\ldots(2)$$

Voltage at the input of band pass filter is given by

$V_0 = V_1 - V_2 = i_1R - i_2R$

$V_0 = 2R[af(t) + 2bf(t)\cos\omega_c t]$

The output of band pass filter centered around $\pm\omega_c$ is given by

Output $= 2bf(t)\cos\omega_c t$

$= Kbf(t)\cos\omega_c t \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad (3)$

Equation (3) given DSB-SC signal.

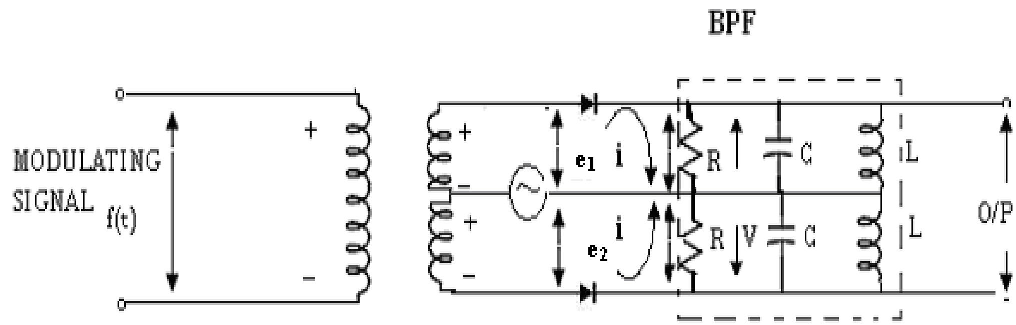


Fig.2.2 Circuit Diagram of BALANCED MODULATOR

PROCEDURE:-

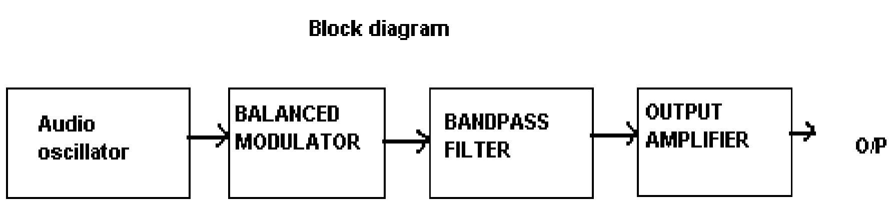


Fig2.3 DSB SC AM TRANSMITTER

- 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 gain preset in fully clockwise position.

(d) SPEAKER switch in OFF position.

- Turn the Audio Oscillator blocks Amplitude Preset to its fully clockwise position, and examine the block output on an oscilloscope. This is audio frequency, which will be our modulating signal. Its frequency and amplitude can change by oscillator frequency preset and oscillator amplitude respectively.

- Turn the BALANCE preset, in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, to its fully clockwise position. It is this block that we will used to perform Double-Side Band Amplitude Modulation.
- Monitor, in turn, the two inputs to the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, at t.p. 1 and t.p. 9. Note that:

(e) The signal at t.p. 1 is the audio-frequency sine waveform the AUDIO OSCILLATOR block. This is the modulating input to our double-sideband modulator.

(f) Test point 9 carries a sine wave of frequency 1 MHZ and amplitude 12mV pk/pk approx. This is the carrier input to our double-sideband modulator.

- The out put from the BALANCED Modulator & BALANCED FILTER circuit block (at tp3) is a double side band Am waveform, which has been formed by amplitude modulating the 1 MHz carries sine wave with the Audio-frequency sine wave from Audio oscillator.

- To determine the depth of modulation, measure the maximum amplitude (Vmax) and the minimum amplitude (Vmin) of the AM waveform at t.p. 3, and use the following

Formula:

$$\textbf{Percentage Modulation} = \frac{V\,max - V\,min}{V\,max + V\,min}$$

Where V_{MAX} and V_{MIN} are the maximum and minimum amplitudes shown in

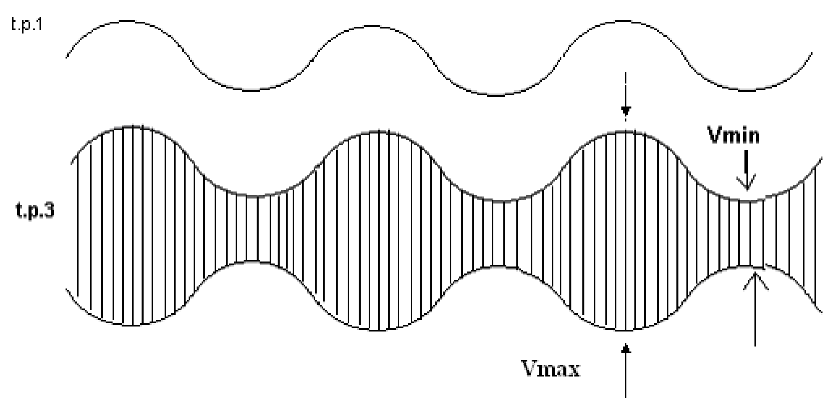


Fig 2. 4

- Now vary the amplitude and frequency of the frequency sine wave, by adjusting the AMPLITUDE and FREQUENCY present in the AUDIO OSCILLATOR block.

Note the effect that varying each preset has on the amplitude modulated waveform.

The amplitude and frequency amplitudes of the two sideband can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. D this by turning the Amplitude preset to its MIN position, and note that the signal at tp3 becomes an unmodulated sine wave of frequency 1 MHz indicating that only the carrier component now remains. Return the AMPLITUDE preset to its maximum position.

Now turn the BALANCE preset in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, until the signal at tp3 is as shows in fig 5

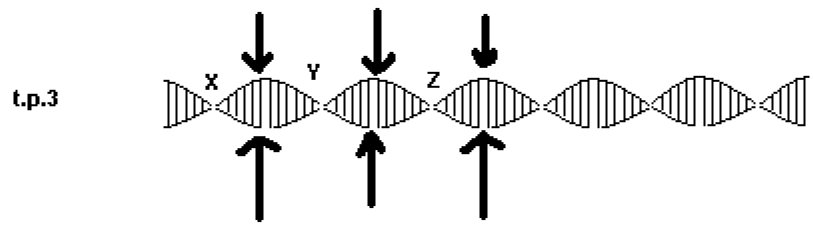


Fig 2.5

The Balanced preset varies the amount of the 1MHz carries component, which is Passed from the modulator's out put

By adjusting the preset until the peaks of the wave form (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 to leave only the two sidebands

Note that once the carrier has been balanced out the amplitude of tp3's waveform should be zero at minimum points XYZ etc. If this is not the case it is because one of the side band is being amplified more than the other. To remove this problem the band pass filter in the BALANCED MODULATOR & BAND PASS FILTER CIRCUIT 1 block must be adjusted so that it passes both side band equally. This is achieved by carefully trimming transformer T1, until the waveform amplitude is as closed to zero as possible at the minimum points.

The waveform at tp3 is known as a Double Side Suppressed Carrier (DSB SC) waveform, and its frequency spectrum is as shown in

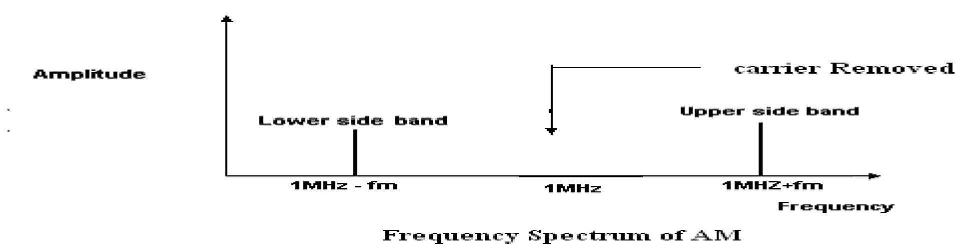


Fig 2.6

Note that only two side band remain the carrier component has been removed.

2. OBJECT: -

Observe the Demodulation of AM Signal.

APPARATUS REQUIRED: -CRO, Trainer kit, Patch Chords

THEORY:-

AM modulated waveform is in the form.

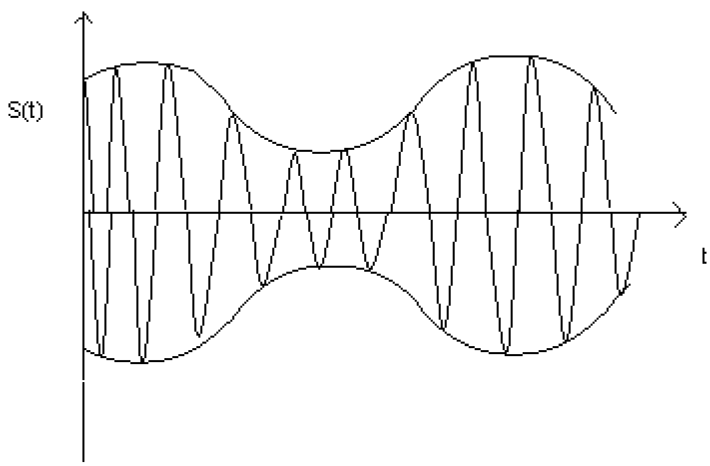


Fig1.1 Amplitude modulated waveform

This wave is transmitted from transmitter and received by antenna at receiver. To obtain original signal from the received signal we give this AM- signal as an input to diode detector. Diode detector follow the envelop of the received signal.

The original resides in the envelop hence we get the original signal at the out put of the detector. Which can be verified with the help of CRO. The process of recovering original from modulated signal is called demodulation.

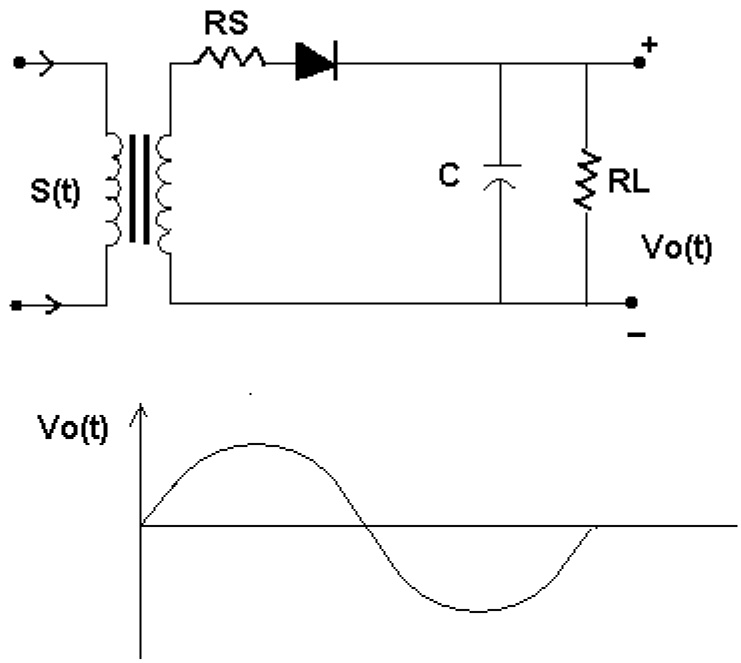


Fig1.2 AM Demodulator

Procedure: -

1. Generate AM signal in AM Transmitter Trainer.
2. Connect the output of AM Transmitter to the input of AM Receiver.
3. Observe the Demodulated wave on $t_{p\ 40,39,38}$ in AM receiver Trainer.
4. Observe the frequency of the demodulated signal be same as that of the original signal.

3. OBJECT: -

1. **To study frequency modulation (FM) and trace its waveforms.**
2. **To find the frequency of carrier.**

APPARATUS REQUIRED: -

FM kit, Signal Generator, Oscilloscope, and Potentiometer.

THEORY: -

In the communication information like audio signal or speech writing signal is to be transmitted from one point to another. But if it is transmitted directly through the antenna then transmitting antenna must be ¼ to ½ wavelength long. To reduce the antenna size, there low frequency information (audio) signal is mixed with higher frequency. Such process is called as frequency modulation (FM). This higher frequency is called as carrier signal. At the receiver side, this signal is converted back to audio frequency called as demodulation process.

Frequency Modulation: -

In this modulating signal em is used to vary the carrier signal frequency. Let the change in the carrier frequency be k_{em} where k is a constant know as frequency deviation constant, then the instantaneous carrier frequency is

$$f_i = f_c + k_{em}$$

Where fc is the unmodulated carrier frequency.

Let modulating signal is sine wave given by

$$e_m = \text{emmax } \sin\omega_{mt}$$

Then instantaneous carrier frequency becomes

$$f_i = f_c + k_{emmax} \sin\omega_{mt}$$

The peak frequency deviation of the signal is defined to be

$$\Delta_f = ke_{mmax}$$

Therefore

$$f_i = f_c + \Delta f \sin\omega_{mt}.$$

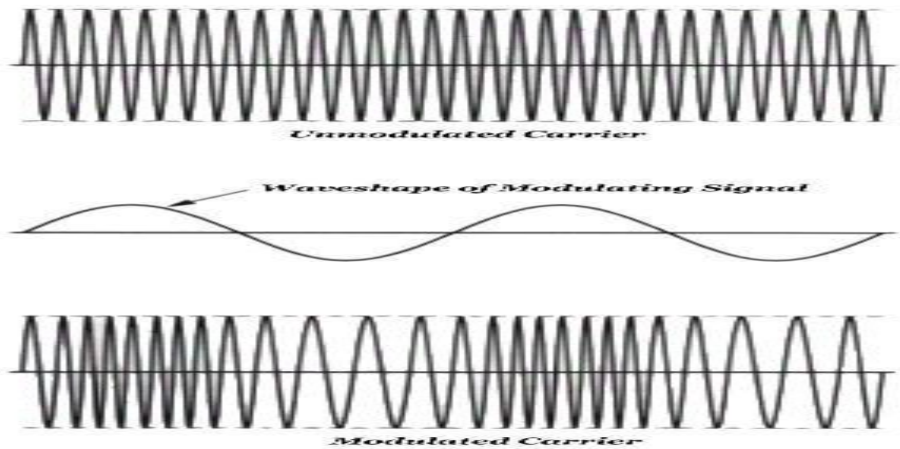


Fig. 3.1 Frequency Modulated Waveform

Fig1.1shows the FM waveform if modulating signal increase in positive direction towards positive peak, then carrier frequency at o/p (FM) increases it become max at positive peak. If modulating signal decreases from positive peak towards negative half cycle it decreases the carrier frequency at o/p. when modulating signal returns to zero, the carrier frequency returns to it's center frequency. When modulating signal is at it's negative peak, the carrier frequency become zero at o/p. therefore the modulating signal produces a frequency- modulated waveform as shows in fig 1. The carrier changes equally above and below it's center frequency. The amount of frequency change is called as the frequency deviation. The rate of frequency deviation is determined by the frequency of the modulating signal but not be the amount of deviation e.g. if carrier changes continuously from 90 MHZ to 90.2 MHZ then frequency deviation is +0.2 MHZ or 200 kHz. If the 500hz audio tone is used to modulate the above carrier then it will changes equally above & below it's center frequency 500 times per second. As the amplitude of modulating signal deviation will change for fixed value of modulating frequency hence the amount of carrier deviation is directly proportional to the amplitude of the modulating signal.

FM o/p wave consists of constant amplitude but changes in frequency.

Modulation Index: -

It is defined as the ratio of maximum frequency deviation of FM from carrier frequency (f_c) f_d to the modulating frequency (fm) and is given by

$$M = \frac{f_d}{f_m}$$

In FM it's value can reach to a high value as compared to AM.

Deviation ratio: - It is the ratio of max frequency deviation to max modulating frequency.

$$\text{Deviation Ratio} = \frac{f_{dmax}}{f_{m\,max}}$$

If fm max is 18KHZ & fdmax = 90KHZ

$$\text{Then deviation ratio} = \frac{90KHZ}{18KHZ} = 5$$

Band with: -

BW = 2fm * highest order side band

[for derived modulation index from chart]

$$\text{Where modulation index } m = \frac{f_d}{f_m}$$

$$\text{e.g. If } m = \frac{10KHZ}{5KHZ} = 2$$

From table for modulation index 2, highest order side band is 5^{th}

Therefore, the bandwidth is

BW = 2fm * highest order side band

= 2 * 5KHZ * 5

BW = 50KHZ

PROCEDURE: -

1. Study the circuit diagram provided on the front panel of the kit.
2. Connect sine wave input of 100Hz, 10Vp-p from signal generator to the point marked “AF i/p”.
3. Connect CRO at pin 7 of IC * R 2206. Observe the i/p waveform change by potentiometer R_1.
4. Now connect your oscilloscope at FM o/p pin 2, observed the waveform for zero amplitude of modulating signal. This o/p is called as carrier.
5. Now change R, the carrier o/p frequency will change producing a waveform shown below, called frequency-modulated waveform.
6. Change position of R_1 amplitude of modulating signal will change which will change the amount of frequency deviation note it’s minimum and max frequency deviation.
7. Keep pot R_1 at mid position change freq of modulating signal, note & observed the change in frequency deviation it should remains constant.

OBSERVATION: -

- Frequency of modulating signal FM
- Amplitude of modulating signal AM
- Frequency of carrier signal Fc
- Amplitude of carrier signal Ac

Voltage / division	Time/ division	Total	Division	1Freq.=___________ Time period
		Amplitude	Time period	

CONCLUSION: -

Thus by changing the amplitude of the modulating signal. The amount of deviation will change. But it remains constant for any change in modulating signal frequency. Any slight change in division for change in modulating frequency were due to generator output amplitude changes.

4. OBJECT:- -To study Frequency Demodulation using Foster Seeley Detector

APPARATUS REQUIRED: - FM kit, Patch cords, Oscilloscope BNC cord

THEORY: -

It is a widely used discriminator. The circuit dig in fig 1.1 is a double tuned circuit in which both primarily and secondary are tuned to same frequency intermediate frequency. The center of the secondary is connected to top of the primary through capacitor c3
The capacitor c3 performs following functions: -
1. It blocks the dc from primary to secondary

2. It couples the signal frequency from primary to center topping of the secondary. The primary voltage V_{fm} (signal voltage) thus appears across l, except a small drop across c_3

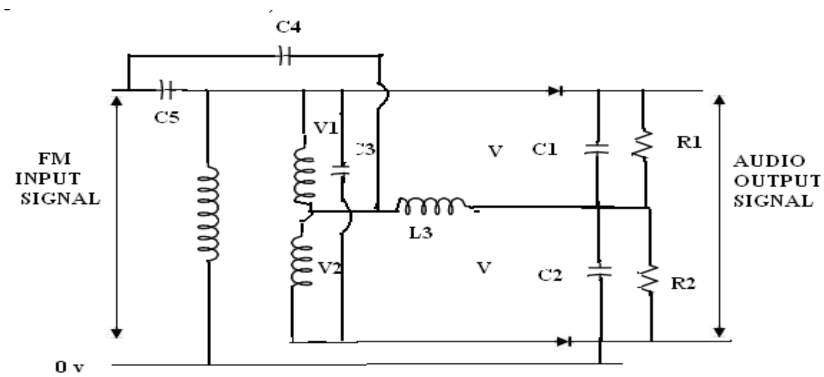


Fig 4.1 FM Demodulator

The center tapping of the secondary has an equal and opposite voltage across each half winding. There fore $V_1$and V_2 are equal in magnitude but opposite in phase

$$V_{a1}=Vbm+V_1$$

$$V_{a2}= V_3-V_2$$

Procedure:

(1) Ensure that the following condition exist on the board.

1. Audio input select switch in INT position.
2. Mode switch in DSB position.
3. Out put amplifiers gain preset in fully clockwise position.
4. Speakers switch in off position.

(2) Turn the AUDIO OSCILLATOR blocks out put (t. p. 14) On an oscilloscope. This is the audio frequency sine wave, which will be as our modulation signal. Note that the sine wave's frequency can be adjusted from about 300Hz to approximately 3.4 KHz, by adjusting the Audio Oscillator's Frequency preset. Return the amplitude present to its MAX position.

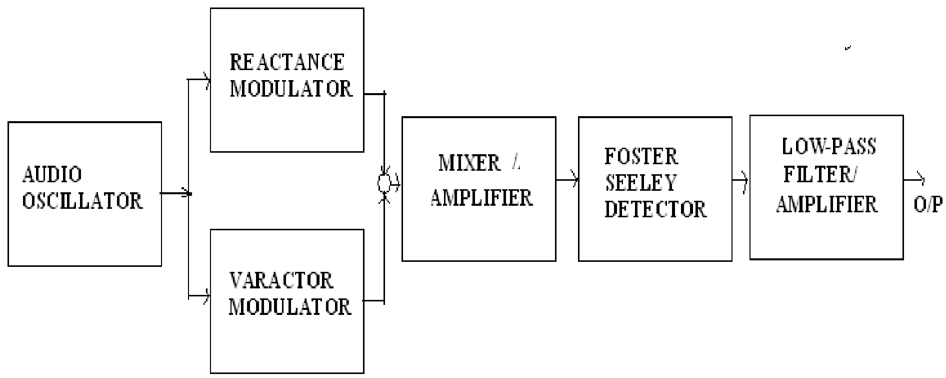


Fig. 4.2 BLOCK DIAGRAM OF FOSTER SEELEY DETECTOR

3) Turn the BALANCE preset, in the BALANCED MODULATIOR 7 BANDPASS FILTER CIRCUIT 1 block, to its fully clockwise position. It is this block that we used to perform double-side band amplitude modulation. Select the foster- seeley detector by putting the switch in the FOSTER- SEELEY position.

4) Initially, we will use the VARACTOR MODULATIOR to generate our FM signal, since this is the more linear of the two modulators, as far as its frequency / voltage characteristic is concerned. To select the VARACTOR MODULATOR put the REACTANCE / VARACTOR switch in the VARACTOR position.

5) The Audio Oscillator's out put signal (which appears at t.p. 1) is now being used by the VARACTOR MODULATIOR, to frequency – modulate a 445 KHz – carrier sine wave. As we saw earlier, this FM waveform appears at the FM OUTPUT socket from the MIXER / AMPLIFIER block

6) Now monitor the audio input signal to the VARACTOR MODULATOR block (at t.p. 14) together with the FOSTER-SEELEY OUTPUT (at t. p. 52) triggering the oscilloscope on t.p. 14.

The signal at t.p. 52 should contain two components:

A sine wave has the same frequency as the audio signal at t.p. 14'

7) The LOW- PASS FILTER/ AMPLIFIER strongly attenuates this high frequency ripple component and also blocks any small D.C. offset voltage that might exist at the detector's out put. Consequently the signal at the output of the LOW – PASS FILTER/ AMPLIFIER block (at t. p. 73) should very closely resemble the original audio modulating signal

8) Monitor the audio input to the VARACTOR MODULATIOR (at t.p. 73) and adjust the GAIN preset (in the LOW PASS FILTER/ AMPLIFIER block's) until the amplitudes of the monitored audio waveforms are the same.

9) Adjust the AUDIO OSCILLATOR block's AMPLITUDE And FREQUENCY presets, and compares the original audio signal with the final demodulated signal.

5. Object: -

Study of the FM Demodulation using Ratio Detector

Apparatus Required:-

Trainer kit, CRO, Patch Chords, BNC Chords

THEORY:-

Diode D_2 has been reserved so that the polarity of the voltage across C_2 will be as shown in the diagram .When the carrier is UN modulated, the voltage across c_1 and c_2 are equal and additive The audio output is taken across c_2 or R_2 capacitor C_6 is a large electrolyte capacitor. It charges to this voltage. Owing to the long time constant of C_6, the total voltage across R_1 & R_2 remains virtually constant at all times. In fact it just acts as a power supply or a battery. The important thing to note is that it keeps the total voltage of C_1+C_2 at a constant value.

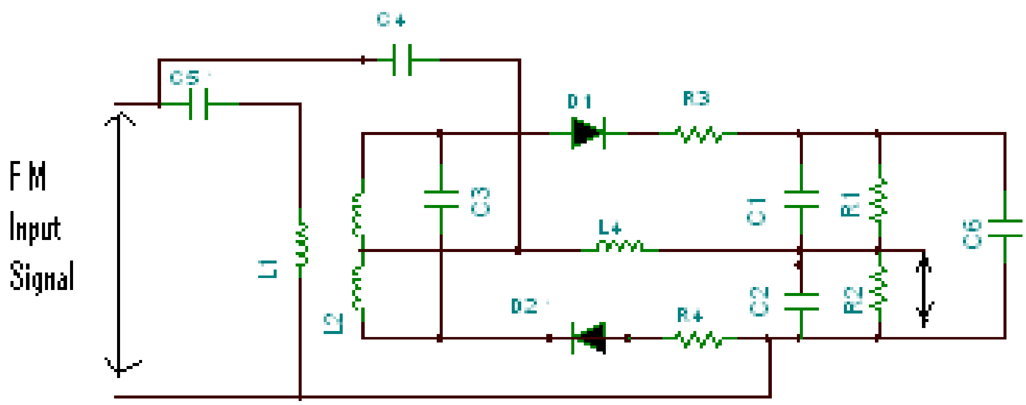


FIG .5.1 CIRCUIT DIAGRAM OF RATIO DETECTOR

As shown in the phasor diagram an un modulated F M signal will result in equal voltage across R_1 and R_2. The voltage across R_2 is the output from the circuit. If frequency of the F M signal increases, the voltage across R_1 will increase and that across R_2 will decrease. Conversely, if the frequency of the F M signal decreases, the voltage across R_1 will decrease and that across R_2 will increase. The final demodulated audio output voltage is taken across R_2 and this voltage changes continuously to follow the frequency variation of the incoming F M signal. Since the sum of the voltages across R_1 and R_2 remains constant. The ratio of the voltages across R_2 to this total voltage change with the F M signal's frequency. It is this changing voltage ratio that gives the Ratio Detector its name.

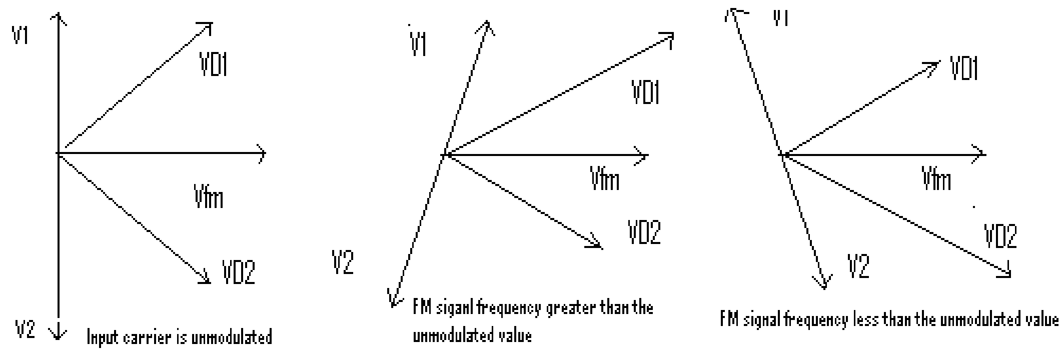


FIG5.2 PHASOR DIAGRAM

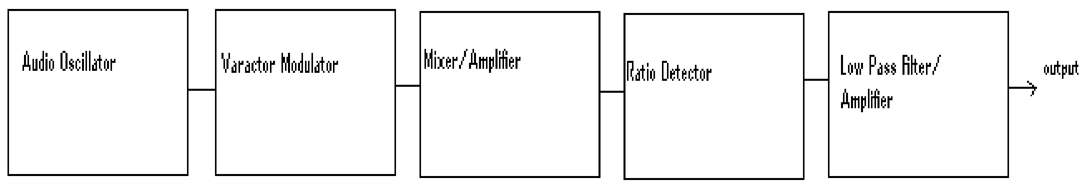


FIG5.3 BLOCK DIAGRAM OF RATIO DETECTOR

Procedure:-

1. Make the connection as shown in the Block diagram.

2. Observe the output at of Varactor Modulator and Ratio Detector block at t.p.14 &53 respectively .At t.p.53 there is

- A positive d.c. offset voltage
- A sinewave of same frequency but 180º phase shift from that of at t.p.14

3. The Low Pass Filter block removes the d.c. offset voltage at the detector output and strongly attenuate any residual high frequency ripple that may be present.Observe the difference in signal at output and input at t.p. 73.

Moniter audio input to the Varactor Modulator (t.p.14) and the output of the Low Pass Filter (t.p.73) and adjust GAIN preset until the amplitude of the monitered audio waveforms are the same

4. Adjust the Audio Oscillator block's Amplitude and Frequency presets, and compare the original audio signal with the final demodulated signal.

6. OBJECT: - Time division multiplexing (TDM) and De multiplexing

APPARATUS: -

CRO, Patch chords, Trainer kit

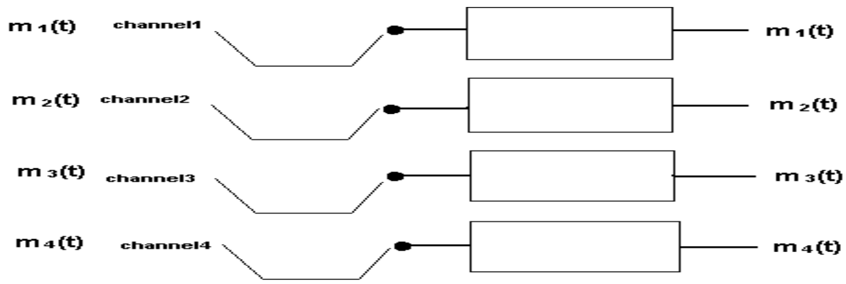
THEORY: -

When there is only signal channel is available for transmission but we have more then one signal available at transmitter. Then we need to multiplex the signal in time domain. Multiplexing is done as fallows.

We take sample of first signal and transmitted it then we take the sample of second signal and transmit the second signal these signals are separated in time therefore there is no overlapping takes places. By using same approach we can send number of signal over a signal communication channel.

At the transmitting end on the left a number of band-limited signal are connected to the correct point of a rotary switch. We assumed that the signals are similarly band limited. For e.g.. They may all be voice signals, limited to 3.3 KHz. As the rotary arm at receiving end is in sync with the switch at the sending end. The two switches make connect simultaneously at similarly numbered contacts. With each revolution of the switch, one sample is taken to each input signal and presenting to the correspondingly numbered contact of the receiving end switch. The train of samples at, say, terminal 1 in the receiver, pass through low pass filter1, and at the filter output the original signals m1 (t) appear reconstructed. Of course, if Fm is the highest frequency spectrum component present in any of the input signals the switches must at least 2 Fm revolutions per second.

In this switching process, the switch at left side that samples the signals is called communicator and switch at the right side is called decommutator.



Procedure: -

1. Connect 250 Hz signal to CH0, 500Hz signals to CH1, 1Khz signal to CH2 and 2Khz signal to CH3.
2. Connect Tx output to Rx Input.
3. Connect Tx Clock to Rx Clock.
4. Connect Tx CH0 to Rx CH0.
5. Measure the signal frequency at TP11, TP13, TP15 and TP17.
6. Observe the TDM modulation out at TP20.
7. Observe the Demodulation signal frequency at TP42, TP44, TP46, and TP48 and verify that they are same as at TP11, TP13, TP15, and TP17 respectively.

7. Objective: To Study & Observe the Sampling and Reconstruction of a given signal

DETAILS OF GROUP:

S. No. Name Group Members Roll No.

Signature

SIGNATURES:
S. K. Chaurasiya
Faculty incharge: **Technical Assistant**

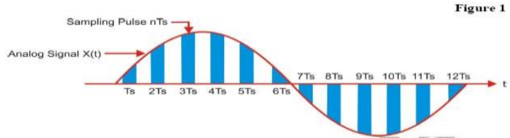
APPARATUS REQUIRED:

S.No.	**Name of Instruments/Kit**	**Specifications**	**Quantity**
1.	ST2101 with power supply cord		
2.	CRO with connecting probe		
3.	Connecting cords	As Per Required	

Theory:- The Sampling Theorem states that a signal can be exactly reproduced if it is sampled at a frequency Fs, where Fs is greater than twice the maximum frequency Fmax in the signal. Fs > 2· Fmax The frequency 2· Fmax is called the Nyquist sampling rate. Half of this value, Fmax, is sometimes called the Nyquist frequency. The sampling theorem is considered to have been articulated by Nyquist in 1928 and mathematically proven by Shannon in 1949. Some books use the term "Nyquist Sampling Theorem", and others use "Shannon Sampling Theorem". They are in fact the same sampling theorem. The signals which are required to be transmitted as information is known as information signal and in the case of voice communication this will be a continuously changing signal containing speech information. The aim of the kit is to transmit the signals in digital form and is to reproduce this information signal in analog form at the receiving end of the communication system with the help

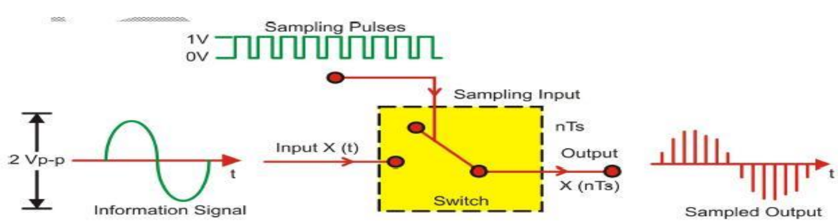
of sampling and reconstruction trainer.
In the exercises to follow, you will simulate audio signal by a 1 KHz test signal provided On-board. The repetitive, non-changing waveform does not contain information. Provided the frequency of the test-signal lies within the frequency range which an information signal will occupy, a test signal of this type can be extremely helpful in system analysis and testing.
The voice signals are limited to the range 300 Hz to 3.4 KHz, a 1 KHz frequency fits conveniently in this range and can be used to demonstrate and test many techniques used in communication system. **Principle of Sampling: -**
Analogue signal x(t) that can be viewed as a continuous function of time, as shown in figure1. We can represent this signal as a discrete time signal by using values of x(t) at intervals of nTs to form x(nTs) as shown in figure 1. We are "grabbing" points from the function x(t) at regular intervals of time, Ts, called the sampling period**.**

Figure 1



Sampling of signal at sampling interval (period) T_s

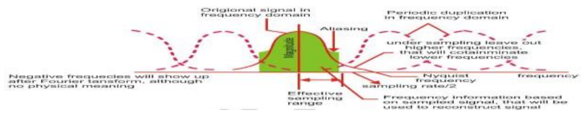
The sampling theorem clearly states what the sampling rate should be for a given range of frequencies. In practice, however, the range of frequencies needed to faithfully record an analog signal is not always known beforehand. Nevertheless, engineers often can define the frequency range of interest. As a result, analog filters are sometimes used to remove frequency components outside the frequency range of interest before the signal is sampled. For example, the human ear can detect sound across the frequency range of 20 Hz to 20 KHz. According to the sampling theorem, one should sample sound signals at least at 40 KHz in order for the reconstructed sound signal to be acceptable to the human ear. Components higher than 20 KHz cannot be detected, but they can still pollute the sampled signal
through aliasing. Therefore, frequency components above 20 KHz are removed from the sound signal before sampling by a band-pass or low-pass analog filter.

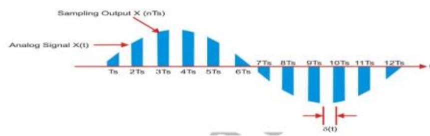


Basic Sampling Process

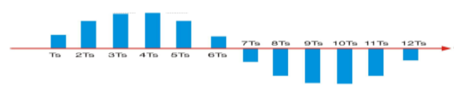
Natural sampling:-

In the analogue-to-digital conversion process an analogue waveform is sampled to form a series of pulses whose amplitude is the amplitude of the sampled waveform at the time the sample was taken. In natural sampling the pulse amplitude takes the shape of the analogue waveform for the period of the sampling pulse as shown in





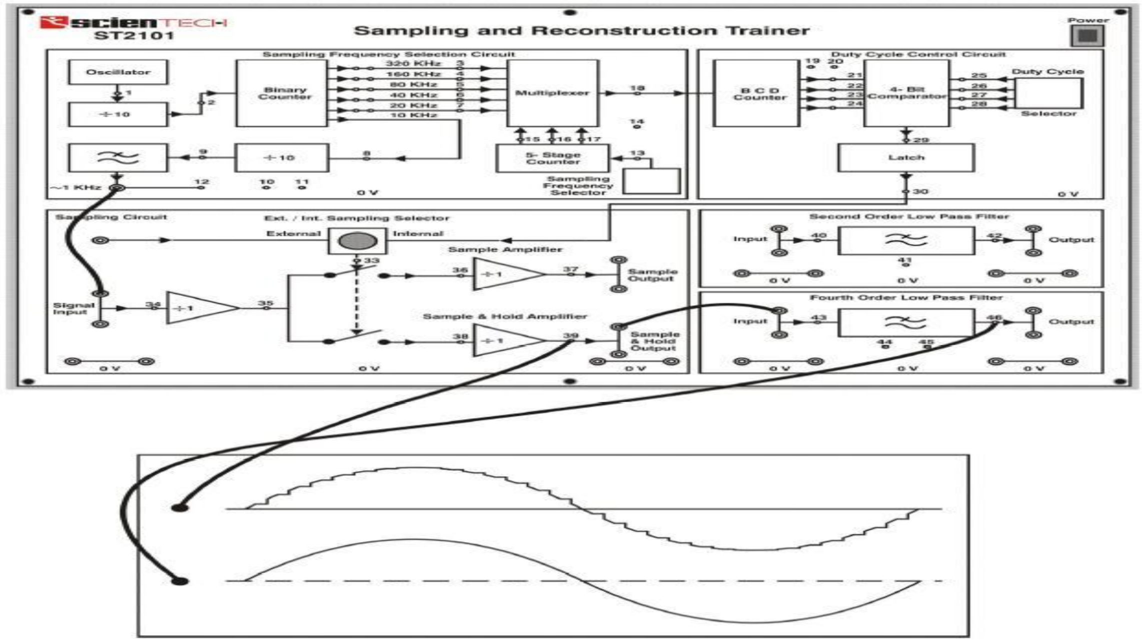
Flat Top sampling:
After an analogue waveform is sampled in the analogue-to-digital conversion process, the continuous analogue waveform is converted into a series of pulses whose amplitude is equal to the amplitude of the analogue signal at the start of the sampling process. Since the sampled pulses have uniform amplitude, the process is called flat top sampling as shown in figure



Types of sampling:-

Over Sampling:
Graphically, if the samplingrate is sufficiently high, i.e., greater than the Nyquist rate, there will be no overlapped frequency components in the frequency domain. A "cleaner" signal can be obtained to reconstruct the original signal. This argument is shown graphically in the frequency-domain figure 1(a) and time-domain figure

Over sampling in Frequency Domain



Procedure:--

A. Setup for sample output & reconstructed signal output. Initial set up of trainer: Duty cycle selector switch position: Position 5 Sampling selector switch: Internal position

1. Connect the power cord to the trainer. Keep the power switch in ‘Off’ position.
2. Connect 1 KHz Sine wave to signal Input. (Shown in Figure)
3. Switch ‘On’ the trainer's power supply & Oscilloscope.
4. Connect BNC connector to the CRO and to the trainer’s output port.
5. Select 320 KHz (Sampling frequency is 1/10th of the frequency indicated by the Illuminated LED) sampling rate with the help of sampling frequency selector switch.

6. Observe the Sample output (TP37) and the Fourth Order low pass filter’s output (TP46). (Shown in Figure)

7. Vary the position of Duty Cycle Selector switch from 0 to 9 & observe the Sample Output changes and the amplitude of Fourth Order low pass filter's output changes. The amplitude of the reconstructed signal increases from 0 to 9 position of Duty Cycle Selector switch.

B. Setup for sample output & reconstructed signal output.

1. Disconnect the Sample Output from Fourth Order low pass filter input. Connect Sample & Hold output to Fourth Order low pass Filter's Input. Set the

Duty Cycle Selector switch to position 5. (Shown in Figure)

2. Observe the waveform at Sample & Hold output (TP39) on oscilloscope. Vary the sampling frequency selector from 32 KHz to 2 KHz to illustrate how each sample is held at the sample/hold output. Also observe the filter output at TP46. (Shown in Figure)

3. Vary the position of Duty Cycle Selector switch from 0 to 9 and note that in contrast to step 7, the filter's output amplitude is now independent of the sampling duty cycle and is equal to the amplitude of the original input signal.

This is an important result - with Sample and Hold Output, the proportion of sampling time to holding time has no effect on reconstructed waveform provided that Nyquist criteria has been followed. If sample/hold feature is utilized in digital communication system many channels can be multiplexed with maximum amplitude of reconstructed signal.

Precautions:-

1. The connections should be made properly and tightly. 2. Checkallthe connections before switching ON the kit.

Conclusion:

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

Book: Lab experiment related theory available in following books: 1. Taub and Schilling: Principles of Communication Systems, TMH.

2. Lathi: Modern Digital and Analog Communication Systems, Oxford University Press. 3. Simon Haykins: Communication Systems, John Wiley.

4. Ranjan Bose: Information Theory, Coding and Cryptography, TMH.

8. OBJECT: - Study of ASK transmitter and receiver.

Apparatus Required:-

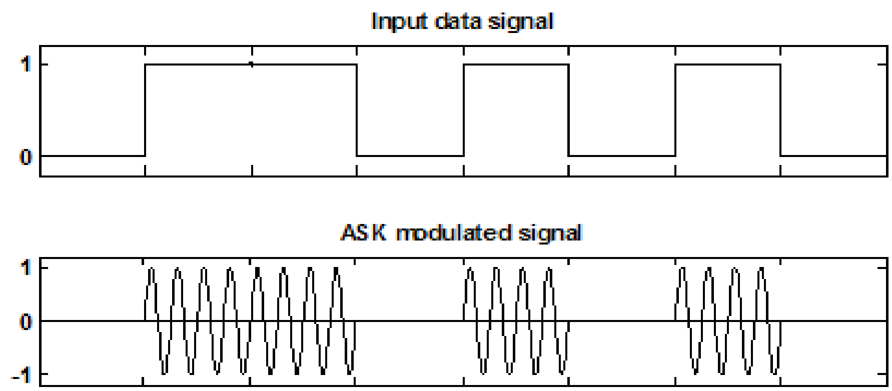
ASK Modulation Demodulation kit, CRO, Patch chords

Theory:-

The simplest method of modulating a carrier with a carrier stream is to change the amplitude of carrier wave every time the data changes. This modulation technology is known as Amplitude Shift Keying.

ASK is obtained by switching ON the carrier to be obtained '1' and switching OFF when D=1, i.e. carrier is transmitted & D=0, i.e. carrier is suppressed. This technology is ON-OFF keying. Fig1 shows ASK for a given data stream. A linear multiplier generates it. O/P voltage is a product of ac coupled carrier and the information signal or modulating signal. For a double balanced modulator, data stream applied is 0V at logic 0 & +5V at logic 1. The O/P is sine wave unchanged in phase when '1' is applied. Carrier is multiplied by +ve constant voltage when bit 0 is applied.

Fig 2. shows demodulator of ASK waveform at receiver. A diode rectifier first rectifies it. After rectification, signal is passed through Low Pass Filter to remove carrier. These rounded pulse are then squared up by passing it through voltage comparator set at a threshold level. If I/P volt > Threshold level; O/P is +5V.



BLOCK DIAGRAM:-

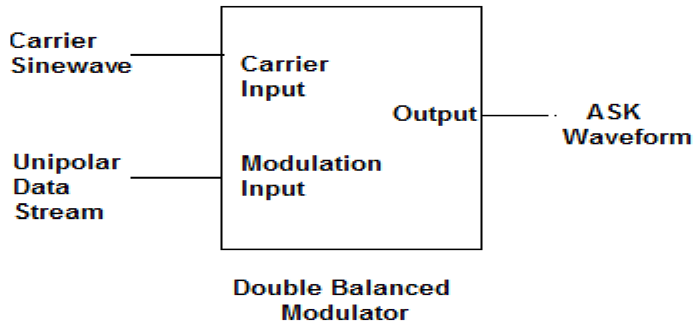


Fig. Amplitude shift Keying

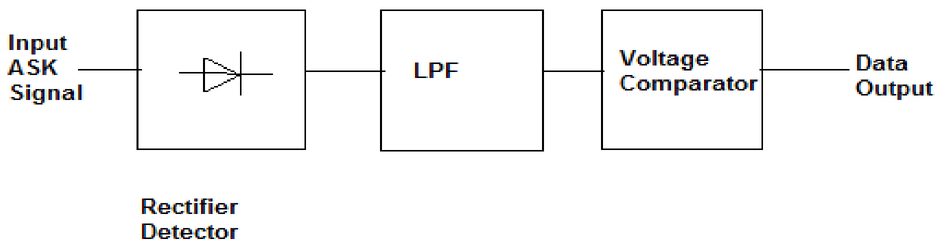


Fig : ASK Demodulator

Precautions:-

1. Check the connections before switching ON the kit.
2. Observations should be taken properly.

Procedure: -

1. Ensure that the group 4 (GP 4) Clock is selected in the clock generation Section. Selection is done with the help of switch S1 . Observe the corresponding LED indication.

2. Observe the Transmitter clock of frequency 250 KHz at Tx Clk Post.
3. Set the data pattern using switch S4 as per the given block Diagram.
4. Observe the 8 bit data pattern at S DATA post.
5. Observe the Carrier sine wave of frequency 1MHz at SIN 3 post in the Carrier section
6. Connect SIN 3 post to the IN2 post and In3 post to the ground in carrier modulation section.

7. Connect S DATA to IN 16 post and Tx Clk to Clk 2 post of the encoded data Section.
8. Select NRZ-L data with the help of the switch S3 and observe the corresponding LED indication in the encoded data section.
9. Connect OUT 10 post of the encoded data section to IN 4 post as a control input for the carrier modulation section.
10. Observe the ASK modulated Signal at the OUT 2 post of the Carrier Modulator section.
11. For the demodulation of the ASK modulated data connect the OUT 2 post of the carrier modulator to the IN 24 post of the ASK demodulator section.
12. Observe the ASK demodulated data OUT 20 post of the ASK demodulator section.
13. Verify the recovered data with the S DATA

9. OBJECT Study of FSK transmitter and receiver.

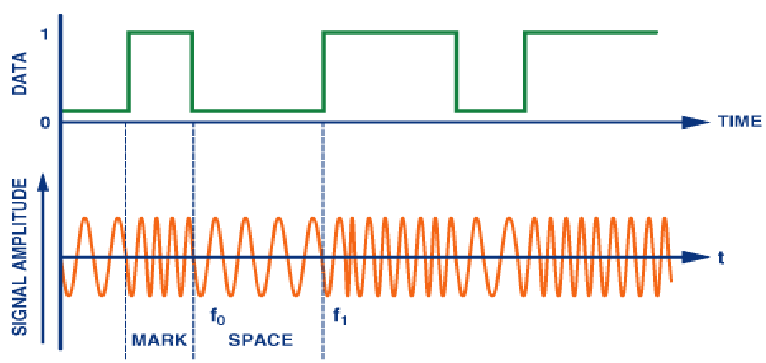
Apparatus Required:-

FSK Modulator/Demodulator Trainer kit ,CRO, CRO probes.
In Frequency shift keying, the carrier frequency is shifted (i.e. from one frequency to another) corresponding to the digital modulating signal. If the higher frequency is used to represent a data ‘1’ & lower frequency a data ‘0’, the resulting FSK waveform appears. Thus

Data =1 High Frequency

Data =0 Low Frequency

It is also represented as a sum of two ASK signals. The two carriers have different frequencies & the digital data is inverted. The demodulation of FSK can be carried out by a PLL. As known, the PLL tries to ‘lock’ the input frequency. It achieves this by generating corresponding O/P voltage to be fed to the VCO, if any frequency deviation at its I/P is encountered. Thus the PLL detector follows the frequency changes and generates proportional O/P voltage. The O/P voltage from PLL contains the carrier components. Therefore to remove this, the signal is passed through Low Pass Filter. The resulting wave is too rounded to be used for digital data processing. Also, the amplitude level may be very low due to channel attenuation.



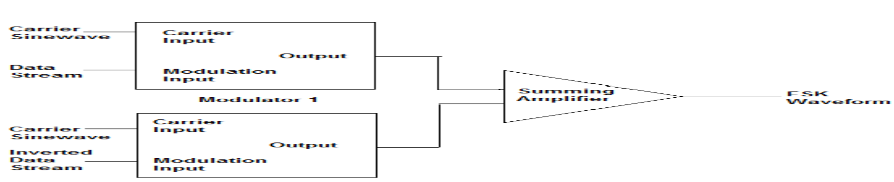


Fig. : FSK Modulator

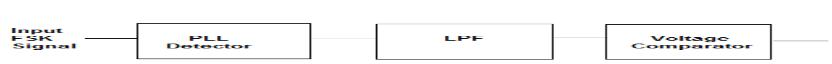


Fig. FSK Demodulator

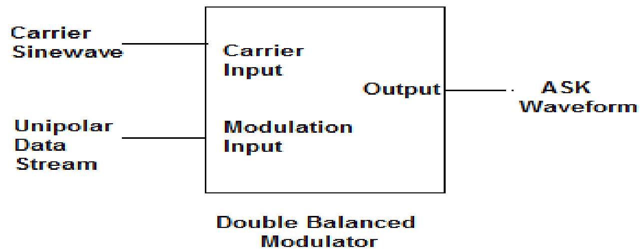


Fig. Amplitude shift Keying

Procedure: -

1. Ensure that the group 4 (GP 4) Clock is selected in the clock generation Section. Selection is done with the help of switch S1 . Observe the corresponding LED indication.
2. Observe the Transmitter clock of frequency 250 KHz at Tx Clk Post.
3. Set the data pattern using switch S4 as per the given block Diagram.
4. Observe the 8 bit data pattern at S DATA post.
5. Observe the Carrier sine wave of frequency 1MHz at SIN 3 post in the Carrier section
6. Connect SIN 3 post to the IN2 post and In3 post to the ground in carrier modulation section.
7. Connect S DATA to IN 16 post and Tx Clk to Clk 2 post of the encoded data Section.
8. Select NRZ-L data with the help of the switch S3 and observe the corresponding LED indication in the encoded data section.
9. Connect OUT 10 post of the encoded data section to IN 4 post as a control input for the carrier modulation section.
10. Observe the ASK modulated Signal at the OUT 2 post of the Carrier Modulator section.
11. For the demodulation of the ASK modulated data connect the OUT 2 post of the carrier modulator to the IN 24 post of the ASK demodulator section.
12. Observe the ASK demodulated data OUT 20 post of the ASK demodulator section.
13. Verify the recovered data with the S DATA

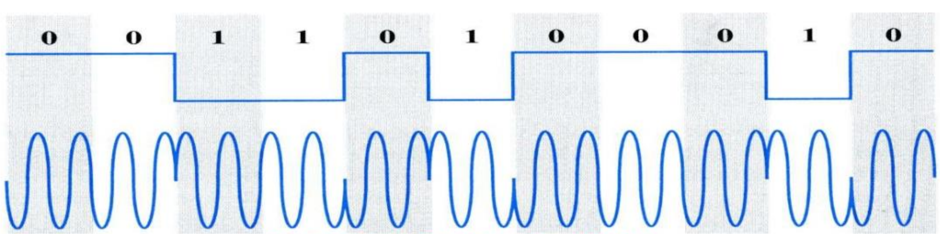
1. **Precautions**:-Check the connections before switching ON the kit.
   2. Observations should be taken properly.

10. OBJECT Study of PSK transmitter and receiver.

Apparatus Required:- PSK Modulator/Demodulator Trainer kit, CRO, CRO probes.

Theory:-

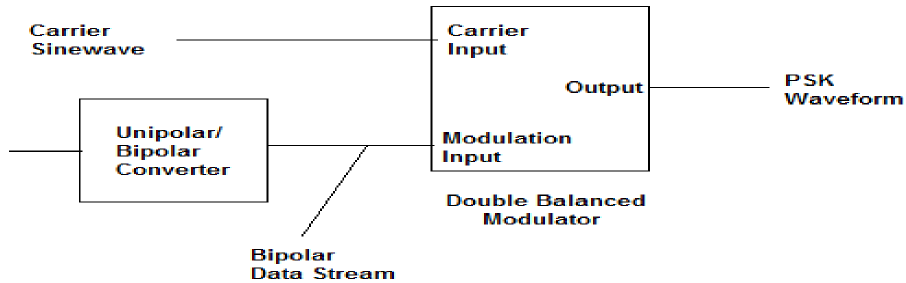
Phase shift keying involves the phase change of the carrier sine wave between 0 and 180 in accordance with the data stream to be transmitted. PSK is also known as Phase reversal keying. PSK modulator is shown in figure 1. Functionally, the PSK modulator is very similar to the ASK modulator. Both uses balanced modulator to multiply the carrier with the modulating signal. But in contrast to ASK techniques, the digital signal applied to the modulator input for PSK generation is bipolar i.e. have equal +ve and –ve voltage levels. The unipolar – bipolar converter converts the unipolar data stream to bipolar data. At receiver, the square loop detector circuit is used to demodulate the transmitted PSK signal. The demodulator is shown in figure 2. The incoming PSK signal with 0 & 180 phase changes is first fed to the signal square, which multiplies the input signal by itself. The phase adjust circuit allows the phase of the digital signal to be adjusted w.r.t the input PSK signal. Also its O/P controls the closing of an analog switch. When the output is high the switch closes and the original PSK signal is switched through the detector.

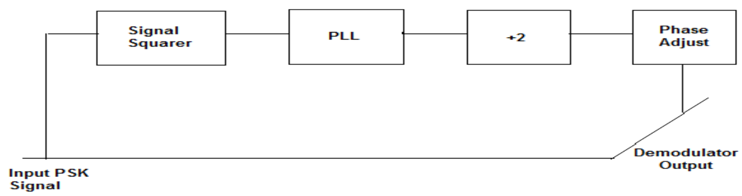


Phase Shift Keying (PSK)

Or called BPSK, uses two phases to represent 0 & 1

BLOCK DIAGRAM -





Procedure: -

1. Ensure that the group 4 (GP 4) Clock is selected in the clock generation Section. Selection is done with the help of switch S1 . Observe the corresponding LED indication.

2 . Observe the Transmitter clock of frequency 250 KHz at Tx Clk Post.
3. Set the data pattern-using switch S4 as per the given block Diagram.
4. Observe the 8 bit data pattern at S DATA post.
5. Observe the Carrier sine wave of frequencies 1MHz at SIN 2 post and 1 MHz with 180o phase at SIN 3 post in the Carrier section.
6. Connect SIN 2 post to the IN2 post and SIN 3 post to the IN 3 post of the carrier modulator section.
7. Connect S DATA to In 16 post and Tx Clk to Clk 2 post of the encoded data Section.
8. Select NRZ-L data with the help of the switch S3 and observe the corresponding LED indication in the encoded data section.
9. Connect OUT 10 post of the encoded data section to IN 4 post as a control input for the carrier modulation section.
10. Observe the PSK modulated Signal at the OUT 2 post of the Carrier Modulator section.
11. For the demodulation of the PSK modulated data connect the OUT 2 post of the carrier modulator to the IN 30 post of the PSK demodulator section.
12. Observe the PSK demodulated data OUT 27 post of the PSK demodulator section.

13. Verify the recovered data with the S DATA.

Precautions:-

1. Check the connections before switching ON the kit.
2. Observations should be taken properly