Analog Communication

LABORATORY MANUAL

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

COLLEGE OF ENGINEERING AND MANAGEMENT, KOLAGHAT.

Revised December, 2024

VISION

Pursuing Excellence in Teaching-Learning Process to Produce High Quality Electronics and Communication Engineering Professionals.

MISSION

To enhance the employability of our students by strengthening their creativity with different innovative ideas by imparting high quality technical and professional education with continuous performance improvement monitoring systems.

To carry out research through constant interaction with research organizations and industry.

PROGRAM EDUCATIONAL OBJECTIVES (PEOs)

PEO-1	Attain a solid foundation in electronics & communication engineering fundamentals with an attitude to pursue continuing education and to succeed in industry/technical profession through global education
PEO-2	Ability to function professionally in an increasingly international and rapidly changing world due to the advances in emerging technologies and concepts.
PEO-3	Exercise excellent leadership qualities on multi-disciplinary and multi-cultural teams at levels appropriate to their experience, which addresses issues in a responsive, ethical and innovative manner.
PEO-4	Contribute to the needs of the society in solving technical problems using electronics & communication engineering principles, tools and practice.

PO1	**Engineering knowledge**	Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.
PO2	**Problem analysis**	Identify, formulate, review research literature, and analyse complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.
PO3	**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.
PO4	**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.
PO5	**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.
PO6	**The engineer and society**	Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.
PO7	**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
PO8	**Ethics**	Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.
PO9	**Individual and team work**	Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings
PO10	**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, and give and receive clear instructions
PO11	**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.
PO12	**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.

PROGRAM SPECIFIC OUTCOMES (PSOs)

PSOs-1	An ability to design and conduct the experiments, analyse and interpret the data using modern software or hard ware tools with proper understanding(basic conceptions) of Electronics and Communication Engineering.
PSOs-2	Ability to identify, formulate & solve problems and apply the knowledge of electronics and communication to develop techno-commercial applications.

Course Outcomes of Analog Communication Lab (EC491), Second year 4th Semester,ECE
CO1: *Concept of AM (DSB, SSB, Simple AM) modulation & Demodulation techniques and modulation index measurement.*
CO2: *Concept of FM (NBFM, WBFM) modulation and demodulation techniques by trainer kit and VCO& PLL.*
CO3: *Measurement of SNR of RF Amplifier. Calculate different parameters of the super heterodyne radio receiver.*

Exp. No.	Experiments Title	CO's
1.	Measurement of modulation index of an AM signal.	1
2.	Measurement of output power with varying modulation index an AM signal (for both DSB- & SSB).	1
3.	Measurement of distortion of the demodulated output with varying modulation index of an AM signal (for both DSB-SC & SSB).	1
4.	Measurement of power of different frequency components of a frequency modulated signal & the measurement of the bandwidth.	2
5.	Design and set up a PLL using VCO & to measure the lock frequency.	2
6.	Design and set up a FM demodulator using PLL.	2
7.	Measurement of SNR of a RF amplifier.	3
8.	Measurement of selectivity, sensitivity, fidelity of a super-heterodyne receiver.	3

Content beyond the syllabus		
1	**Signal generation and modulation:** **AM with Nonlinear devices** • Use a diode or transistor as a nonlinear element to generate AM.	

MAPPING OF COURSE OUTCOMES (COs) WITH PROGRAM OUTCOMES(POs):

	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12
CO1	3	2	1	2	2	1	2	1	2	1	2	3
CO2	3	2	1	2	2	1	2	1	2	1	2	3
CO3	3	2	1	2	2	1	2	1	2	1	2	3
CO4	3	2	1	2	2	1	2	1	2	1	2	3
AVG.	3	2	1	2	2	1	2	1	2	1	2	3

Table of Contents

Table of Contents

Introduction

This first laboratory in Analog Communication has the objective to familiarize the student with the circuit, experiment kit, oscilloscope reading and handling of different instrument related to experiment. Another goal is to reinforce theoretical knowledge with practice and vice-versa, and also to learn correct laboratory procedures and techniques. This is accomplished by building, testing and taking measurements on circuits.

In the execution of the experiments, highest benefit is gained if someone can distinguish between performing the experiment by following step by step instruction, and actually understanding the reasons and methodology. To understand the experiments, theory of the circuit must be understood and to convey the results a correct laboratory report is to be learnt to write.

Guidelines for Laboratory Notebook

The laboratory notebook is a record of all work pertaining to the experiment. Organization in notebook is important. Descriptive heading should be used to separate and identify the various parts of the experiment. A neat, organized and complete record of an experiment is just as important as the experimental work.

1. **Heading:** The experiment identification (number) should be at the top of each page.

2. **Title:** The name of the experiment.

3. **Objective:** A brief but complete statement of what to find out or verify in the experiment should be at the beginning of each experiment.

4. **Theory:** A brief theory of the experiment should be written in report.

5. **Apparatus Required:** Laboratory equipment refers to the various tools and equipment used in a laboratory for experiment.The laboratory apparatus depends upon the type of laboratory you are in and the experiment you are going to perform.

6. **Block/Connection Diagram:**Always use the simplest diagrams that will serve the purpose. Often this will mean drawing them specifically for the report, rather than reproducing existing diagrams. Each diagram should have a figure number and a caption and should be referred to in the text.

7. **Procedure:** How to perform the experiment. In general, lengthy explanations of procedure are unnecessary. Short commentaries alongside the data may be used.

8. **Graphs / Waveform / Figures:** Figures are used to present large amounts of data in concise visual form. Each curve if more than one on a graph should be labelled.

9. **Analysis and Results:** This section should summarize and display the results of the experiment. This section should be purely factual, where the results are displayed primarily in the form of graphs. Describe the results clearly and concisely .The results should be presented in a form which makes the interpretation easy. Theoretical and experimental results should be on the same graph or arrange in the same table in a way for easy correlation of these results.

10. **Conclusion:** This is the interpretation of results of the experiment as an engineer or designer and should be brief and specific.

EXPERIMENT # 1

Title: STUDY OF AMPLITUDE MODULATION & DEMODULATION

Objective:

- Double Sideband **AM**and **DSB-SC**Generation (MODULATION)
- Double Sideband **AM** Reception (DEMODULATION)

Theory:

In amplitude modulation the amplitude of a high frequency carrier is modulated with a low frequency message signal called modulating signal.
Therefore,
If V_s **(t) =** $A_s\cos\mu_s t$ is the message signal and V_c **(t) =** $A_c\cos\mu_c t$ is the carrier signal then, the modulated carrier is given by
Vm (t) = $(A_c + kA_s\cos\mu_s t)\cos\mu_c t$, **k** being a constant of proportionality
Or, $V_m(t) = A_c(1+m\cos\mu_s t)\cos\mu_c t$
Where **m= k.** A_s / A_c is called the modulation index. ….. (1)
At 100% modulation, the modulation index is **m=1**, hence
$k=1/m = A_c / A_s$ ….. (2)
If at another value of modulation index, **m**, the Maximum and minimum of the modulated carrier peak are respectively V_{max} and V_{min},
Then $m = (V_{max} - V_{min}) / (V_{max} + V_{min})$ ……………… (3)

Power of the DSB-SC modulation:
Equation for DSB-SC is given by the product of message signal and carrier:

$$A_m A_c cos(2\pi f_m t) cos(2\pi f_c t) \ldots\ldots\ldots\ldots.(4)$$

Using the trig. identity: $cos(A)cos(B) = \frac{1}{2}$
We can rewrite our DSB SC as: $\frac{A_m A_c}{2}[cos\{2\pi(f_m+f_c)t\} + cos\{2\pi(f_c-f_m)t\}] \ldots\ldots\ldots.(6)$
The total power is the sum of the power of the sidebands $P_t = P_{USB} + P_{LSB} \ldots\ldots\ldots\ldots(7)$
But upper side band power will be the same as the lower side band power since they have the same amplitude. Hence

$$P_t = 2.P_{USB} \ldots\ldots\ldots\ldots.(8)$$

We also know that power can be calculated as: $P = \frac{V_{RMS}^2}{R} = \frac{\left(\frac{V_m}{\sqrt{2}}\right)^2}{R} \ldots\ldots\ldots\ldots.(9)$

Power of upper side band will be given as:

$$P_{USB} = P_{LSB} =$$

Therefore, the power of DSBSC wave is

$$P_t = P_{USB} + P_{LSB} = 2.P_{USB} = \frac{{A_m}^2 {A_c}^2}{4R} \ldots\ldots\ldots\ldots\ldots\ldots.(11)$$

$WhereRisload \in Ohm$

Apparatus Required:

- Amplitude Mod./Demod. Experiment Kit.
- Function Generator.
- A CRO / DSO

Block Diagram:

Modulation Section-

Demodulation Section-

Fig.1.1

Procedure:

This experiment investigates the generation of double sideband amplitude modulated (AM) waveforms, using the **ST2201** module. By removing the carrier from such an AM waveforms, the generation of double sideband suppressed carrier (DSBSC) AM is also investigated.

1. Ensure that the following initial conditions exist on the board.

   a. Audio input select switch should be in INT position:
   b. Mode switch in DSB position.
   c. Output amplifier's gain potentiometer in full clockwise position.
   d. Speakers switch in OFF position.

2. Turn on power to the ST2201 board.
3. Turn the audio oscillator block's amplitude pot to its full clockwise (MAX) position, and examine the block's output (TP14) on an oscilloscope. This is the audio frequency sine wave which will be as our modulating signal. Note that the sine wave's frequency can be adjusted from about 300 Hz to approximately 3.4 KHz, by adjusting the audio oscillator's frequency potentiometer. Note also that the amplitude of this audio modulating signal can be reduced to zero, by turning the Audio oscillator's amplitude present to its fully counterclockwise (MIN) position. Return the amplitude present to its max position.
4. Turn the balance pot, in the balanced modulator & band pass filter circuit 1 block, to make *100% modulation*. It is this block that we will use to perform *double-side band amplitu demodulation.*
5. Monitor, and measure the amplitude (As and Ac) in turn, the two inputs to the balanced modulator & band pass filter circuits 1 block, at TP1 and TP9. Note that:
   a. The signal at TP1 is the audio-frequency sine wave from the audio oscillator block. This is the modulating input to double-side band modulator.
   b. Test Point 9 carries a sine wave of 1MHz frequency. This is the carrier input to double-sideband modulator.
6. Next, observe the output of the balanced modulator & band pass filter circuit 1Block (at tp3),together with the modulating signal at TP1Trigger the oscilloscope on theTP1 signal.

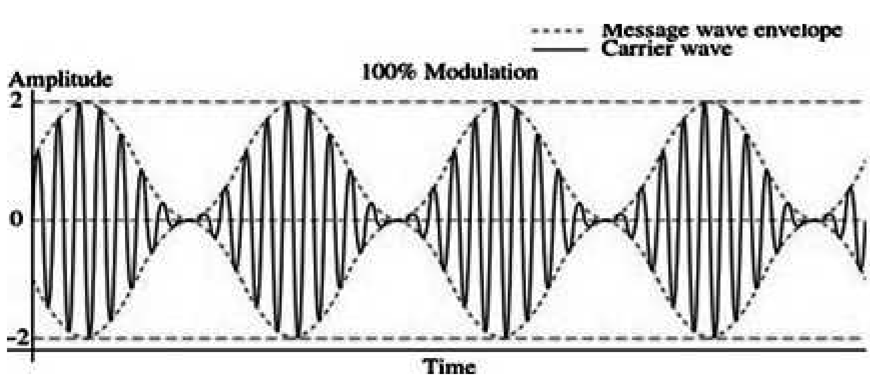


Fig.1.2

Now find the value of 'k' (constant of proportionality) using the equation (2) from the output wave form.The output from the balanced modulator & band pass filter circuit 1 block (at TP3) is a double-sideband. AM waveform, which has been formed by amplitude-modulating the 1MHz carrier sine wave with the audio-frequency sine wave from the audio oscillator.

The frequency spectrum of this AM waveform is as shown below in Figure1.3, where fm is the frequency of the audio modulating signal.

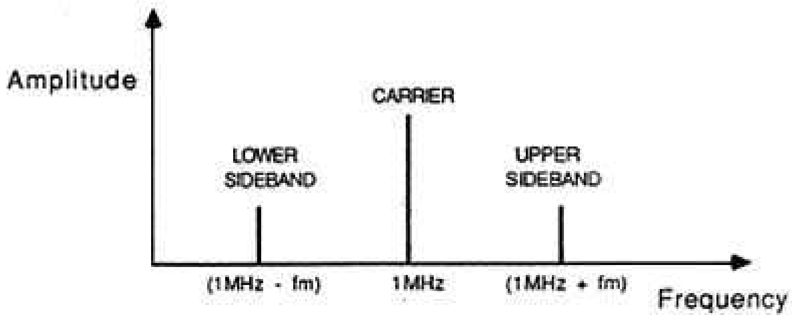


Fig. 1.3

7. Keeping the carrier amplitude unchanged calculate the message amplitude required for 30%, 50%, 75%, 85% and 90% modulations using the equation (1) respectively and measure the percentage of modulation from the output waveforms respectively using the equation (3).

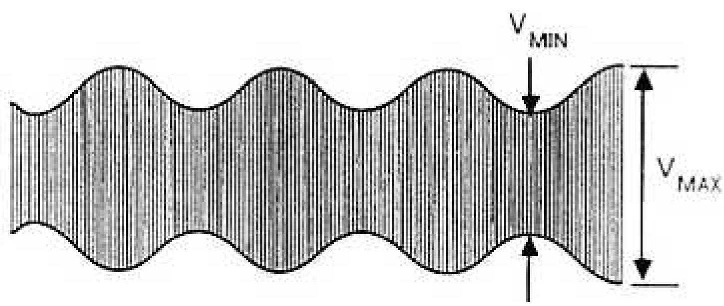


Fig. 1.4

DSB-SC

8. Now vary the amplitude and frequency of the audio-frequency sine wave, by adjusting the amplitude and frequency present in the audio oscillator block. Note the effect that varying each pot has on the amplitude modulated waveform. The amplitude and frequency amplitudes of the two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. Do this by turning the amplitude pot to its MIN position, and note that the signal at TP3 becomes an un-modulated sine wave of frequency 1 MHz, indicating that only the carrier component now remains. Return the amplitude pot to its maximum position. Now turn the balance pot in the balanced modulator & band pass filter circuit 1 block, until the signal at TP3 is as shown in Figure 1.5

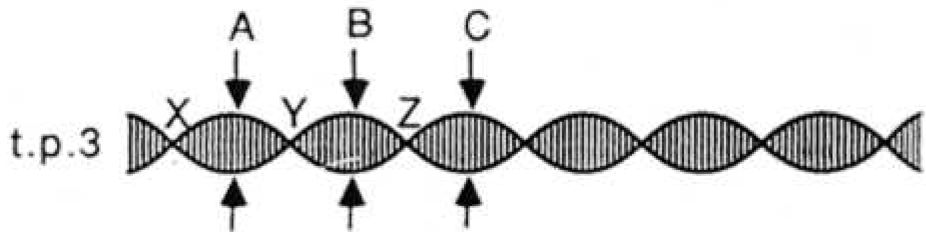


Fig. 1.5

The balance pot varies the amount of the 1 MHz carrier component, which is passed from the modulator's output. By adjusting the pot 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. Note that once the carrier has been balanced out, the amplitude of TP3's waveform should be zero at minimum points X, Y, Z etc. If this is not the case, it is because one of the two sidebands 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 sidebands equally. This is achieved by carefully trimming transformer T1, until the waveform's amplitude is as close 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 Figure 1.6&1.7 respectively.

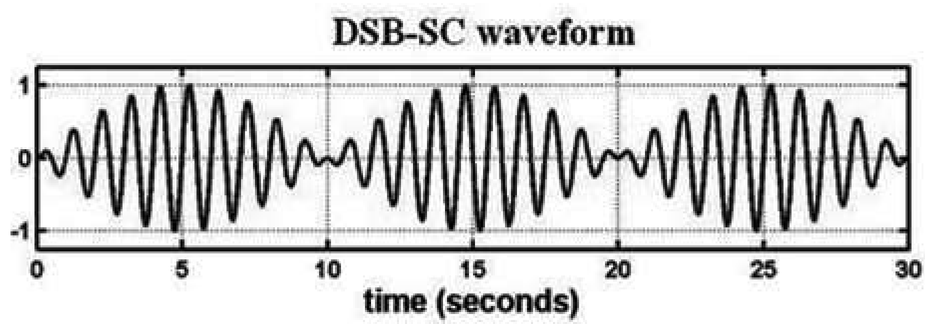


Figure 1.6

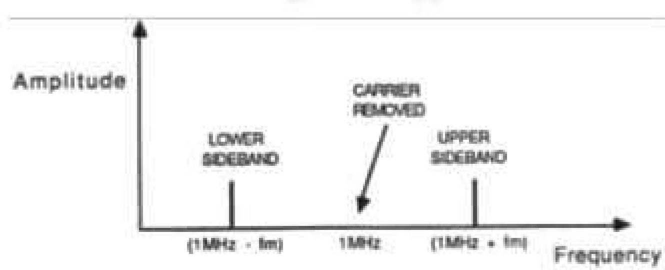


Figure 1.7

DOUBLE SIDEBAND AM RECEPTION (DEMODULATION)

Procedure:

This experiment investigates the reception and demodulationof AM waveforms by the ST2201/ ST2202 module. Both AM broadcast signals, and AM transmissions from ST2201, will be examined, and the operation of automatic gain control at the receiver will be investigated.

1. Position the ST2201 & ST2202 modules, with the ST2201 board on the left, and a gap of about three inches between them.

2. Ensure that the following initial conditions exist on the ST2201 board.
   a. Audio oscillator's amplitude pot in fully clockwise position.
   b. Audio input select switch in ANT position for **wireless** transmission.
   c. Balance pot in balanced modulator & band pass filter circuit 1 block, in full clockwise position.
   d. Mode switch in DSB position.
   e. Output amplifier's gain pot in full counter-clockwise position.
   f. TX output select switch in ANT position.
   g. Audio amplifier's volume pot in fully counter-clockwise position.
   h. Speaker switch in ON position.
   i. On-board antenna in vertical position, and fully extended for **wireless** transmission.

3. Ensure that the following initial conditions exist on the ST2102 board.
   a. RX input select switch in ANT position for **wireless** reception.
   b. R.F. amplifier's tuned circuit select switch in INT position.
   c. R.E amplifier's gain pot in fully clock-wise position;
   d. AGC switch in INT position.
   e. Detector switch in diode position.
   f. Audi o amplifier's volume pot in fully counter-clockwise position.
   g. Speaker switch in ON position.
   h. Beat frequency oscillator switch in OFF position.
   i. On-board antenna in vertical position, and fully extended for **wireless** reception.

4. Turn on power to the modules.

5. On the ST2202 module, slowly turn the audio amplifier's volume pot clockwise, until sounds can be heard from the on-board loudspeaker. Next, turn the vernier tuning dial

until a broad cast station can be heard clearly,and adjust the volume control to a comfortable level.

Note: If desired, headphones (supplied with the module) may be used instead of the on-board loudspeaker. To use the headphones, simply plug the headphone jack into the audio amplifier block's headphones socket, and adjust controlled block's volume pot.

6. The final stage of the receiver is the audio amplifier 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 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 TP39 (providing that the audio amplifier block' s volume pot is not in its minimum volume position). Compare this signal with that at the diode detector' s output (TP31), and note how the audio amplifier block's low pass filter has 'cleaned up' the audio signal . You may notice that the output from the audio amplifier block (tp39) is inverted with respect to the signal at the output of the diode detector (TP31) this inversion is performed by the audio power amplifier IC, and in no way affects the sound produced by the receiver. Trace the output from the audio amplifier block and measure the frequency.

Experimental Data:

Table 1

Sl. No.	Modulating Signal		Carrier Signal		Modulation Index (m)		Demodulated Signal	
	Freq. (f_m)	Ampl. (A_m)	Freq. (f_c)	Ampl. (A_c)	Theo. Value (m_T)	Pract. Value (m_P)	Freq. (f_D)	Ampl. (A_D)

LPF Response:

Table 2

Sl. No	Input Signal		Output Signal	
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)
1.	f_1	A_{in}	f_{o1}	A_1
2.	f_2	A_{in}	f_{o2}	A_2
3.	f_3	A_{in}	f_{o3}	A_2
n.	f_n	A_{in}	f_{o4}	A_n

Linearity:

Table 3

Sl. No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_{o1}	A_{o1}	M_1
2.	f_1	A_2	f_{o1}	A_{o2}	M_2
3.	f_1	A_2	f_{o1}	A_{03}	M_3
n.	f_1	A_n	f_{o1}	A_{on}	M_n

Table 4

Sl. No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_{o1}	A_{o1}	M_1
2.	f_2	A_1	f_{o2}	A_{o2}	M_2
3.	f_3	A_1	f_{o3}	A_{03}	M_3
n.	f_n	A_1	f_{on}	A_{on}	M_n

Data Analysis:

1. Calculate the ratio of amplitudes of input and output signals and show the linearity of LPF.
2. Find the cut off frequency of LPF. Tally the frequency of input and output signal of modulator and demodulator.

Discussion:

Write briefly your comments about the above experiment.

Precautionary Measure to be taken:

1. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
2. For connecting any signal from one equipment to another equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.

3. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.
4. For unusual spark or burning smell, immediately switch off the main supply to the kit.
5. Ensure that, a signal is connected to an appropriate junction destined for it.
6. Always cover the equipment for dust protection after the experiment.

Some Sample questions:

1. Draw the LPF response curve taking the base frequency 1 KHz and amplitude (P-P) 2v and increasing the input frequency in steps of 1 KHz. Measure the cut off frequency range.
2. Draw the linearity response curve of the a AM modulator taking a input frequency 2KHz, 4KHz,6KHz and 2v (p-p). Specify the linearity region of the curve with rage of magnitude of input signal.
3. Taking suitable signal and carrier, perform an amplitude modulation for 30%modulation index. Trace the modulated signal and demodulated output.

EXPERIMENT # 2

Title: STUDY OF SINGLE SIDE BAND SUPPRESSED CARRIER (SSB-SC) & DEMODULATION

Objective:

- Single Sideband **AM** Generation (MODULATION)
- Single Sideband **AM** Reception (DEMODULATION)

Theory:

In amplitude modulation the amplitude of a high frequency carrier is modulated with a low frequency message signal called modulation signal.

Therefore,

If Vs (t) = $A_s cos_s t$ is the message signal and V_c (t) = $A_c cos_c t$ is the carrier signal then, the modulated carrier is given by

V_m (t) = (A_c + $kA_s cos_s t$) $cos_c t$, k being a constant of proportionality

Or, V_m (t) = A_c (1+m $cos_s t$) $cos_c t$ (1)

Where m = k. A_s / A_c is called the modulation index.

If the carrier part of the modulated waveform along with one of them sidebands (preferably the lower sideband) is suppressed then we get the SSB-SC signal and the corresponding equation of the modulated signal becomes

V_m (t) = 1/2 A_c m cos ($_c$ - $_s$)t

Or V_m (t) = 1/2 A_c m cos ($_c$ + $_s$) t (2)

An SSB-SC signal can be generated in several ways. One way of generating an SSB-SC signal is to generating a DSB-SC signal (as demonstrated in the experimental kit) by multiplying the modulating signal with a carrier sinusoid and then suppressing one of the two sidebands of the DSB-SC signal. A band pass filter with high quality factor (Q) and sharp transition band is used for commercial demodulation.

A coherent demodulation technique is used for demodulating a DSB-SC signal. In a coherent demodulation scheme in practice, the carrier waveform is extracted from the modulated signal by using a special "carrier recovery circuit" and the recovered carrier multiplies the modulated signal. Then the desired signal is recovered by passing the multiplier output signal through an appropriately designed filter (LPF) which allows only the modulating signal to pass through it without any distortion and rejects the second term which is entered along the high frequency of 2 Wc. It may be noted here that any phase or frequency error in the recovered carrier introduces noticeable disturbances in the demodulated signal.

This type of amplitude modulation is called Single Side Band (SSB) modulation. Since the carrier is usually not transmitted in this type of modulation, it is called SSB-Suppressed Carrier (SSBSC). For short, we will call it SSB modulation.

Power of the SSB-SC modulation:

Consider the following equation of SSBSC modulated wave:

For Upper Side band

$$s(t) = \frac{\mathbf{A_m A_c}}{\mathbf{2}}[\mathbf{\cos\{2\pi(f_c + f_m)t\}}] \ldots\ldots\ldots..(3)$$

For Lower Side band

$$s(t) = \frac{\mathbf{A_m A_c}}{\mathbf{2}}[\mathbf{\cos\{2\pi(f_c - f_m)t\}}] \ldots\ldots\ldots..(4)$$

Power of SSBSC wave is equal to the power of any one sideband frequency components.

$$\mathbf{P_t = P_{USB} = P_{LSB}} \ldots\ldots\ldots\ldots(5)$$

We know that the standard formula for power of cos signal is

$$\mathbf{P} = \frac{\mathbf{V_{RMS}}^2}{\mathbf{R}} = \frac{(\frac{\mathbf{V_m}}{\sqrt{2}})^2}{\mathbf{R}} \ldots\ldots\ldots\ldots\ldots.(6)$$

In this case, the power of the upper sideband is

$$\mathbf{P_{USB}} = \frac{(\frac{\mathbf{A_m A_c}}{2\sqrt{2}})^2}{\mathbf{R}} = \frac{\mathbf{A_m}^2\mathbf{A_c}^2}{\mathbf{8R}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots(7)$$

Similarly, we will get the lower sideband power same as that of the upper side band power.

$$\mathbf{P_{LSB}} = \frac{\mathbf{A_m}^2\mathbf{A_c}^2}{\mathbf{8R}} \ldots\ldots\ldots\ldots\ldots\ldots\ldots(8)$$

Therefore, the power of SSB-SC wave is

$$\boxed{\mathbf{P_t = P_{USB} = P_{LSB}} = \frac{\mathbf{A_m}^2\mathbf{A_c}^2}{\mathbf{8R}} \ldots\ldots\ldots\ldots\ldots(9)}$$

Where R is load in Ohm

Apparatus Required:

- Amplitude Mod./Demod. Experiment Kit.
- Function Generator.
- A CRO / DSO

Block Diagram:

Modulation Section-

Demodulation Section-

Fig. 2.1

Procedure:

This experiment investigates the generation of signal sideband (SSB) amplitude modulated waveforms, using the **ST2201** module.

1. Ensure that the following initial conditions exist on the board:
    a. Audio input select switch in INT position.
    b. Mode switch in SSB position.
    c. Output amplifier's gain pot in fully clockwise position.
    d. Speaker switch in OFF position.
2. Turn on power to the ST2201 board.
3. Turn the audio oscillator block's amplitude pot to its fully clockwise (MAX) position, and examine the block's output (TP14) on an oscilloscope.
   This is the audio frequency sine wave which will be used as out modulating 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 pot.
   Note:*That the amplitude of this audio modulating signal can be reduced to zero, by turning the audio oscillator's pot to its fully counter-clockwise (MIN) position.*
   Leave the amplitude pot on its full clockwise position, and adjust the frequency pot for an audio frequency of 2 KHz, approx. (mid-way).
4. To achieve signal- sideband amplitude modulation, we will utilize the followingthree blocks on the ST2201 module.
    a. Balanced modulator.
    b. Ceramic band pass filter.
    c. Balanced modulator & band pass filter circuit 2.
5. Monitor the two inputs to the balanced modulator block, at TP15 and TP6noting that:
    a. The signal TP15 is the audio frequency sine wave from the audio oscillator block. This is the modulating input to the balanced modulator block.
    b. The signal at TP6 is a sine wave whose frequency is slightly less than 455KHz. It is generated by the 455 KHz oscillator block, and is the carrier input to the balanced modulator block.
6. Next, examine the output of the balanced modulator block (at TP17), together with the modulating signal at TP15 trigger the oscilloscope on the modulating signal.

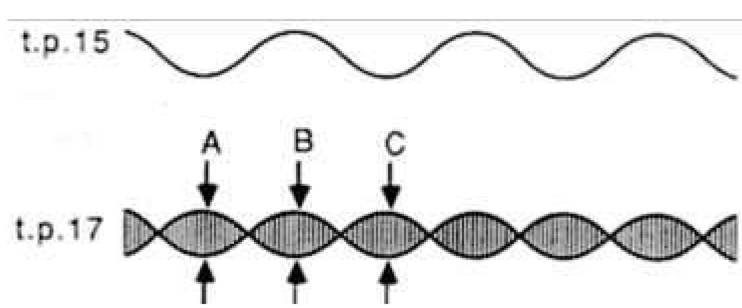


Fig.2.2

Note that it may be necessary to adjust the balanced modulator block's balance pot, in order to ensure that the peaks of TP17's waveform envelope (labeled A, B, C etc. in the above diagram) all have equal amplitude.

You will recall that the waveform at TP17 was encountered in the previous experiment this is a double-sideband suppressed carrier (DSBSC) AM waveform, and it has been obtained by amplitude-modulating the carrier sinewave at TP6 of frequency fc with the audio-frequency modulating signal at

TP15 of frequency fm, and then removing the carrier component from the resulting AM signal, by adjusting the balance pot. The frequency spectrum of this DSBSC waveform is shown in Figure 2.3.

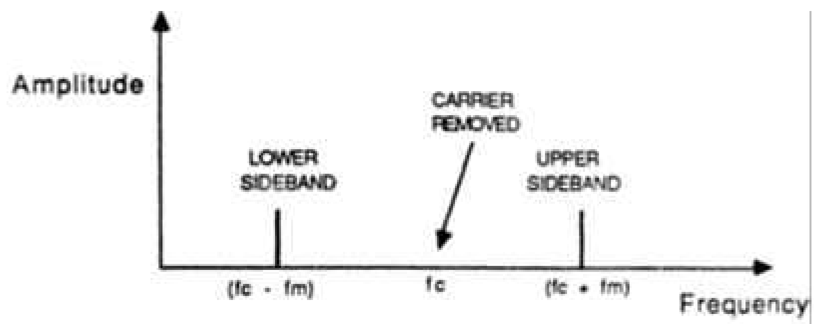


Fig.2.3

7. The DSBSC output from the balanced modulator block is next passed on to the ceramic filter block, whose purpose is to pass the upper sideband, but block the lower sideband. The upper sideband will suffer little attenuation, while the lower sideband will be heavily attenuated to such an extent that it can be ignored.
8. Monitor the output of the ceramic band pass filter block (at TP20) together with the audio modulating signal (at TP15) using the later signal to trigger the oscilloscope. Note that the envelope of the signal at TP20 now has fairly constant amplitude, as shown in Figure 3.

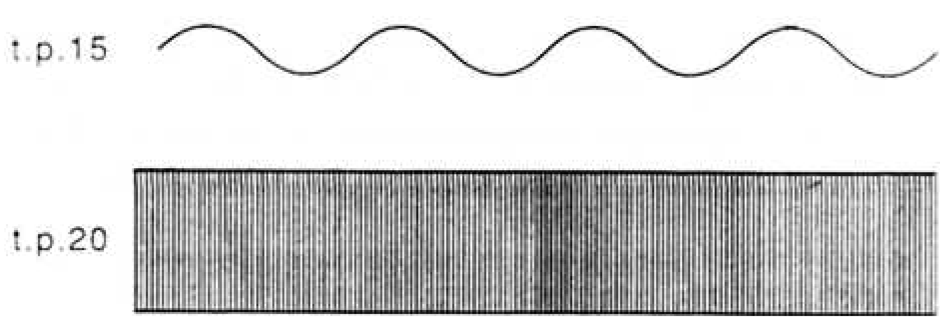


Fig. 2.4

If the amplitude of the signal at TP20 is not reasonably constant, adjust the balance pot in the balance modulator block to minimize variations in the signal's amplitude.

9. Now, trigger the oscilloscope with the ceramic band pass filter's output signal (TP20) and note that the signal is a good, clean sine wave, indicating that the filter has passed the upper sideband only.

   Next, turn the audio oscillator block's frequency pot throughout its range. Note that for most audio frequencies, the waveform is a good, clean sine wave, indicating that the lower sideband has been totally rejected by the filter. For low audio frequencies, you may notice that the monitored signal is not such a pure sinusoid. This is because the upper and lower sidebands are now very close to each other, and the filter can no longer completely remove the lower sidebands are now very close to each other, and the filter can no longer completely remove lower sideband.

   Nevertheless, the lower sideband's amplitude is sufficiently small compared with the upper sideband, that its presence can be ignored. Since the upper sideband dominates for all audio modulating frequencies, we say that single sideband (SSB) amplitude modulation has taken place.

   Note: *If the monitored waveform is not a good sine wave at higher modulating frequencies (i.e. when the frequency pot is near the MAX position), then it is likely that the frequency of the 455 KHz oscillator needs to be trimmed.*

10. Note that there is some variation in the amplitude of the signal at the filter's output (TP20) as the modulating frequency changes. This variation is due to the frequency response of the ceramic band pass filter, and is best explained by considering the spectrum of the filter's input signal at the MIN and MAX positions of the frequency pot, as shown in Figure 2.5 and Figure 2.6 respectively.

    a. **Modulating frequency f_m = 300Hz (pot in MIN position)**

.

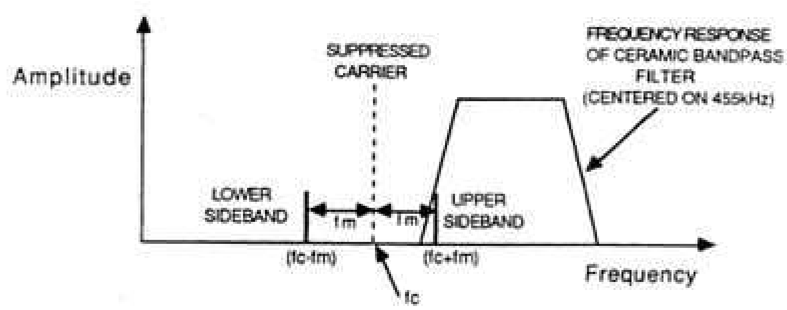


Fig.2.5

    b. **Modulating frequency f_m = 3.4 KHz (pot in MAX position)**

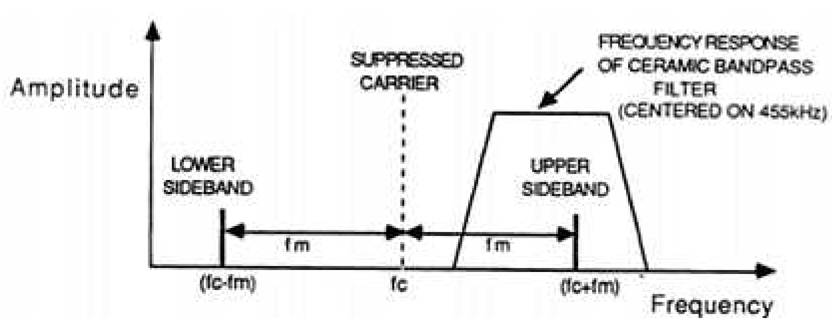


Fig.2.6

11. We have used a ceramic band pass filter to pass the wanted upper sideband, but reject the unwanted lower sideband which was also produced by the amplitude modulation process. We used this type of filter because it passes the upper sideband, yet has a sufficiently sharp response to strongly attenuate the lower sideband, which is close by. However, there is a disadvantage of this type of filter is the range of frequencies that the filter will pass is fixed during the filter's manufacture, and cannot subsequently be altered. The particular filter we are using has a pass band centered on 455 KHz, and this is why we have arranged for the wanted upper sideband to also be at about 455 KHz. As we will see in later experiments, the **ST2201/ST2202** receiver will accept audio frequency signals in the AM broadcast band, i.e. signals which fall in the frequency range of 525 KHz. However, since the SSB output from the ceramic band pass filter occupies a narrow band of frequencies around 455 KHz, it is not suitable for direct transmission to the receiver.
To overcome the problem, this narrow band of frequencies must be shifted upso that it falls within the AM broad cast band. This frequency-shifting operationis performed by the balanced modulator & band pass filter circuit 2, block, which contains a balanced modulator followed by a tuned circuit.

The operation is performed in two stages:

1. By amplitude-modulating at 1MHz carrier sinewave with the output from the ceramic band pass filter, and 'balancing out' the carrier component.
   This is shown in Figure 2.7&2.8
   a. Spectrum of output from ceramic band pass filter block

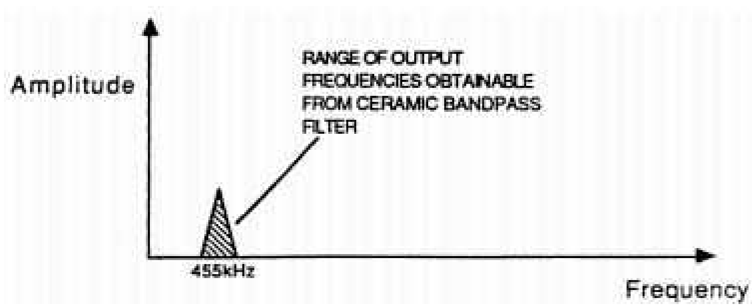


Fig.2.7

   b. Spectrum obtained by modulating 1MHz carrier with output from ceramic band pass filter.

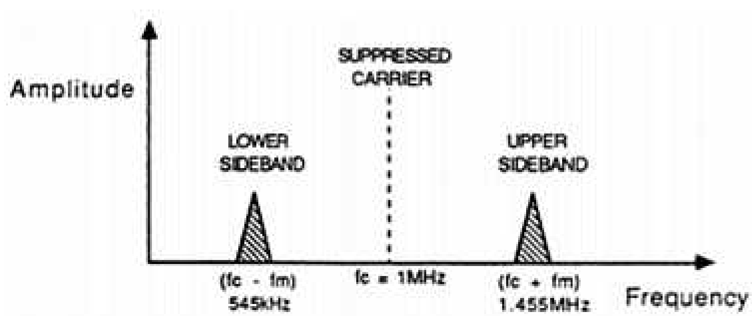


Fig.2.8

2. By passing the Upper Side, and blocking the Lower Sideband, using a tuned circuit band pass filter, as shown in Figure 2.9&2.10.

a. **Rejection of lower side band with tuned circuit band pass filter.**

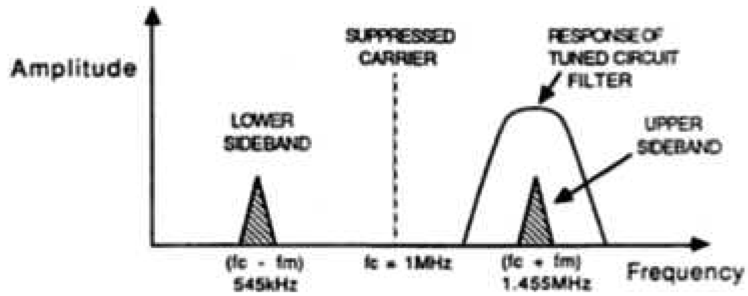


Fig.2.9

b. **Final SSB output from balanced modulator and band pass filter circuit2.**

Fig.2.10

Note that since there is a large gap between the upper and lower sidebands (a gap of about 910 KHz), a band pass filter with a very sharp response is not needed to reject the lower sideband, a simple tuned circuit band pass filter is quite sufficient.

12. Now examine the output of the balanced modulator & band pass filter circuit 2 block (TP22), and check that the waveform is a good sine wave of frequency approximately 1.45MHz.

SINGLE SIDEBAND AM RECEPTION (DEMODULATION)

Procedure:

This experiment investigates the reception and demodulation of the single sideband amplitude modulated waveforms generated by **ST2201**, using the **ST2202** receiver module.

1. Position the ST2201 &ST2202 modules, with the ST2201 board the left, and a gap of about three inches between them.
2. Ensure that the following initial conditions exist on the ST2201 board.
   a. Audio oscillator's amplitude pot in full clockwise position.
   b. Audio input select switch in INT position.
   c. Mode switch in SSB position.
   d. Output amplifier's gain pot in full clockwise position.
   e. TX output select switch in ANT position.
   f. Audio amplifier's volume pot in full counter-clockwise position.
   g. Speaker switch in ON position.
   h. On board antenna in vertical position, and fully extended.
3. Ensure that the following initial conditions exist on the ST2202 board.
   a. RX input select switch in ANT position.
   b. R.F amplifier's tuned circuit select switch in INT position.
   c. R.F amplifier's gain pot in full clockwise position.
   d. AGC switch in out position.
   e. Detector switch in product position.
   f. Amplifier's volume pot in fully counter clockwise position.
   g. Speaker switch in 'ON' position.
   h. Beat frequency oscillator switch in 'ON' position.
   i. On - board antenna in vertical position, and fully extended.
4. Turn on power to the modules.
5. On the ST2201 module, examine the transmitter's output signal (TP13), and make sure that this is a good SSB waveform, by checking that the signal is a reasonably good sine wave.

   Note: *The amplitude of the transmitter's output signal will change as the pot is tuned; also, the monitored sine wave may be slightly less pure at low modulating frequencies. These characteristics are due to the fact that the ceramic band pass filter is not a perfect filter, and they will have negligible effect on the quality of the Receiver audio output.*

   If the monitored waveform is not a good sine wave at higher modulating frequencies i.e. when the frequency pot is approximately in centre, try adjusting the balance pots in the following two blocks, in order to ensure that the 455 KHz and 1 MHz carrier components have been completely balanced out.
   a. Balanced modulator block, and
   b. Balanced modulator & band pass circuit 2 block.

If the waveform at TP13 is still not good sine wave at higher modulating frequencies, then if it is likely that the frequency of **ST2202**'s 455 KHz oscillator block needs adjusting.

6. Turn ST2201's amplitude pot (in the audio oscillator block) to its full counter clockwise (minimum amplitude) position and note that amplitude of the monitored output signal from ST2201 (at TP13) drops to zero. This illustrates that the SSB waveform contains no carrier - if the amplitude of the modulating audio signal drops to zero, so does the amplitude of the transmitted SSB signal. In ST2201's audio oscillator block, return the amplitude pot to its fully clockwise (MAX) position, and put the frequency pot in its midway position.
7. We will now transmit the SSB waveform to the ST2202 receiver.
   Since ST2201's TX output select switch is in the ANT position, the SSB signal at TP13 is fed to the transmitter's antenna. Prove this by touching ST2201's antenna, and noting that the loading caused by your hand reduces the amplitude of the SSB waveform at TP13. The antenna will propagate this SSB waveform over a maximum distance of about 1.4 ft. We will now attempt to receive the propagated SSB waveform with the ST2202 board, by using the receiver's on board antenna.
   Note: *If more than one* **ST2201** *transmitter/receiver system is in use at one time, it is possible that there may be interference between nearby transmitters if antenna propagation is used. To eliminate this problem,* ***<u>use a cable between each transmitter/receiver pair</u>****, connecting it between* **ST2201***'s TX output socket and* **ST2202***'s RX input socket. If you do this, make sure that the transmitter's TX output select switch, and the receiver's RX input select switch, are both in the SKT position, then follow the steps below as though antenna propagation were being used.*
8. On the ST2202 module, monitor the output of the IF amplifier 2 block (TP28) and turn the tuning dial until the amplitude of the monitored signal is at its greatest. Check that you have tuned into the SSB signal, by turning ST2201's amplitude pot (in the audio oscillator block) to its MIN position, and checking that the monitored signal amplitude drops to zero. (This should occur at about 85-95) Return the amplitude pot to its MAX position.
9. Since the incoming SSB signal contains no carrier component, the receiver's AGC circuit cannot make use of incoming carrier amplitude, in order to control the receiver's gain. This means that the receiver's AGC circuit cannot be used for SSB reception, and must be switched off.
   Consequently, it is very important to avoid overloading the receiver by transmitting an SSB signal which is too large for the receiver to handle. To ensure that overloading does not occur.
   a. Turn the gain pot, in **ST2201**'s output amplifier block, so that the pots arrowhead is horizontal, and pointing to the left. This ensures that the amplitude of the transmitted SSB signal is small.
   b. On the **ST2202** module, fine tune the tuning dial until the amplitude of monitored signal (at TP28) is at its greatest.

c. Adjust the gain pot, in **ST2202**'s RF amplifier block, until the amplitude of the monitored signal is about 2 volts pk/pk.

d. Repeat steps (2) and (3).

There should now be no risk of the **ST2202** receiver overloading.

10. The receiver's beat frequency oscillator (BFO) produces a sine wave at the IF frequency of 455 KHz. This 455 KHz sine wave is input to the receiver's product detector block, where it is mixed with the SSB from I.F. amplifier.

The actual frequency of the output signal from I.F. amplifier 2 will lie within a limited range of frequencies, which lie in the region of 455 KHz. The output signal can be varied over this limited range of frequencies, by adjusting the frequency of the transmitter's modulating signal from center (slightly lower and slightly higher).

In addition, the position of the limited range of frequencies from IF amplifier 2 will depend on the exact frequency of the receiver local oscillator output. If the oscillator's frequency is varied slightly from its present frequency, this range of frequencies can be moved both above, and below, 455 KHz.

The product detector block mixes the output from the BFO with the output from I.F. amplifier block mixing process results in the generation of two new frequency components.

- Component whose frequency is the sum of the two input frequencies.
- Component whose frequency is the difference between the two input frequencies.

A low-pass filter at the output of the product detector rejects all frequencies except the difference frequency. Consequently, any slight difference in frequency between the BFO's output and I.F. amplifier 2's output will result in and audio frequency at the product detector's output. This audio frequency is then converted into sound by the receiver's audio amplifier block.

To demodulate out incoming SSB signal, we tune the Receiver's local oscillator so that the output frequency range form IF amplifier 2 is slightly below the 455KHz. BFO frequency, such that the difference frequency generated by the product detector is the same as the original transmitter audio modulating frequency. Then, as the frequency of the transmitter's modulating signal changes, the output from the product detector should follow it.

11. Monitor the output of ST2202's beat frequency oscillator block (TP50), and note that this carries a sine wave of 455 KHz.

On the **ST220l/2** receiver, adjust the volume pot so that the receiver's output is clearly audible.

Note: If desired, headphones may be used instead of the on board loudspeaker. To use the headphones, simply plug the headphone jack into the audio amplifier block's headphones socket, and put the speaker switch in the OFF position. The volume from the headphones is still controlled by the block's volume pot.

Slowly turn the tuning dial, and notice that the tone at the receiver's output changes. This is because the frequency of the output signal from IF amplifier 2 changes as the dial is turned. Product detector's output, as the tuning dial is turned.

On the **ST2201** module, turn the volume pot (in the audio amplifier block) clock wise, until you can hear the tone of the audio oscillator's output signal, in addition to the tone from the **ST2202** board. With the receiver's tuning dial on the counter-clockwise side of the minimum frequency position (i.e. using dial positions lower than the minimum frequency position), find the position where the two tones are approximately the same.

Then, turn the frequency pot in **ST2201'**s audio oscillator block, throughout its range, noting that the frequency of the tone generated by **ST2202** remains close to that generated by **ST2201** for all pot positions.

Demodulation of the SSB signal has now been achieved, so the volume pot in the transmitter's audio amplifier block can now be returned to its full counter clockwise(minimum) position.

Note: If the tuning dial is tuned on the clockwise side of the minimum frequency position, rather than the counter-clockwise side, a position will stillbe found where the transmitter and receiver tones are approximately the same. However, if the transmitter's audio frequency is then increased, the receiver's audio frequency will decrease, and vice-versa. The reason for this is that the frequency of IF amplifier 2's output is now above the BFO frequency, instead of below it, converting all high frequency components in the transmitter's modulating waveform into low-frequency components, and vice-versa.

Consequently, SSB demodulation is not achieved with tuning dial on the clockwise side of the minimum frequency position.

12. On the ST2202 module, monitor the output of the product detector block (at TP37), together with the output of the audio amplifier block (TP39), triggering the scope with the later signal.

Note: *There will be no signal at TP39 if the audio amplifier's volume pot is in its fully counter-clockwise (minimum) position. Vary the frequency of the Transmitter's audio modulating signal by adjusting the audio oscillator's frequency pot on the* **ST2201** *module.*

Note: *There will be no signal at TP39 if the audio amplifier's volume pot is infully counter-clockwise (minimum) position. Also, try briefly reducing the amplitude of the Transmitter's modulating signal to zero (by turning the audio oscillator's amplitude pot fully clockwise), and note that the receiver's output amplitude also drops to zero.*

13. With the receiver's tuning dial adjusted for correct demodulation of the transmitted SSB signal, you may notice that there is a slight drift in the tone generated by the Receiver oscillator circuits, leading to changes in the difference frequency produced by the product detector. Oscillator drift is a serious problem in SSB communication, since it shifts all the frequency components which make up the Receiver's audio output signal, by the same amount. If we try to use our SSB communications system to transmit music, then oscillator drift will cause the harmonic relationship between notes to be lost.

This makes SSB useless for transmitting music.

Experimental Data:

Table 1

Sl. No.	Modulating Signal		Carrier Signal		Demodulated Signal	
	Freq. (f_m)	Ampl. (A_m)	Freq. (f_c)	Ampl. (A_c)	Freq. (f_D)	Ampl. (A_D)

Linearity:

Table 2

Sl. No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_{o1}	A_{o1}	M_1
2.	f_1	A_2	f_{o1}	A_{o2}	M_2
3.	f_1	A_2	f_{o1}	A_{03}	M_3
n.	f_1	A_n	f_{o1}	A_{on}	M_n

Table 3

Sl. No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_{o1}	A_{o1}	M_1
2.	f_2	A_1	f_{o2}	A_{o2}	M_2
3.	f_3	A_1	f_{o3}	A_{03}	M_3
n.	f_n	A_1	f_{on}	A_{on}	M_n

Data Analysis:

1. Calculate the ratio of amplitudes of input and output signals and show the linearity of LPF.
2. Find the cut off frequencies of BPF. Tally the frequency of input and output signal of modulator and demodulator.

Discussion:

Write briefly your comments about the above experiment.

Precautionary Measure to be taken:

1. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
2. For connecting any signal from one equipment to other equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.
3. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.
4. For unusual spark or burning smell, immediately switch off the main supply to the kit.
5. Ensure that, a signal is connected to an appropriate junction destined for it.
6. Always cover the equipment for dust protection after the experiment.

Some Sample questions:

1. Design a circuit to get SSB-SC signal from a DSB-SC signal for sinusoidal modulating signal. Draw a U.S.B and L.S.B. waveform for a fixed sinusoidal modulating signal of frequency 2 KHz. Indicate the center frequency and 3 Db band width of the B.P.F. of your system.

2. Design a circuit to get SSB-SC taking sinusoidal input signal amplitude 2v (p-p). Decrease the amplitude of input signal in steps of 0.5 volts and draw the modulated output signal.

EXPERIMENT # 3

Title: STUDY OF SUPER HETERODYNE RECEIVER

Objective:

To plot

- Selectivity curve for radio receiver
- Sensitivity curve for radio receiver
- Fidelity curve for radio receiver

Theory:

The important characteristics of receivers are sensitivity, selectivity, & fidelity described as follows:

Sensitivity:

The sensitivity of radio receiver is that characteristic which determines the minimum strength of signal input capable of causing a desired value of signal output.Therefore, expressing in terms of voltage or power, sensitivity can be defined as the minimum voltage or power at the receiver input for causing a standard output.
In case of amplitude-modulation broadcast receivers, the definition of sensitivity has been standardized as "amplitude of carrier voltage modulated 30% at 400 cycles, which when applied to the receiver input terminals through a standard dummy antenna will develop an output of 0.5 watt in a resistance load of appropriate value substituted for the loud speaker".

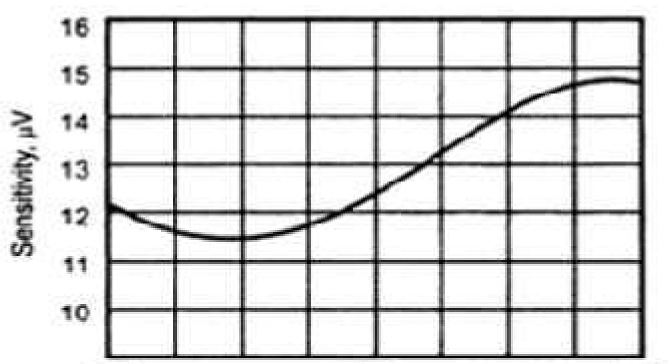


Fig. 3.1

Sensitivity is a determined by impressing different RF voltages in series with a standard dummy antenna and adjusting the intensity of input voltage until standard outputs obtained at resonance for various carrier frequencies. Sensitivity is expressed in microvolt.

Selectivity:

The selectivity of a radio receiver is that characteristic which determines the extent to which it is capable of differentiating between the desired signal and signal of other frequencies.
Selectivity is expressed in the form of a curve that give the carrier signal strength with standard modulation that is required to produce the standard test output plotted as a function off resonance of the test signal.

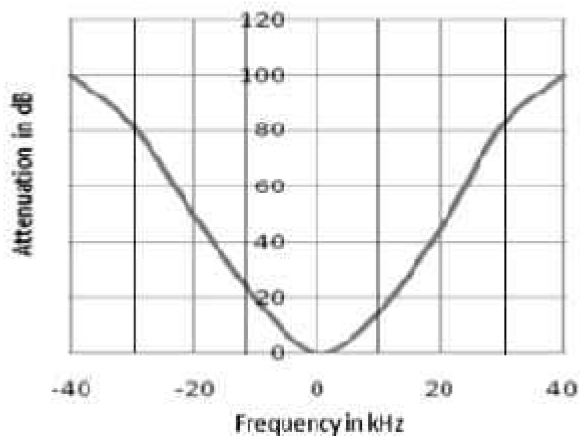


Fig. 3.2

The receiver is tuned to the desired frequency and manual volume control is set for maximum value. At standard modulation, the signal generator is set at the resonant frequency of the receiver. The carrier output of the signal generatoris varied until the standard test output is obtained. At the same tuning of receiver, the frequency of signal generator is varied above and below the frequency to which the receiver is tuned. For every frequency, the signal generator voltage, applied to the receiver input, is adjusted to give the standard test output from the receiver.

Fidelity:

This is defined as the degree with which a system accurately reproduces at its output the essential characteristics of signals which is impressed upon its input.

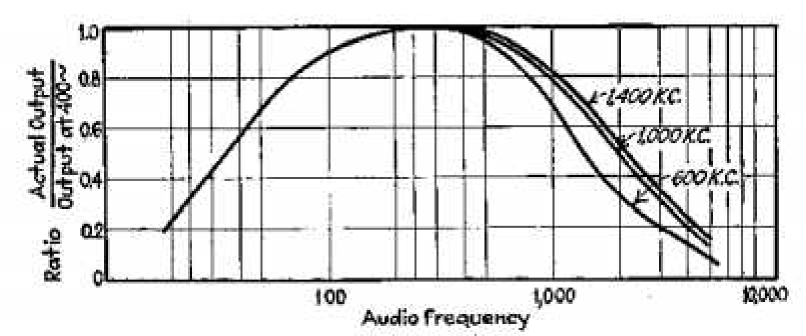


Fig. 3.3

Fidelity is the term expressing the behavior of receiver output with modulation frequency of inputvoltage. To obtain a fidelity curve, the carrier frequency of the signal generator adjusted to resonance with the receiver, standard 400 cycles modulation is applied, thesignal generator carrier level is set at a convenient arbitrary level and the manual volume control of the receiver is adjusted to give the standard test output. The modulation frequency is then varied over the audio range, keeping degree of modulation constant. A graph is then plotted in the ratio of actual output in volts to the output at 400 c/s against modulation frequency.

Apparatus Required:

- Amplitude Mod./Demod. Experiment Kit.
- Function Generator.
- A CRO / DSO

Block Diagram:

Modulation Section-

Fig. 3.4

Demodulation Section-

Fig. 3.5

SELECTIVITY

Procedure:

1. Setting on ST2202
   a. Set the detector in diode mode.
   b. AGC on.
   c. Set the volume control full clockwise.
2. Apply AM signal with 400 Hz modulating frequency and 30% modulation taken from AM generator into Rx input socket.
3. Set the input carrier frequency to suitable value that lies within the AM band (525 KHz - 1600 KHz). Al so set signal level to 100mV.
4. Tune the Receiver using tuning control. Also adjust gain potentiometer provided in R. F. amplifier section of ST2202 so as to get unclipped demodulated signal at detector's output (output of audio amplifier).
5. Note the voltage level at receiver's final output stage i .e. audio amplifier's output on CRO (voltage at resonance (Vr)).

6. Now gradually offset the carrier frequency in suitable steps of 5 KHz or 10 KHz below and above the frequency adjusted in step 2 without changing the tuning of receiver while maintaining the input signal level.
7. Now record the signal level at output of audio amplifier for different input carrier frequency, on CRO (i .e. voltage off resonance (Vi).

8. Tabulate the readings as under:

Carrier Frequency	Output Voltage	Ratio = 20 log (Vi / Vr) dB

9. Plot the curve between ratio and carrier frequency.

SENSITIVITY

1. Setting on ST2202
   a. Set the detector in diode mode.
   b. AGC on.
   c. Set the volume control full clockwise.
2. Apply AM signal with 400 Hz modulating frequency and 30% modulation taken from AM generator into Rx input socket.
3. Set the input carrier frequency to suitable value that lies within the AM band (525 KHz - 1600 KHz). Also tune the detector to that carrier frequency using tuning control. (You will hear atone).
4. Set the input AM level to 100mV. Also adjust the gain potentiometer provided in R. F. amplifier section of ST2202 so as to get unclipped demodulated signal at detectors output.
5. Record input carrier frequency & signal level at the final output stage i .e. output of audio amplifier (observed on CRO).
6. Change the input carrier frequency & also tune the receiver to that frequency & repeat step 4.
7. Tabulate the collected readings as under:

Carrier frequency	Output (p-p)

8. Plot the graph between carrier frequency & output level.

FIDELITY

1. Setting on ST2202
   d. Set the detector in diode mode.
   e. AGC on.
   f. Set the volume control full clockwise.
2. Apply AM signal with 400 Hz modulating frequency and 30% modulation taken from AM generator into Rx input socket.
3. Select a suitable carrier frequency that lies within AM Band (525 KHz - 1600 KHz). Tune the ST2202 receiver to that frequency using tuning control. Also adjust gain potentiometer provided in R. F. amplifier section so as to get unclipped demodulated signal at detector's output.
4. Note the demodulated signal level (Vr) at the final output stage i .e. output of audio amplifier (on CRO) for the applied AM signal with 400Hz modulating signal.
5. Now vary the modulating signal frequency over audio range (300 Hz-3 KHz) in suitable steps say 100Hz. Note the corresponding output level (Vi) at the output of audio amplifier (on CRO).
6. Tabulate readings as under :

Carrier frequency	Modulating frequency	Output Voltage

7. Plot the graph between modulating frequency and relative response.

Relative response = 20 log (Vi / Vr) dB

ACADEMIKA:
ACL-AM

Apparatus Required:

- Modules ACL-AM & ACL-AD Experiment Kit.
- Power Supply GND, +5V, +/- 12V
- A CRO / DSO
- Connecting Links
- Frequency counter

Block Diagram:

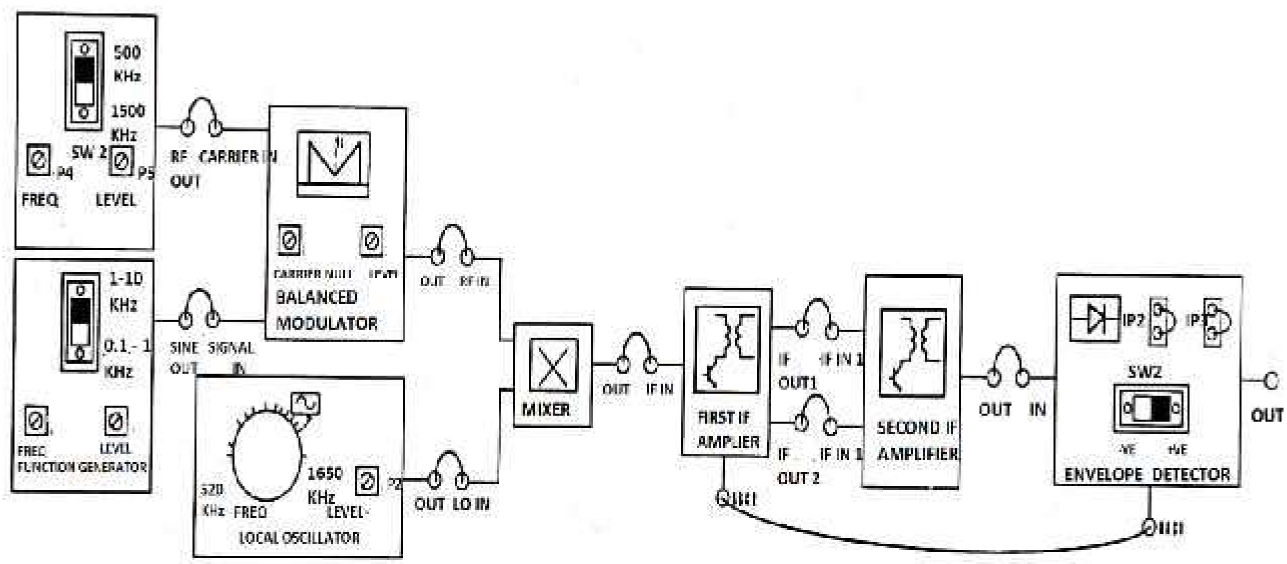


Fig. 3.6

A1. SENSITIVITY OF A RADIO RECEIVER VIA CABLE

Procedure:

1. RefertotheFIG.3.6 &Carryoutthefollowingconnections.
2. Connect o/p of FUNCTION GENERATOR section (ACL-AM) OUT post to the i/pof Balance Modulator(ACL-AM)SIGNALINpost.
3. Connect o/p of VCO (ACl-AM) OUT post to the input of Balance modulator (ACL-AM) CARRIERIN post.
4. Connect the power supply with proper polarity to the kit ACL-AM &ACL-AD, While connecting this, ensure that the power supply is OFF.
5. Switch on the power supply and Carry out the following presetting:

- FUNCTION GENERATOR: sine LEVEL about0.5VVpp;FREQ.about1kHz
- VCO:LEVEL about 2Vpp;FREQ.about600kHz,Switchon1500KHz

- BALANCED MODULATOR1:CARRIER NULL completely rotates Clockwise or counter clock wise, so that the modulator is "unbalanced" and an AM Signal with not suppressed carrier is obtained across the output: adjust OUTLEVEL to obtain an AM signal across the output which amplitude isabout200mVpp.
- LOCAL OSCILLATOR (ACL-AD):1050KHz,2V.

6. Connect local oscillator OUT post to LO IN of the mixer section.
7. Connect balance modulator1 out to RF IN of mixer section in ACL-AD.
8. ConnectmixerOUTtoIFINof1st IFAMPLIFIER in ACL-AD.
9. Connect IF OUT 1 of 1st IF to IFIN1 and IFOUT2 of 1stIF to IFIN2 of 2ND IF AMPLIFIER.

- Connect OUT post of 2nd IF amplifier to IN post of envelope detector.
- Connect post AGC 1 to post AGC2 and jumper position as per diagram.
- Observe the modulated signal envelope, which corresponds to the waveform of the modulating signal at OUT post of the balanced modulator 1 of ACL-AM. Connect the oscilloscope to the IN and OUT post of ENVELOPE DETECTOR and detect the AM signal.

10. Check that the detected signal follows the behavior of the AM signal envelope. Vary the amplitude of the Balanced modulator output and check the corresponding variations at the demodulated signal.
11. Adjust the inputtoRFINpostbyvaryingtheoutputofBM1insuchawaythatyou should get minimum detected output of about 0.3Volt at the output of Envelope detector.
12. You can take the readings as per the table mentioned below for various carrier frequencies and corresponding Local Oscillator frequency settings.

Experimental Data:

CARRIER FREQ.(KHz)	600	700	800	900	1000	1100	1200	1300
LOCAL OSC.FREQ.(KHz)	1050	1150	1250	1350	1450	1550	1650	1750
OUTPUT OF MIXER								

Data Analysis:

1. Plot the graph of output of BM1 on Y- axis and local oscillator frequency on X-axis(Fig. 3.7)

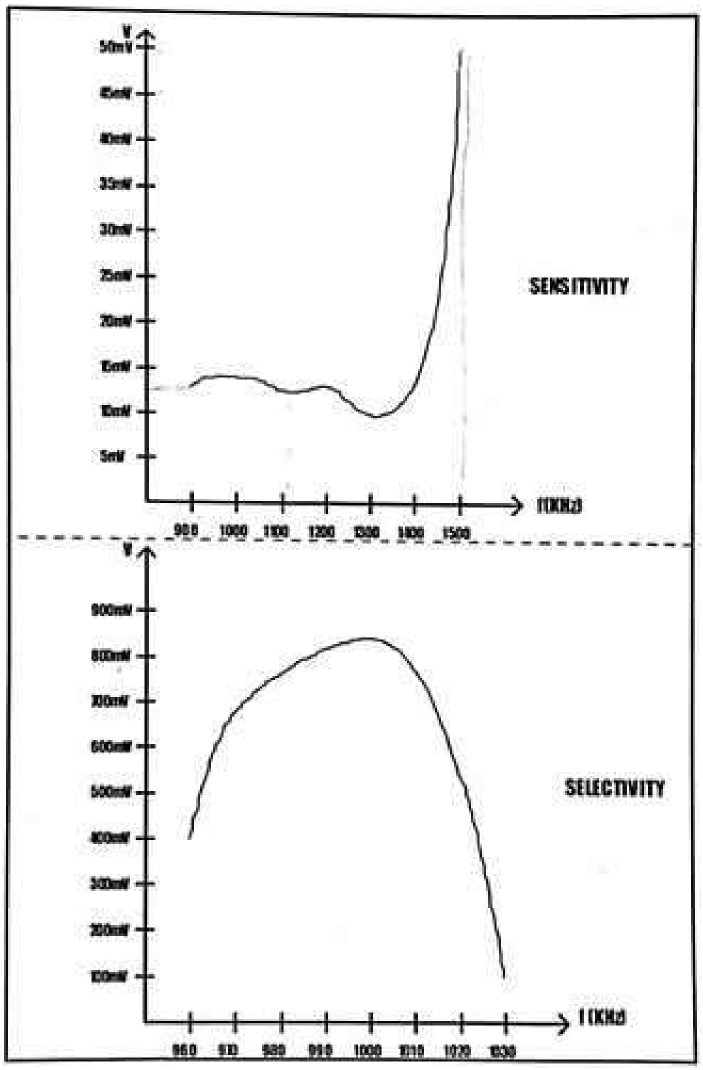


Fig. 3.7

B1. STUDY OF SELECTIVITY OF A RADIO RECEIVER VIA CABLE

Apparatus Required:

- Modules ACL-AM&ACL-AD

- Power supply
- Oscilloscope.
- Connecting Links
- Frequency counter

Procedure:

NOTE: KEEP ALL THE SWITCH FAULTS IN OFFPOSITION.

1. RefertotheFIG.3.1&Carryoutthefollowingconnections.
2. Connect o/p of FUNCTIONGENERATOR section(ACL-AM)OUTposttothei/pofBalanceModulator(ACL-AM)SIGNALINpost.
3. Connect *o/p* of VCO (ACL-AM) OUT post to the input of Balance modulator (ACL-AM) CARRIER IN post.
4. Connect the power supply with proper polarity to the kit ACL-AM&ACL-AD, while connecting this, ensure that the power supply is OFF.
5. Switch on the power supply and Carry out the following presetting:
    - FUNCTIONGENERATOR: sine LEVEL about 0.5Vpp;FREQ.About1KHz
    - VCO: LEVEL about2Vpp;FREQ.about850kHz, Switch on1500KHz
    - BALANCED MODULATOR1: CARRIER NULL completely rotates clockwise or counters clockwise, so thatthemodulatoris"unbalanced"andanAMsignalwithnotsuppressedcarrierisobtainedacrossthe output:adjustOUTLEVELtoobtainanAMsignalacrosstheoutputwhichamplitudeisabout50mVpp.
    - LOCALOSCILLATOR (ACL-AD):1300KHz,2V.
6. Connect local oscillator OUT post to LO IN of the mixer section.
7. Connectbalancemodulator1outtoRFIN of mixer section in ACL-AD.
8. ConnectmixerOUTtoIFINof1st IF AMPLIFIER in ACL-AD.
9. Connect IF OUT1 of 1^{st}IF to IF IN 1 and IFOUT 2 of 1^{st}IF to IF IN 2 of 2^{ND}IFAMPLIFIER.
10. Connect OUT post of 2^{ND} IF amplifier to IN post of envelope detector.
11. ConnectpostAGC1topostAGC2andjumperpositionasperdiagram.
12. Observe the modulated signal envelope ,which corresponds to the waveform of the modulating signal at OUT post of the balanced modulator of ACL-AM Connect the oscilloscope to the IN and OUT post of envelope DETECTOR and detect the AM signal.
13. Check that thedetected signal follows the behavior ofthe AM signal envelope

Measure the detected signal amplitude.

14. Vary the Carrier frequency of (ACL-AM) in terms of 10 KHz on both sides from860KHzto940 KHz.
15. Take readings of Amplitude levels at Envelope Detector output for various Carrier frequencies.

Experimental Data:

Table similar to the one given below can be observed at the users end

Carrier Freq.(KHz)	Envelope Detector Output(mV)
860	6
870	40
880	120
890	350
900	500
910	380
920	150
930	80
940	20

Carrier Freq.(KHz)	Envelope Detector Output(mV)

Plotthegraphaccordinglyaspertheareadingstaken.Forthecarrierfrequencyand *o/p* Amplitude (FIG.3.7).

A2. STUDY OF SENSITIVITY OF A RADIO RECEIVER VIA ANTENNA

Apparatus Required:

- Modules ACL-AM&ACL-AD
- Power supply
- 20MHzOscilloscope.
- Connecting Links.
- Frequency counter.

Block Diagram for Study of Sensitivity and Selectivity:

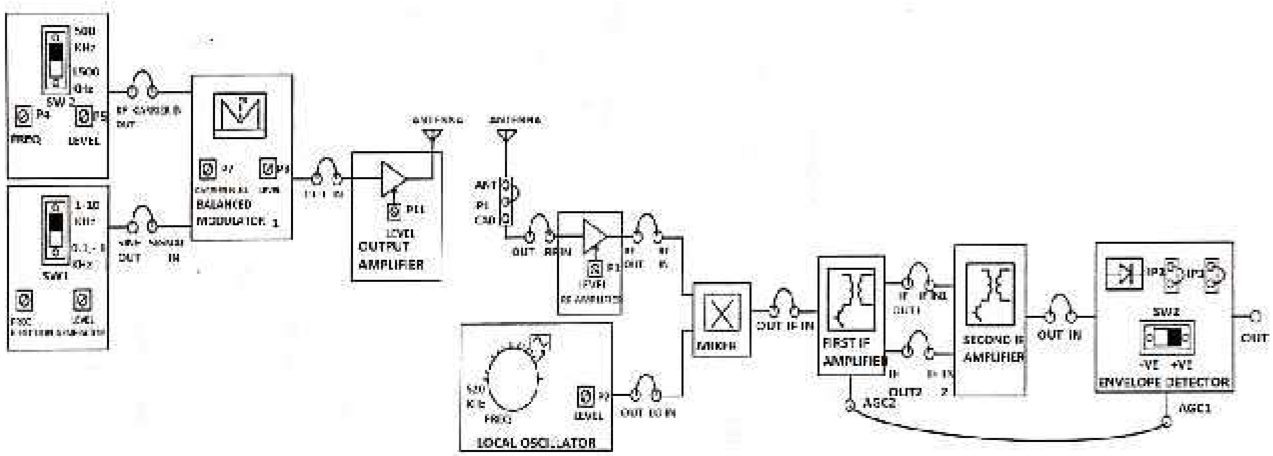


Fig.3.8

NOTE: KEEP All THE SWITCH FAULTS IN OFF POSITION.

Procedure:-

1. RefertotheFIG.3.8 & Carry out the following connections.
2. Connect o/p of FUNCTION GENERATOR section (ACL-AM) OUT post to the I/P of Balance Modulator1(ACL-AM) SIGNAL IN post.
3. Connect o/p of VCO (ACL-AM) OUT post to the input of Balanced modulator1 (ACL-AM)CARRIER IN post.
4. Connect the power supply with proper polarity to the kit ACL-AM&ACL-AD, while connecting this, ensure that the power supply is OFF.
5. Switch on the power supply and Carry out the following presetting:
   - FUNCTIONGENERATOR: sineLEVELabout0.5VVp-p; FREQ. About 1 kHz
   - VCO: Level about 2Vpp;FREQ.about850kHz,Switchon1500KHz

- BALANCED MODUlATOR1: CARRIER NULL completely rotates clockwise or counter clockwise, so that the modulator is" unbalanced" and an AM signal with not suppressed carrier is obtained across the output: adjust OUTLEVEL to obtain an AM signal across the output which amplitude isabout500mVpp.
- OUTPUT AMPLIFIER(ACl-AM):Max.(Fully Clockwise Position.) SwitchSW3onDSBPosition
- LOCAL OSCILLATOR(ACl-AD): About1300KHz,2V.
- RF LEVEL (ACl-AD): on max. Position or adjust to achieve as per input signal

6. ConnecttheOUTpostofbalancemodulator1totheINpostofoutputamplifier, in which OUT post of amplifier is directly connected to the antenna.
7. Connect local oscillator OUT post to LO IN of the mixer section.
8. Connect RF OUT post to RF IN of MIXER of ACL-AD.
9. ConnectmixerOUTtoIFINof1stIFAMPLIFIERinACL-AD.
10. Connect IFOUT1of1stIFtoIFIN1andIFOUT2of1stIFtoIFIN2of2ND IF AMPLIFIER.
11. Connect OUT postof2ND IF amplifier to IN post of envelope detector.
12. ConnectpostAGC1topostAGC2andjumperpositionasperdiagram.
13. Observe the modulated signal envelope, which corresponds to the waveform of the modulating signal at OUT post of the balanced modulator of ACL-AM. Connect the oscilloscope to the IN and OUT post of envelope DETECTOR and detect the AM signal.
14. CheckthatthedetectedsignalfollowsthebehavioroftheAMsignalenvelope.Vary the amplitude of the balanced modulator output and checks the corresponding variations at the demodulated signal.
15. Adjust the output LEVEL OFBM1insuchawaythatyoushouldgetminimum detectedoutputofabout0.3VoltattheoutputofEnvelopedetector.

You can take the readings as per the table mentioned below for various carrier frequencies and corresponding Local Oscillator frequency settings.

Experimental Data:

Carrier Freq.(KHz)	700	800	900	1000	1100	1200	1300	1400
Local Osc. Freq.(KHz)	1155	1255	1355	1455	1555	1655	1755	1855
Output of BM1(mV)								

Carrier Freq.(KHz)								
Local Osc. Freq.(KHz)								
Output of BM1(mV)								

Plot The Graph of Output of BM1 on Y-Axis And Local Oscillator Frequency On X-Axis(Fig.3.9)

B2. STUDY OF SELECTIVITY OF A RADIO RECEIVER VIA ANTENNA

Apparatus Required:

- Modules ACL-AM&AC-AD.
- Power supply
- Oscilloscope.
- Connecting Links.
- Frequency counter.

Procedure:

1. RefertotheFig.3.8 & Carry out the following connections.
2. Connect O/P of FUNCTION GENERATOR section (ACL AM)OUT post to the i/p ofBalanceModulator1(ACL-AM) SIGNAL IN post.
3. Connect o/p of VCO (ACL-AM) OUT post to the input of Balance modulator1 (ACL-AM)CARRIER IN post.
4. Connect the power supply with proper polarity to the kit ACL-AM&ACL-AD, while connecting this, and ensure that the power supply is OFF.
5. Switch on the power supply and Carry out the following presetting:
   - FUNCTION GENERATOR: sine LEVEL about 0.5Vpp; FREO. About 1 kHz
   - VCO: LEVEL about1Vpp; FREQ. about 850kHz, Switch on 1500 KHz
   - BALANCED MODULATOR1: CARRIER NULL completely rotates Clockwise or counter clockwise so that the modulator is "unbalanced" and an AM signal with not suppressed carrier is obtained across the output: adjust OUTLEVEL to obtain an AM signal across the output which amplitude is about 400mVpp.
   - OUTPUT AMPLIFIER (ACL-AM): Max. Fully Clockwise Position.)SwitchSW3onDSB Position
   - LOCAL OSCILLATOR (ACL-AD):1300 KHz, 1V.
   - RF LEVEL (ACL-AD): on max. Position or adjust to achieve as per input signal.
6. Connect the OUT post of balance modulator1 to the IN post of output amplifier. In which output of amplifier is directly connected to the antenna.
7. Connect local oscillator OUT post to LO IN of the mixer section.

8. Connect RF OUT post to RFIN of MIXER of ACL-AD.
9. Connect mixer OUTtoIFINof1st IF AMPLIFIER in ACL-AD.
10. Connect IFOUT1 of 1st IF to IF IN 1 and IF OUT2 of 1st IF to IFIN 2 of 2NDIF AMPLIFIER.
11. Connect OUT postof2nd IF amplifier to IN post of envelope detector.
12. Connect post AGC1topost AGC2and jumper position as per diagram.
13. Observe the modulated signal envelope, which corresponds to the wave form of the modulating signal at OUT post of the balanced modulator of ACL-AM Connect the oscilloscope to the IN and OUT post of envelope DETECTOR and detect the AM signal.
14. Check that the detected signal follows the behavior of the AM signal envelope, measure the detected signal amplitude.
15. Vary the Carrier frequency of (ACL-AM) in terms of 10KHz on both sides from 860KHz to940 KHz.
16. Take readings of Amplitude levels at Envelope Detector output for various Carrier frequencies.

DATA:

Table similar to the one given below can be observed at the users end.

Carrier Freq. (KHz)	Envelope Detector (mV)
810	6
820	30
830	100
840	400
850	500
860	300
870	150
880	80
890	20

Carrier Freq. (KHz)	Envelope Detector (mV)

Plot the graph accordingly as per the readings taken. For the carrier frequency and *o/p* Amplitude (FIG. 3.9).

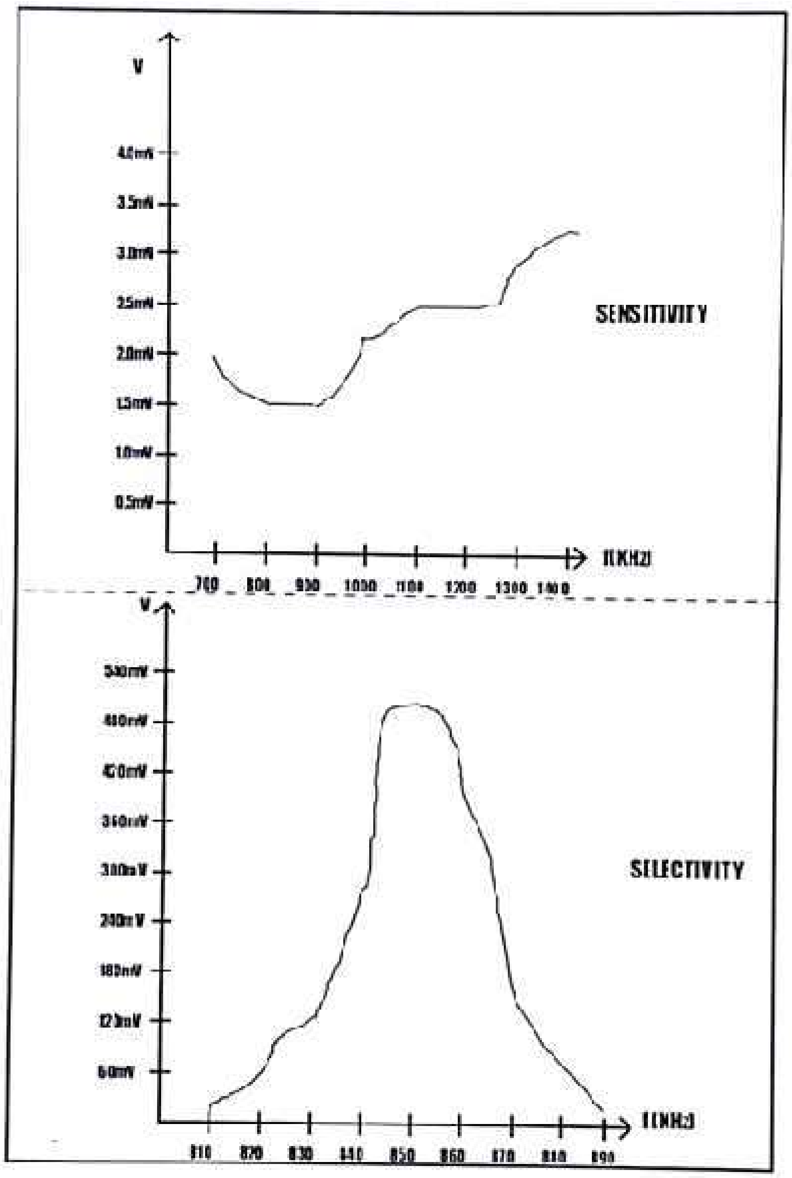


Fig 3.9

Data Analysis:

1. Plot Selectivity, Sensitivity and Fidelity curve.

Discussion:

Write briefly your comments about the above experiment.

Precautionary Measure to be taken:

1. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
2. For connecting any signal from one equipment to equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.
3. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.
4. For unusual spark or burning smell, immediately switch off the main supply to the kit.
5. Ensure that, a signal is connected to an appropriate junction destined for it.
6. Always cover the equipment for dust protection after the experiment.
7.

Some Sample questions:

1. Plot Selectivity, Sensitivity and Fidelity curve of radio receiver.

Discussion:

Write briefly your comments about the above experiment.

Precautionary Measure to be taken:

8. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
9. For connecting any signal from one equipment to equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.
10. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.

11. For unusual spark or burning smell, immediately switch off the main supply to the kit.
12. Ensure that, a signal is connected to an appropriate junction destined for it.
13. Always cover the equipment for dust protection after the experiment.

Some Sample questions:

1. Plot Selectivity, Sensitivity and Fidelity curve of radio receiver.

EXPERIMENT # 4

Title: STUDY OF FREQUENCY MODULATION & DEMODULATION

Objective:

- FM Generation (MODULATION) using Varactor, Reactance Modulator
- FM Generation (MODULATION) using VCO
- Determine Frequency Deviation and Modulation Index
- FM Reception (DEMODULATION) using different Detectors

Theory:

It is a process in which the frequency of the carrier is varied in accordance with the instantaneous value of modulating voltage.

If $V_s(t) = A_s \cos\omega_s t$ is the message signal

And $V_c(t) = A_c \cos\omega_c t$ is the carrier signal then, the modulated carrier is given by

$$V_m(t) = A_c\ (\cos \omega_c t + m_f \sin \omega_c t) \qquad (1)$$

Where, $$\boxed{m = \Delta_f / f_s} \qquad (2)$$

'm' is called the modulation index.

The amount of change in frequency is determined by the amplitude of the modulating signal. Frequency of the carrier signal increases as the amplitude of the modulating signal rises while decreases as the amplitude of the modulating voltage drops down. When the amplitude of the modulating signal is zero the carrier signal has frequency at its normal value "f_o" called resting or center frequency.

- The amounts of frequency variation (frequency deviation or shift) depend on the amplitude of the audio signal. Greater the audio signal, higher will be the frequency shift and vice versa. According to the international law regarding frequency spectrum the maximum <u>frequency deviation should be 75 KHz. The highest AF is transmitted is 15 KHz.</u>

The rate of frequency deviation depends on signal frequency. The total variation in frequency from lowest to highest called Carrier swing (CS), obviously the carrier swing =2 * frequency deviation (Δf). We know that max deviation of 75Hz is allowed for commercial FM broadcast stations in range 88-168MHz within VHF band, hence FM channel width =2 * 75 =150Hz,with a guard band of 25KHz on either side. The channel width become 2(75 +25) =200 KHz. The purpose of guard band is to prevent a signal from its adjacent channel interference.

$\Delta_f = f_{max} - f_c = f_c - f_{min} = (f_{max} - f_{min})/2 = K_f A_m$ where K_f is **constant proportionality**

Therefore $$\boxed{K_f = \Delta_f / A_m} \qquad (3)$$

Deviation Ratio: In the modulation index equation if we allow max frequency deviation and max modulating frequency then it becomes "Deviation Ratio"

Deviation ratio = (Δf) max /f_m (max).

Apparatus Required:

- Frequency Modulation/Demodulation Experiment Kit (ST2203 or ST2204).
- Function Generator.
- A CRO / DSO

ST2203

❖ Frequency Modulation using Varactor modulator: -

Block Diagram:

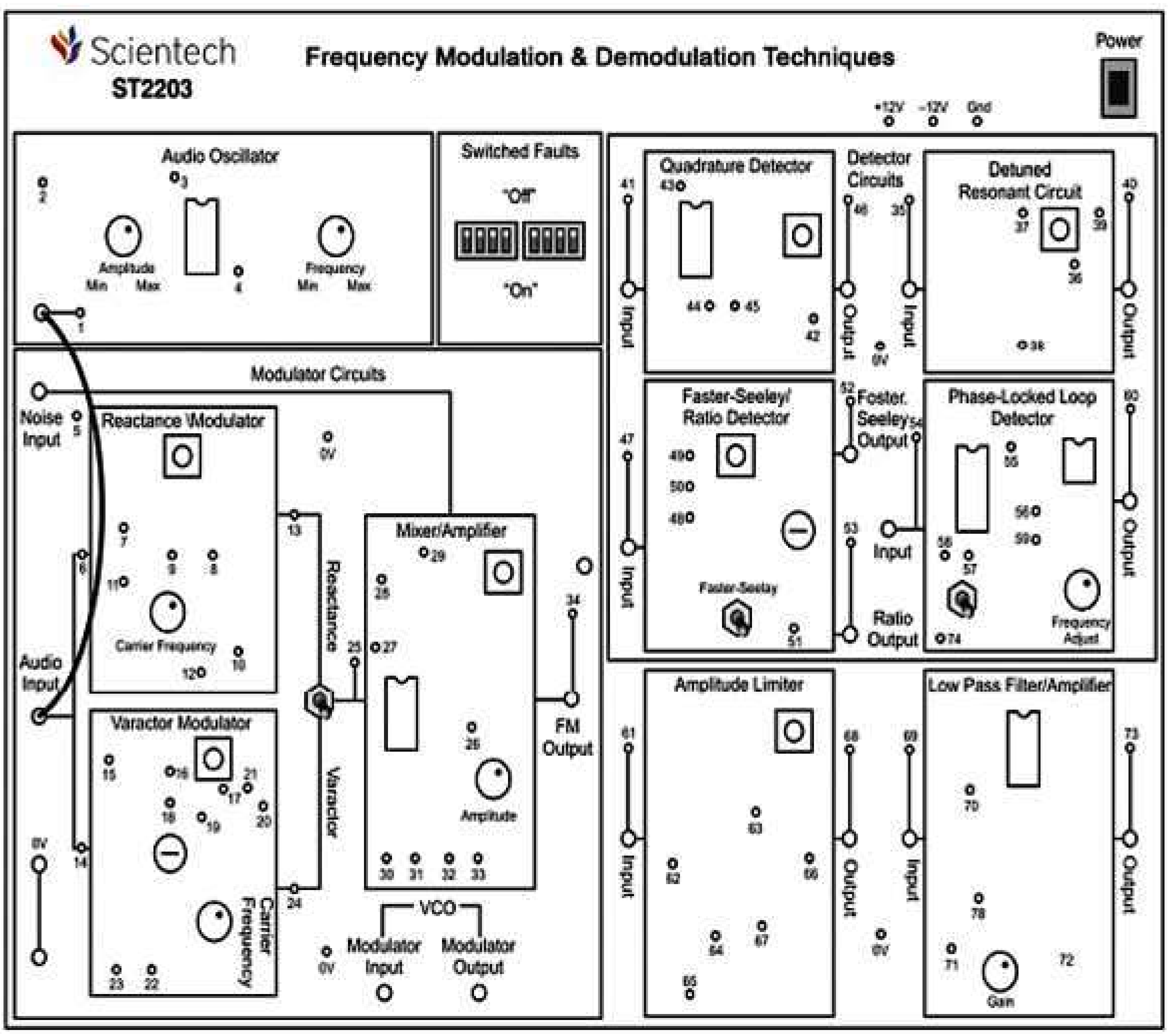


Fig.4.1

Procedure:

This experiment investigates how **ST2203**'s varacter modulator circuit performs frequency modulation. This circuit modulates the frequency of a carrier sine wave, according to the audio signal applied to its modulating input.

1. Ensure that the following initial conditions exist on the **ST2202** board.
    a. All Switched Faults in 'Off' condition.
    b. Amplitude potentiometer (in mixer amplifier block) in fully clockwise position.
    c. VCO switch (in phase locked loop detector block) in 'Off' position.
2. Make the connections as shown in figure 1.
3. Switch On the power.
4. Turn the audio oscillator block's amplitude potentiometer to its fully clockwise position, and examine the block's output TP1 on an Oscilloscope. This is the audio frequency sine wave, which will be used as our modulating 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 potentiometer.
   Note also that the amplitude of this modulating signal is adjusted by audio oscillator amplitude potentiometer Leave the amplitude potentiometer in minimum position.
5. Connect the output socket of the audio oscillator block to the audio input socket of the modulator circuit's block.
6. Set the reactance / varactor switch to the varactor position. This switch selects the varactor modulator and also disables the reactance modulator to prevent any interference between the two circuits.
7. The output signal from the varactor modulator block appears at TP24 before being buffered and amplified by the mixer/amplifier block, any capacitive loading (e.g. due to Oscilloscope probe) may slightly affect the modulators output frequency. In order to avoid this problem we monitor the buffered FM output signal the mixer / amplifier block at TP34.
8. Put the varactor modulator's carrier frequency potentiometer in its midway position, and then examine TP34. Note that it is a sine wave of approximately 1.2 Vp-p, centered on 0V. This is our FM carrier, and it is un-modulated since the varactor modulators audio input signal has zero amplitude.
9. The amplitude of the FM carrier (at TP34) is adjustable by means of the mixer/amplifier block's amplitude potentiometer, from zero to its potentiometer level. Try turning this potentiometer slowly anticlockwise, and note that the amplitude of the FM signal can be reduced to zero. Return the amplitude potentiometer to its fully clockwise position.
10. Try varying the carrier frequency potentiometer and observe the effects.

11. Also, see the effects of varying the amplitude and frequency potentiometer in the audio oscillator block.
12. Turn the carrier frequency potentiometer in the varactor modulator block slowly clockwise and note that in addition to the carrier frequency increasing there is a decrease in the amount of frequency deviation that is present.
13. Return the carrier frequency potentiometer to its midway position, and monitor the audio input (at TP6) and the FM output (at TP34) triggering the Oscilloscope on the audio input signal. Turn the audio oscillator's amplitude potentiometer 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 signals frequency and not in its amplitude.
14. *By using the optional audio input module **ST2108** the human voice can be used as the audio modulating signal, instead of using **ST2203**'s audio oscillator block. If you have an audio input module, connect the module's output to the audio input socket in the modulator circuit's block. The input signal to the audio input module may be taken from an external microphone be (supplied with the module) or from a cassette recorder, by choosing the appropriate switch setting on the module. Consult the user manual for the audio input module, for further details.*

❖ Frequency Modulation using Reactance modulator : -

Procedure:

This experiment investigates how **ST2203**'s reactance modulator circuit performs frequency modulation. This circuit modulates the frequency of a carrier sine wave, according to the audio signal applied to its modulating input

1. Set the reactance / varactor switch to the reactance position.
2. Follow the same as previous (varactor) method.

❖ Frequency Modulation using VCO: -

Simply connect the audio output to the socket labeled VCO modulation in and observe the FM modulated waveform on the Oscilloscope at the VCO modulation out terminal. Keep the amplitude of audio output to approximately 4 V p-p and frequency 2 KHz approximately Observe a stable FM modulated waveform on CRO.
This should look like as under.

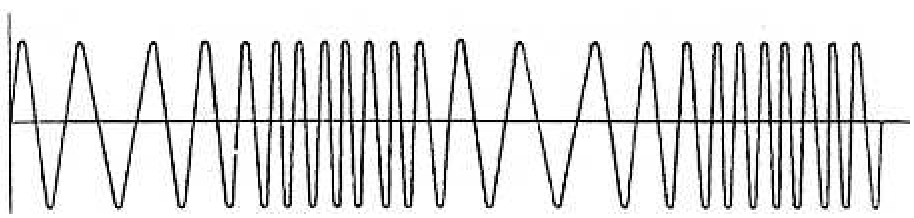


Fig.4.2

Now turn the time base speed of CRO little higher and you will observe the same waveforms as under (like Bessel function).

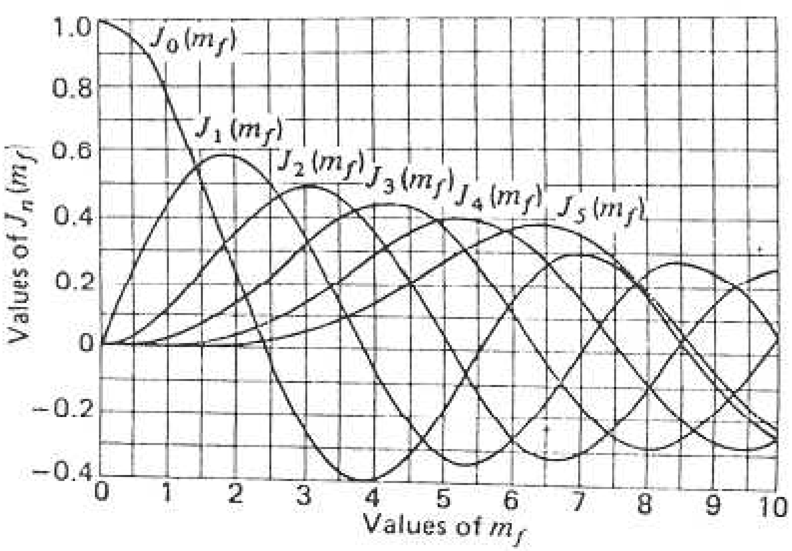


Fig.4.3

Now disconnect the audio amplifier's output from modulation In and connect it to audio In, keep the reactance/Varactor switch in Varactor position. Observe the output of mixer / amplifier circuit. The Frequency modulation in VCO was more because the Frequency difference between the carrier and the modulating signal was very less. But in **real life applications reactance and Varactor modulation techniques are used which utilizes high frequency carrier and you will not observe signal as shown in figure 2 above, but you will see as shown in figure 4.4.**

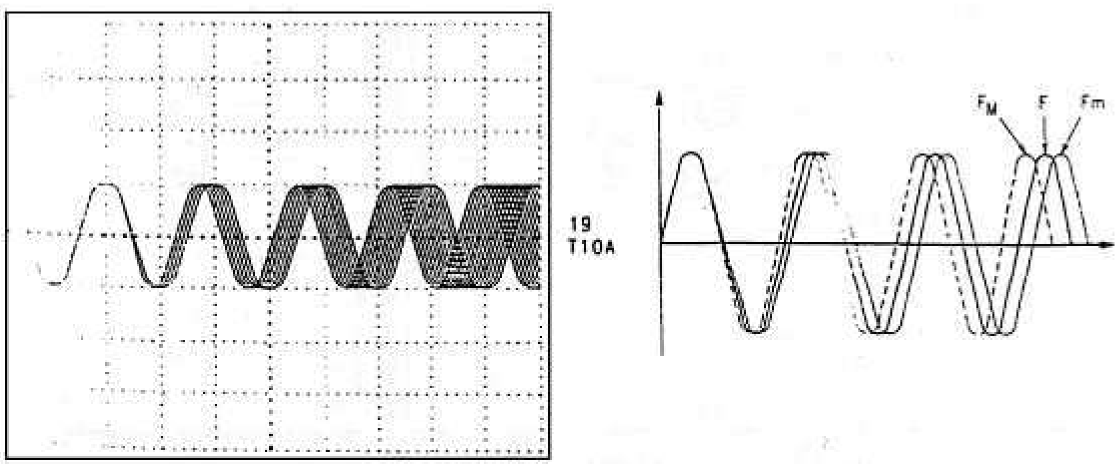


Fig.4.4

FM RECEPTION (DEMODULATION) USING DIFFERENTDETECTORS:

These experiments investigate how the detectors block on the **ST2203** module performs frequency demodulation. **The on-board amplitude limiter will then be used to remove any amplitude variations due to noise, before they reach the detector**. This allows the student to draw conclusions as to whether it is necessary to precede this type of detector with an amplitude limiter stage, in a practical FM receiver.

A. Demodulation using Detuned Resonant Circuit

Block Diagram:

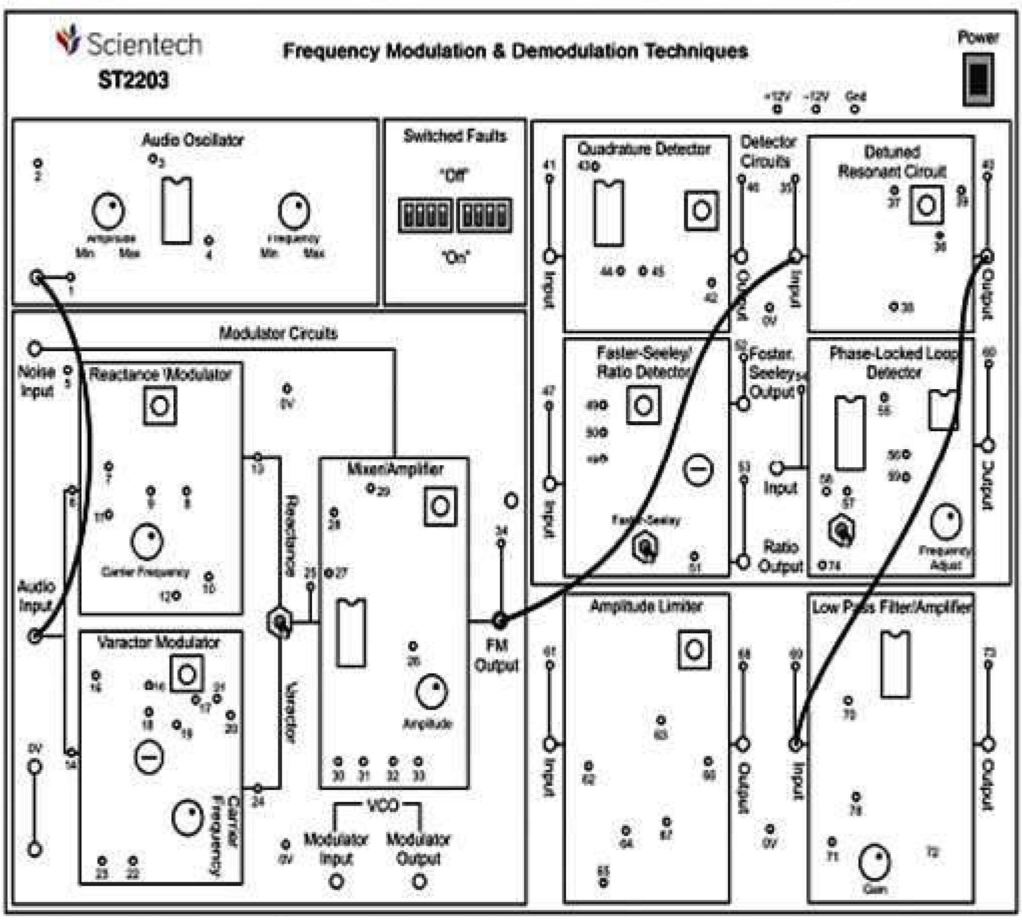


Fig.4.5

Procedure:

1. Ensure that the following initial conditions exist on the **ST2203** module.
   a. All Switched Faults in 'Off' condition.
   b. Audio amplifier block's amplitude potentiometer in fully clockwise (maximum) position.
   c. Audio amplifier block's frequency potentiometer in fully counterclockwise position.
   d. Amplitude potentiometer (in the mixer/amplifier block) in fully clockwise position.
   e. VCO switch (in phase - locked loop detector block) in 'Off' position.
2. Make the connections as shown in figure 5.
3. Switch on the power to the **ST2203** module.
4. Initially, we will use the varactor modulator to generate our FM signal, since this is the more linear of the two frequency modulators.
5. To select the varactor modulator, put the reactance/varactor switch in the varactor position. Ensure that the varactor modulator's carrier frequency potentiometer is in the midway position (arrowhead pointing towards top of PCB).
6. The audio oscillator's output signal (which appears at TP1) is now being used by the varactor modulator, to frequency'- modulate a 455 KHz carrier sine wave. As we saw earlier, this FM waveform appears at the FM output socket from the mixer/amplifier block.You may like to examine this FM waveform at TP34. However, with thevaractor modulator's carrier frequency potentiometer in its present (midway)position, the frequency deviation is quite small. To be able to notice such asmall frequency deviation, you will probably need to have a control on your Oscilloscope.If you have such a control, display 20-25 cycles of the waveform on the Oscilloscope, and then use the X-expansion control to 'expand up' the right most cycles of the display. There should be a slight ambiguity in the positions of these cycles, indicating that the sine wave at TP34 is being frequency modulated.
7. Now monitor the audio input signal to the varactor modulator block (at TP14),together with the output from the detuned resonant circuit block (at TP40)triggering the Oscilloscope on TP14). *The signal at TP40 should contain three components :*
   - A positive DC offset voltage;
   - A sine wave at the same frequency as the audio signal all TP14.
   - A high-frequency ripple component of small amplitude.

   Check that the audio-frequency component is a reasonable sine wave. If it is not, it is likely that the centre frequency of the varactor modulator's FM output needs adjusting slightly. To do this, trim transformer T2 in the varactor modulator block, in accordance with the instructions given inchapter.(Adjustment of **ST2203**'s tuned circuits).

8. The low-pass filter/amplifier block strongly attenuates the high frequency ripple component at the detector's output, and also blocks the DC offset voltage. Consequently, the signal at the output of the low-pass filter/amplifier block (atTP73) should very closely resemble the original audio modulating signal.
Monitor the input (TP69) and output (TP73) of the low pass filter/amplifier block (triggering on TP 73) and note how the quality of the detector's output signal has been improved by low pass filtering. Note also that the DC offset has been removed.

9. Monitor the audio input to the varactor modulator (at TP14) and the output ofthe low-passfilter/amplifier block (at TP73) and adjust the gain potentiometer in the low pass filter/amplifier block, until the amplitudes of the two monitored audio waveforms are the same.

10. Adjust the audio oscillator block's amplitude and frequency potentiometer, and compare the original audio signal with the final demodulated signal. You may notice that the demodulated output suffers attenuation as the audio modulating frequency is increased. This is caused by low-pass filtering, which takes place in the detuned resonant circuit's envelope detector, and in the low pass filter/amplifier block.
In spite of this high-frequency limitation to the range of audio frequencies, which can be received, the bandwidth of the system is perfectly adequate for normal speech communication.
In the audio oscillator block, put the amplitude potentiometer in its Maximum position and the frequency potentiometer in its Minimum position.

11. We will now investigate the effect of noise on the system. Adjust the external signal generator for a sinusoidal output of amplitude 100m Vp-p, and frequency2 KHz; this will be our 'noise' input. Connect the output of the signal generator to the noise input socket in **ST2203**'s modulator circuit's block. Then, monitor the noise input (at TP5) and the FM output (at TP34) triggering the Oscilloscope on TP5. Note that the FM signal is now being amplitude modulated by the 'noise' input, in addition to being frequency-modulated by the audio input from the audio oscillator block. The amplitude modulations simulate the effect that transmission path noise would have on the amplitude of the FM waveform reaching the receiver. This allows us to investigate the effect of transmission path noise would have on the final demodulated audio signal.

12. Monitor the audio modulating signal (at TP14) and the output of the low pass filter/amplifier block (at TP73), triggering the Oscilloscope from TP14. Note that there is now an additional component at TP73a sine wave at the frequency of the 'noise' input. To see this clearly, it may be necessary to slightly adjust the frequency of the signal generator's output, until the superimposed 'noise' sine wave can be clearly seen.

13. Remove the Oscilloscope probe form TP73, and place it on TP40, the output form the detuned resonant circuit detector. Note that the 'noise' component is still present, illustrating that this type of detector is very susceptive to amplitude variations in the incoming FM signal.Put the Oscilloscope probe on TP39 (the collector of the detuned resonant circuit's transistor) to ensure that you fully understand why this type of detector is so sensitive to amplitude variations.

14. Turn the audio oscillator block's amplitude potentiometer to its minimum position, so that no frequency modulation takes place. Then monitor the 'noise' input (at TP5) and the output from the low pass filter/amplifier block (at TP73), triggering the Oscilloscope from TP5. The signal at TP73 in now purely composed of the 'noise' output resulting from amplitude variations occurring at the input to the detuned resonant circuit. Measure and record the peak-to-peak amplitude of the 'noise' output at TP73; this measurement will be valuable in allowing us to compare the detuned resonant circuit with other types of FM detector, as far as susceptibility to amplitude modulation is concerned.

15. To overcome the problem of the detuned resonant circuit detector's susceptibility to noise, **we can connect an amplitude limiter block between the FM output and the input to the detuned resonant circuit**. The amplitude limiter removes amplitude variations from the FM output signal, so that the input signal to the detuned resonant circuit detector has constant amplitude. Reconnect the amplitude limiter block between the mixer/amplifier block and the detuned resonant circuit block as shown in figure 4.6 at the end.

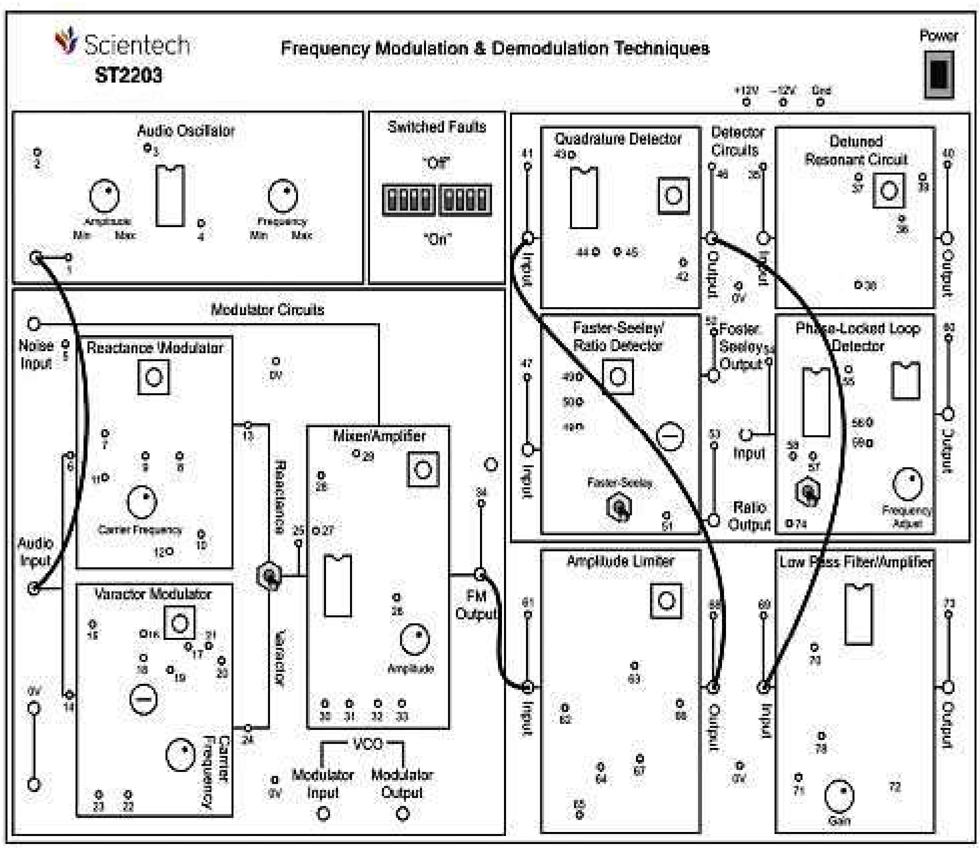


Fig.4.6

16. Monitor the amplitude limiter's output at TP68, triggering the Oscilloscope from TP5, the noise input form the signal generator. Note that the amplitude modulations due to the noise input have been removed. Remove the Oscilloscope probe from TP68, and put it on TP73, the output form the low pass filter/amplifier block. Note that the amplitude of any remaining 'noise' component at TP73 is now minimal.
17. Return the audio oscillator blocks amplitude potentiometer to its maximum position, and monitor TP73, triggering the Oscilloscope on the audio modulating input at TP14. **Note** that amplitudes now have no effect on the final audio output. This shows how an amplitude limiter can be used in a practical FM receiver, to remove amplitude variations caused by noise, before they reach the detector.
18. By using the optional audio input module and audio output module the human voice can be used as the audio modulating signal, instead of using **ST2203**'s audio oscillator block. If you have these modules, make the following connections :
    - Output of audio input module to audio input socket in **ST2203**'s modulator circuits block;
    - Output of **ST2203**'s low pass filter/amplifier block to input socket of audio output module.
    - Refer the user manuals for the audio input module **ST2108** and audio output module **ST2109** for further details of how to use them.
19. Throughout this experiment, frequency modulation has been performed by **ST2203**'s varactor modulator block. Equally, using the reactance modulator block may perform frequency modulation. If you wish to repeat any of the above experimentation with the reactance modulator, simply put the reactance/varactor switch in the reactance position.

    Note: However, that the linearity of the reactance modulator is not as good as that of the varactor modulator. This means that, when the reactance modulator is used, some distortion of the demodulated audio signal may be noticeable at the detector's output, if the amplitude of the audio-modulating signal is too large.
20. Finally, make sure that you fully understand the working of the detuned resonant circuit detector by examining the circuit diagram for the detector at the end of this manual, and monitoring Test Points within the circuit.

B. Demodulation using Quadrature Detector Circuit

Block Diagram:

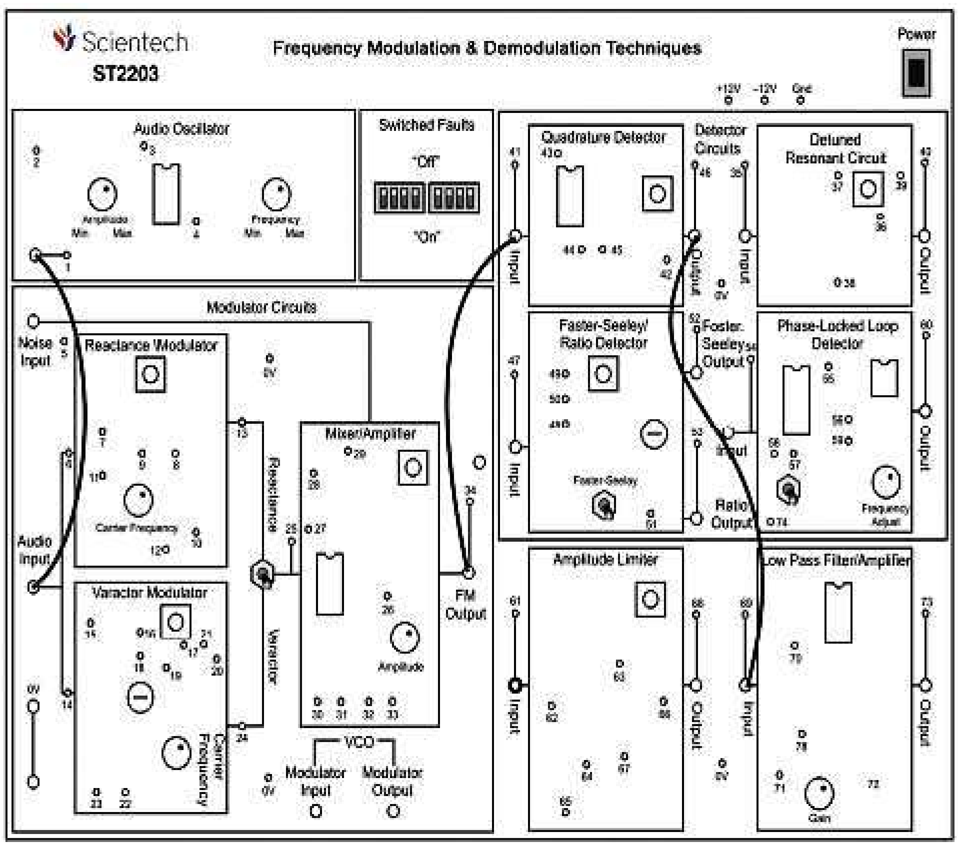


Fig.4.7

Procedure:

1. Ensure that the following initial conditions exist on the **ST2203** module.
   a. All Switch Faults in 'Off' condition.

b. Audio amplifier block's amplitude potentiometer in fully clockwise (maximum) position;
c. Audio oscillator block's frequency potentiometer in fully counter clockwise (minimum)position.
d. Amplitude present (in the mixer/amplifier block) in fully clockwise position.
e. VCO switch (in phase-locked loop detector block) in 'Off' position.

2. Make the connections shown in figure 4.7.
3. Turn on power to the ST2203 module.
4. Initially, we will use the varactor modulator to generate our FM signal, since this is the more linear of the two frequency modulators as far as its frequency/voltage characteristic is concerned.
   To select the varactor modulator, put the reactance/varactor switch in the varactor position. Ensure that the varactor modulator's carrier frequency potentiometer is in the midway position.
5. The varactor modulator, to frequency-modulate a 455 KHz-carrier sine wave is now using the audio oscillator's output signal (which appears at TP1). As we saw earlier, this FM waveform appears at the FM output socket from the mixer/ amplifier block. You will probably need to have an X-expansion control on your Oscilloscope.
6. Now monitor audio input signal to the varactor modulator block (at TP14) together with the output form the quadrature detector block (at TP46), triggering the Oscilloscope. The signal at TP46 should contain three components.
   a. A positive DC offset voltage.
   b. A sine wave at the same frequency as the audio signal at TP14.
   c. A high-frequency ripple component of small amplitude.

   Check that the audio frequency component is a reasonable sine wave. It is likely that the entire frequency of the varactor modulator's FM output needs right adjustment. To do this, trim transformer T2 in the varactor modulator block, in accordance with the instructions given in chapter coil adjustments.
7. The low-pass filter/amplifier block strongly attenuates the high frequency ripple component at the detector's output, and also blocks the DC offset voltage. Consequently, the signal at the output of the low-pass filter/amplifier block (at TP73) should very closely resemble the original audio modulating signal.
8. Monitor the audio input to the varactor modulator (at TP14) and the output of the low-pass filter/amplifier block (at TP73) and adjust the gain potentiometer (in the low pass filter/amplifier block) until the amplitudes of the monitored audio waveforms are the same.
9. Adjust the audio oscillator block's amplitude and frequency potentiometer and compare the original audio signal with the final demodulated signal.

10. We will now investigate the effect of noise on the system. Adjust the signal generator for a sinusoidal output of amplitude 100m Vp-p, and frequency 2 KHz, this will be our 'noise' input.
    Connect the output of the signal generator to the noise input socket in **ST2203**'s modulator circuit's block. Monitor the noise input (at TP5) and the FM output (at TP34) triggering the Oscilloscope on TP5. Note that the FM signal is now being amplitude-modulated by the 'noise' input, in addition to being frequency-modulated by the audio input from the audio oscillator block. The amplitude modulations simulate the effect that transmission path noise would have on the amplitude of the FM waveform reaching the receiver. This allows us to demodulated audio signal.
11. Monitor the audio modulating signal (at TP14) and the output of the low pass filter/amplifier block (at TP73), triggering the Oscilloscope from TP14.
12. Remove the Oscilloscope probe form TP73 and place it on TP46 the output form the quadrature detector block. Note that the small 'noise' component is still visible.
13. Turn the audio oscillator block's amplitude potentiometer to its MIN position, so that no frequency modulation takes place. Then monitor the 'noise' input (at TP5) and the output from the low pass filter/amplifier block (at TP73, triggering the Oscilloscope from TP5.
14. To reduce the effect of amplitude variations even further, we can connect an amplitude limiter block between the FM output and the input to the quadrature detector. The amplitude limiter removes amplitude variations from the FM output signal, so that the input signal to the quadrature detector has constant amplitude. Reconnect the amplitude limiter block between the mixer/amplifier block and the quadrature detector block.
15. Monitor the amplitude limiter's output at TP68, triggering the Oscilloscope from TP5, the 'noise' input from the signal generator. Note that the amplitude modulations due to the 'noise' input have been removed. Remove the Oscilloscope probe from TP68, and put it on TP73, the output form the low pass filter/amplifier block. Note that the amplitude of any remaining 'noise' component at TP73 is now minimal.
16. By using the optional audio input module and audio output module, the human voice can be used as the audio modulating signal, instead of using **ST2203**'s audio oscillator block.
17. Throughout this experiment, frequency modulation has been performed by **ST2203**'s varactor modulator block. Using the reactance modulator block we may perform frequency modulation.

C. Demodulation using Phase-Locked Loop Detector Circuit

Block Diagram:

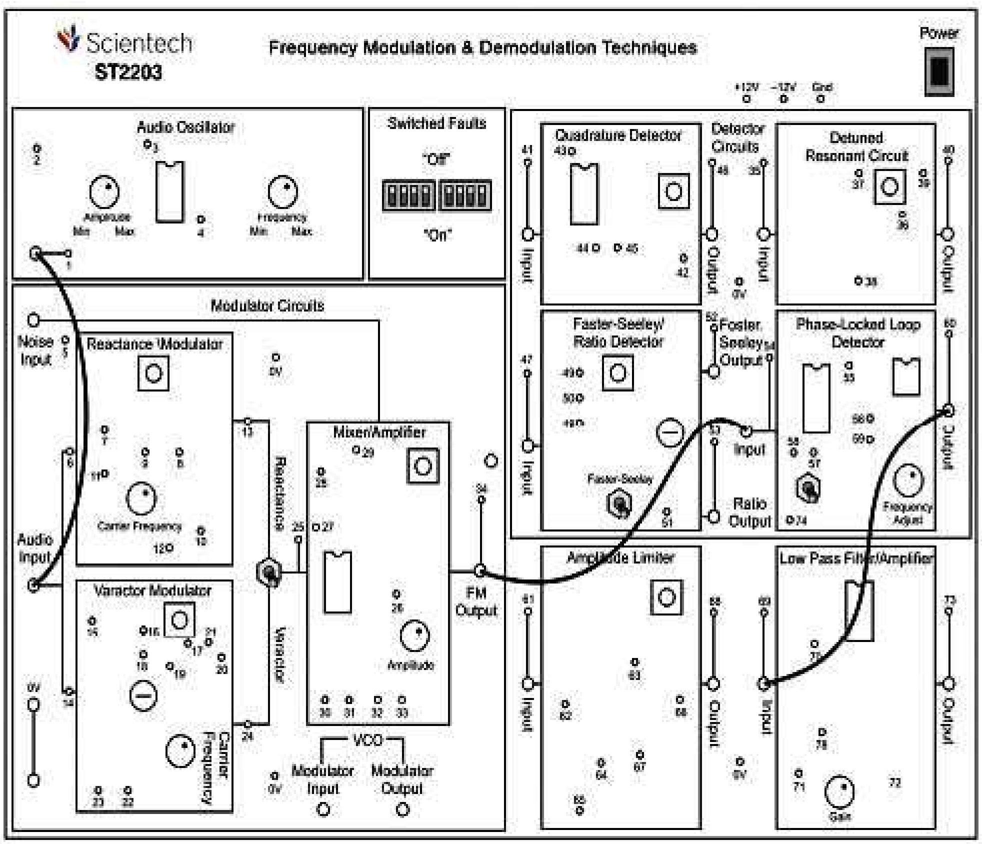


Fig.4.8

Procedure:

1. Ensure that the following initial conditions exist on the **ST2203** module:
    a. All Switched Faults in 'Off' condition.
    b. Audio amplifier block's amplitude potentiometer in fully clockwise (maximum) position.
    c. Audio amplifier block's frequency potentiometer in fully counter clockwise. Ensure that the following initial conditions exist on the **ST2203** clockwise (minimum) position.
    d. Amplitude potentiometer (in the mixer/amplifier block) in fully clockwise position
    e. VCO switch (in phase-locked loop detector block) in 'On' position.
2. Make the connections shown in figure 8.
3. Turn on power to the ST2203 module.
4. Now monitor the audio input signal to the varactor modulator block (at TP14) together with the output from the phase-locked loop detector block (at TP60), triggering the Oscilloscope in TP14. The signal at TP68 should contain three components.
    - A positive DC offset voltage.
    - A sine wave at the same frequency as the audio signal at TP14.
    - A high - frequency ripple component.
5. The low pass filter/amplifier block strongly attenuates the high-frequency ripple component at the detector's output and also blocks the DC offset voltage. Consequently the signal at the output of the low- pass filter/amplifier block (at TP73) should be very closely resemble the original audio making signal, if not then slowly adjust the frequency adjust potentiometer of PLL block.
6. Adjust the audio oscillator block's amplitude and frequency potentiometer and compare the original audio signal with the final demodulated signal.
7. We can investigate the effect of noise on the system by following the procedure given in earlier experiments. The only change will be that we will use phase locked loop detector instead of quadrature or detuned resonant circuit.

D. Demodulation using Foster - SeeleyDetector Circuit

Block Diagram:

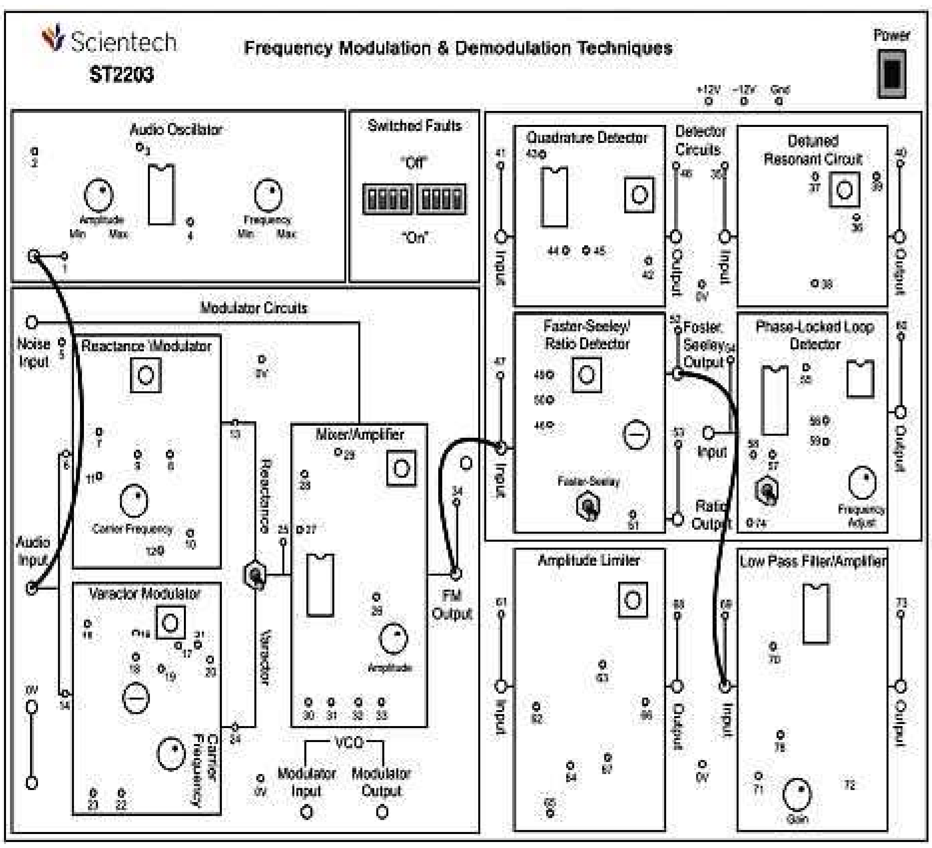


Fig.4.9

Procedure:

1. Ensure that the following initial conditions exist on the ST2203 module:
   a. All Switched Faults in 'Off' condition.
   b. Audio amplifier block's amplitude potentiometer in fully clockwise(maximum) position.
   c. Audio amplifier block's frequency potentiometer in fully counter-clockwise(minimum) position.
   d. Amplitude potentiometer (in the mixer/amplifier block) in fully clockwise position.
   e. VCO switch (in phase-locked loop detector block) in 'Off' position.
2. Make connection as shown in figure 4.9.
3. Turn on power to the ST2203 module.
4. We will now investigate the operation of the Foster-Seeley detector on theST2203 module.In the Foster-Seeley/ratio detector block, select the Foster-Seeley detector byputting the switch in the Foster-Seeley position.
5. Initially, we will use the varactor modulator to generate our FM signal, sincethis is the more linear of the two modulators, as fast as its frequency/voltage characteristic is concerned. To select the varactor modulator, put the reactance/varactor switch in the varactor position. Ensure that the varactor modulator's carrier frequency potentiometer is in the midway position.
6. The audio oscillator's output signal (which appears at TP1) is now being usedby the varactor modulator, to frequency-modulate a 455 KHz carrier sine wave.As we saw earlier, this FM waveform appears at the FM output socket from themixer/amplifier block. You will probably need to have an X-expansion controlon your Oscilloscope.
7. Now monitor the audio input signal to the varactor modulator block (at TP14)together with the Foster-Seeley output from the Foster-Seeley/ratio detector block (at TP52), triggering the Oscilloscope on TP14.The signal at TP52 should contain two components:
   - A sine wave at the same frequency as the audio signal at TP14.
   - A High frequency ripple component of small amplitude.
8. The Low-Pass Filter/amplifier strongly attenuates this high-frequency ripple component, and blocks any small DC offset voltage that might exist at the detector's output. Consequently, the signal at the output of the Low-Pass Filter/ amplifier block (at TP73) should very closely resemble the original audio modulating signal.
9. Monitor the audio input to the varactor modulator (at TP14) and the output of the Low Pass Filter/amplifier block (at TP73) and adjust the gain potentiometer (in the Low Pass Filter/amplifier block) until the amplitudes of the monitored audio waveforms are the same.

10. Adjust the audio oscillator block's amplitude and frequency potentiometer, and compare the original audio signal with the final demodulated signal.
11. We can investigate the effect of noise on the system by following the procedure given in earlier chapters by merely substituting quadrate detector by Foster-Seeley Detector.

E. Demodulation using Ratio Detector Circuit

Block Diagram:

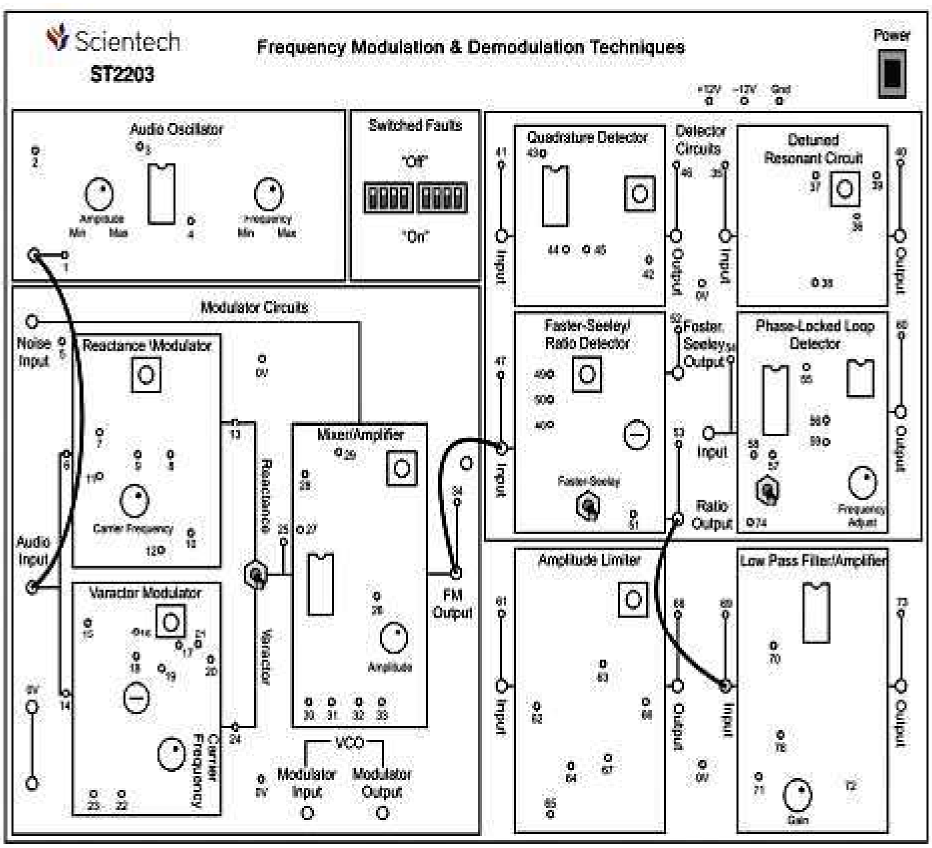


Fig.4.10

Procedure:

1. Ensure that the following initial conditions exist on the ST2203 module.
    - All Switched Faults in 'Off' condition.
    - Audio amplifier block's amplitude potentiometer in fully clockwise (maximum) position.
    - Audio amplifier block's frequency potentiometer in fully counter clockwise (minimum) position.
    - Amplitude potentiometer (in the mixer/amplifier block) in fully clockwise position.
    - VCO switch (in phase-locked loop detector block) in 'Off' position.
2. Make connections as in figure 10.
3. Turn on power to the module.
4. Now monitor the audio input signal to the varactor modulator block (at TP14) together with the ratio output from the Foster-Seeley/ratio detector block (at TP53) triggering the Oscilloscope on TP14. The signal at TP53 should be contain two main components:
    - A positive DC offset voltage.
    - A sine wave at the same frequency as the audio signal at TP14, but shifted in phase by 180°
    - Note that the amount of high-frequency ripple present on the signal is very small this is due to the smoothing effect of the large output capacitor.
5. The Low-Pass Filter/amplifier block removes the DC offset voltage at the detector's output, and strongly attenuates any residual high-frequency ripple that may be present. Consequently, the signal at the output of the low-pass filter/amplifier block (at TP73) should very closely resemble the original audio-modulating signal.
   Monitor the input (TP73) and output (TP73) of the low pass filter/amplifier block (triggering on TP73) and note how the two signals differ.
6. Monitor the audio input to the varactor modulator (at TP14) and the output of the low-pass filter / amplifier block (at TP73) and adjust the gain potentiometer (in the low pass filter/amplifier block) until the amplitudes of the monitored audio waveforms are the same.
7. Adjust the audio oscillator block's amplitude and frequency potentiometer, and compare the original audio signal with the final demodulated signal.
8. We can investigate the effect of noise on the system by following the procedures given in earlier chapters by substituting the quadrature detector by ratio detector.

ST2204

❖ Frequency Modulation and Demodulation : -

Block Diagram:

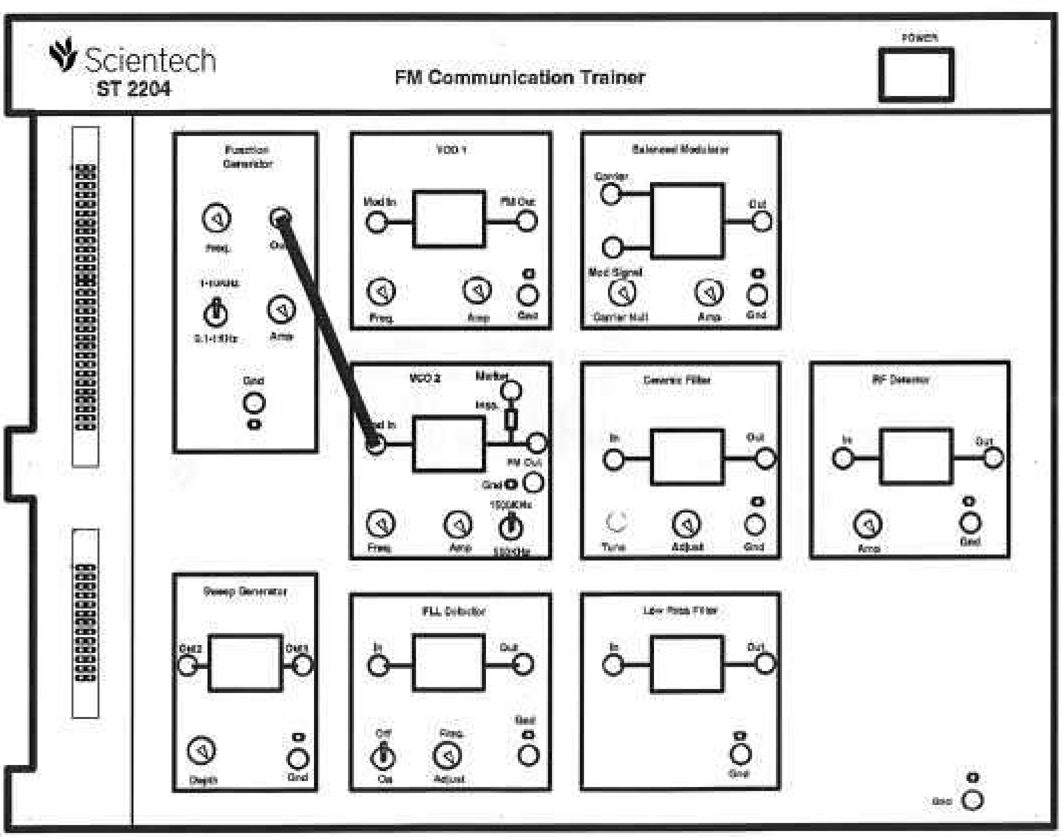


Fig.4.11

Procedure:

1. Identify the modulators (VCO1 and VCO2) and demodulator (PLL Detector) portions on the experiment kit. Also identify the input port, output port and test terminal/points.
2. Connect the 230V mains to the experiment kit.
3. Connect the audio output to the input of VCO1 or VCO2 and observe the FM modulated waveform on the oscilloscope at the VCO modulation out terminal. Keep the amplitude of audio output to approx.4 Vp-p and Frequency 2 KHz approx. Observe a stable FM modulated waveform on CRO and calculate the frequency deviation and take the frequency deviations at three different amplitudes. Waveform should look like as under (Fig.12). To get the better result adjust the frequency and amplitude of the VCO level.

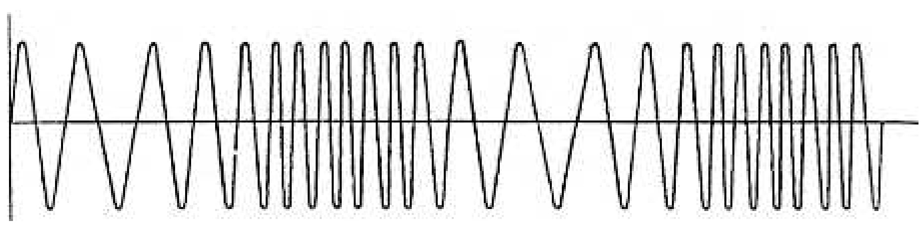


Fig.4.12

4. Now turn the time base speed of the CRO little higher and you will observe the same waveforms as under (like Bessel function) (Fig.13)

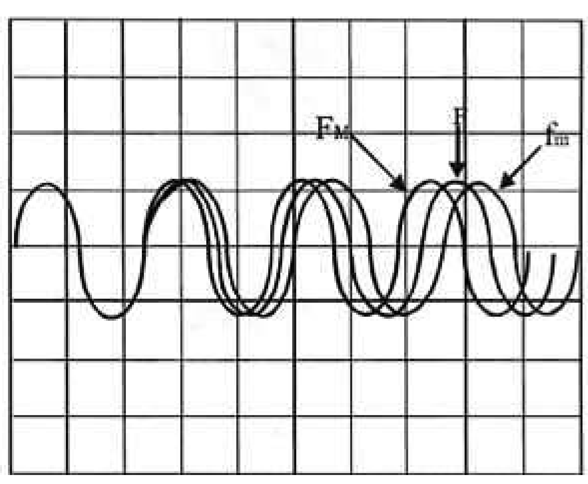


Fig. 4.13

5. Calculate frequency deviation , modulation index and constant proportionality with the help of equations 2 & 3
6. Changing the frequency and amplitude of the modulating signal repeat steps 3, 4 and 5.

<u>Spectrum of the FM signal</u>

Connection Diagram:

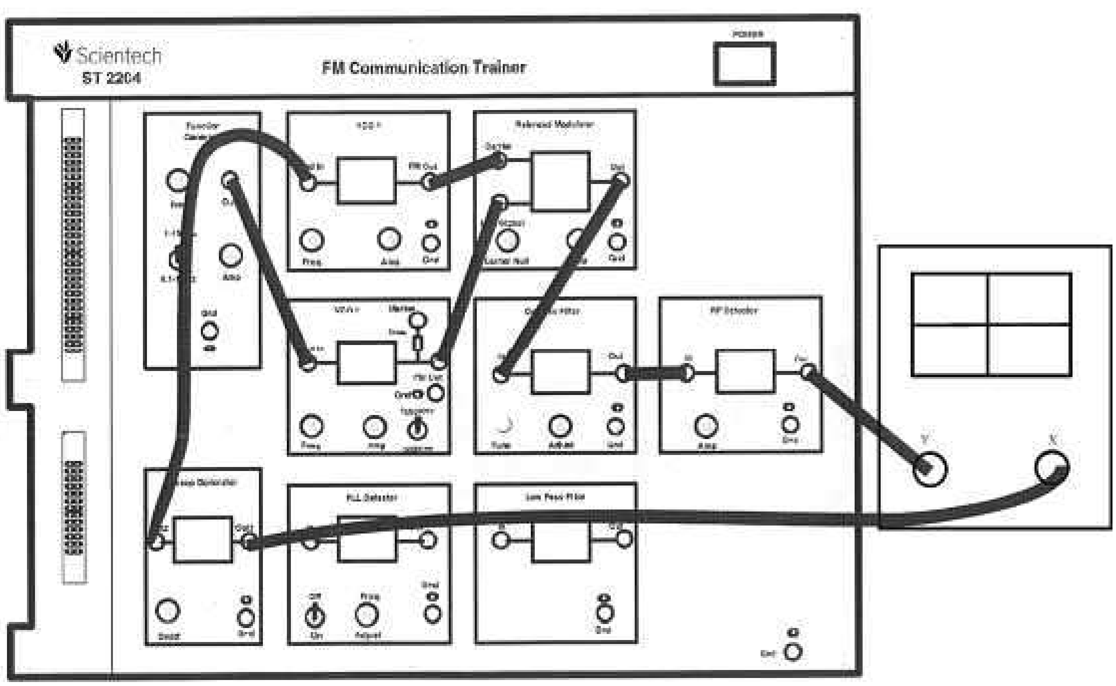


Fig.4.14

Procedure:

1. Carry out the following connections and presetting.
   a. Function Generator: Level about 5Vpp and frequency about 10 KHz.
   b. VCO 1: Level about 3Vpp and frequency potentiometer on center.
   c. VCO 2: Level about 2Vpp, frequency selection switch on 1500 KHz and frequency potentiometer completely clockwise.
   d. Sweep depth: almost completely clockwise.
   e. RF Detector Level: Completely clockwise.
   f. Balanced Modulator 1: Carrier Null Completely clockwise, so that the circuit operates as frequency converter (balanced modulator with suppressed carrier), Level fully clockwise.
   g. Trimmer of the ceramic filter completely clockwise.
2. Set the Oscilloscope in XY mode (X = .1V/div., Y = .2 V/div.). Connect the sweep generator OUT 1 to the X axis, and the detected output to Y axis.
3. Vary the frequency of VCO 1 until the Oscilloscope gives a similar representation to the one of figure 8. To obtain the best waveform adjust the Depth of the sweep generator, slight adjustment of VCO 2 (if needed), Carrier null of balanced modulator.

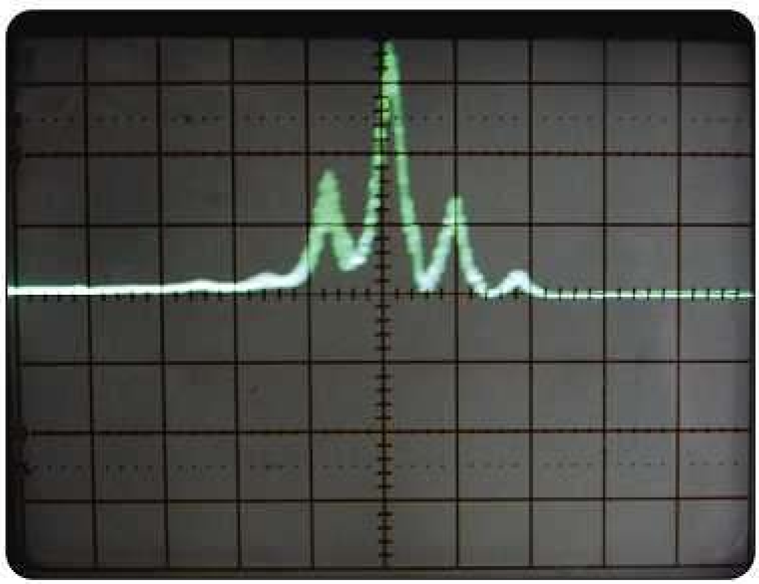


Fig.4.15

4. The waveform represents the spectrum of the FM signal. Check the distribution of the harmonics is symmetrical in respect to the one of the carriers.
5. Vary the amplitude and frequency of the modulating signal, and examine how the spectrum varies. Check that when the amplitude increases also the number of spectral

lines increases (the increase of the amplitude of the modulating signal causes the frequency deviation to increase D_f, this causes the modulation index m to increase and consequently the spectrum to enlarge). Check that the spectral line corresponding to the carrier varies in amplitude and annuls in correspondence to frequency deviation values.

FM RECEPTION (DEMODULATION) in KIT 2204

Connection Diagram:

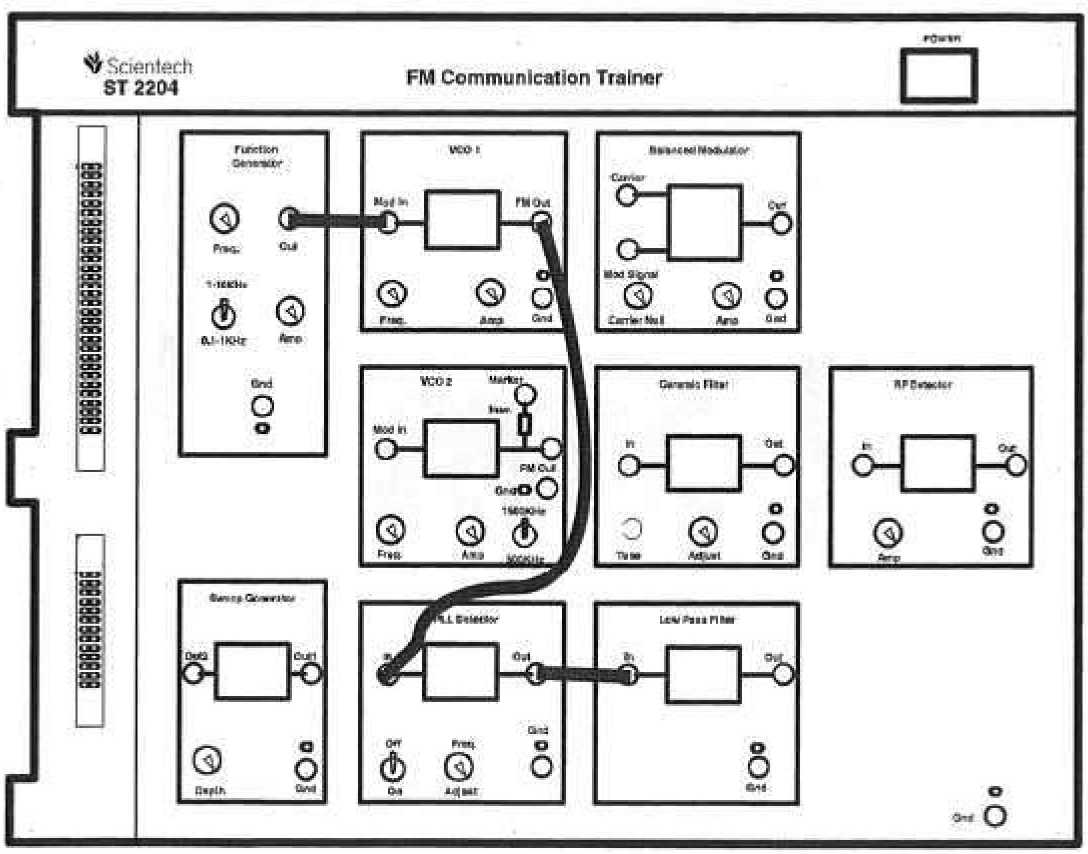


Fig. 4.16

Procedure:

1. Carry out the connections and presetting as under:
    a. VCO 1/2: Frequency at 1MHz, and amplitude at maximum
    b. Function generator: Frequency at 10 KHz and amplitude 0.5V.
    c. Connect output of FG block to input of either of VCO
    d. Output of VCO to input of PLL detector
    e. Output of PLL detector to input of low pass filter
    f. Switch in PLL detector block in 'On' position.
2. Observe the output of low pass filter circuit, while adjusting the frequency adjust potentiometer in the PLL detector.
3. Note that the sine wave observed on the CRO resembles the modulating signal. Vary the modulating signal's frequency and amplitude to confirm that it is the demodulated output.

ME 742

❖ Frequency Modulation and Demodulation : -

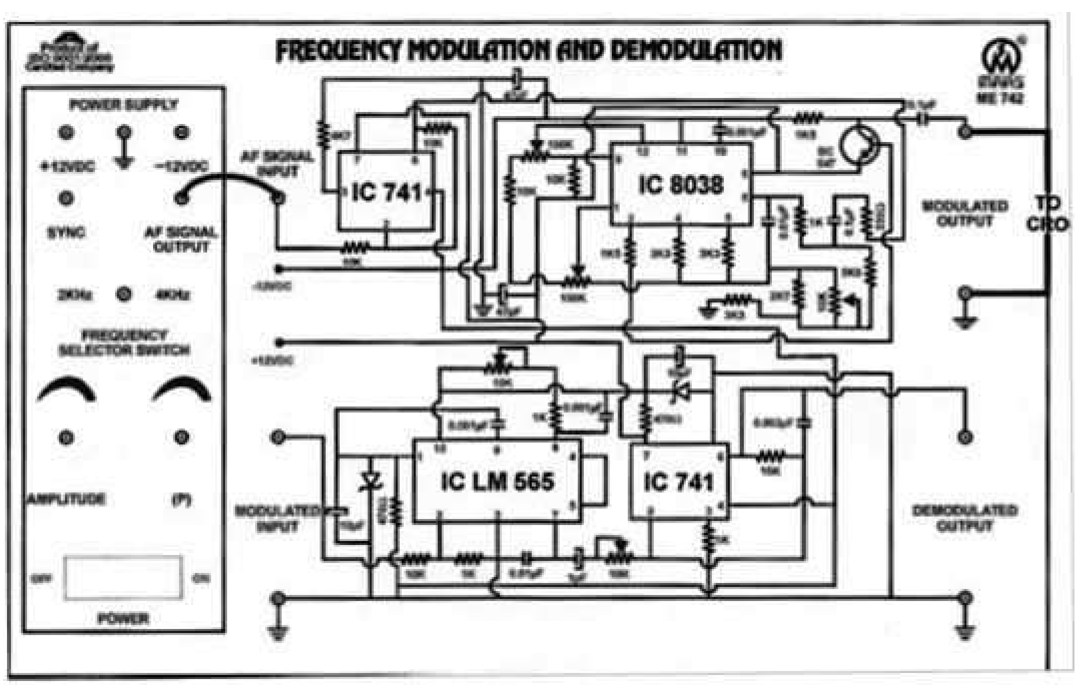


Fig. 4.17

Procedure:

1. Carry out the connections and presetting as under:
    a. Function generator: Frequency at 2-3 KHz and amplitude 1-2 V.
    b. Connect between AF Signal output and AF Signal input (Input of Modulator) as in figure4.17
2. Observe the FM modulated waveform on the oscilloscope at the modulated output terminal. Keep the amplitude of audio output to approx.4 Vp-p and Frequency 2 KHz approx. Observe a stable FM modulated waveform on CRO. Waveform should look like as under (Fig. 4.12). To get the better result adjust the frequency and amplitude AF signal.
3. Now turn the time base speed of the CRO little higher and you will observe the same waveforms as under (like Bessel function) (Fig. 4.13)
4. Calculate the frequency deviation and take the frequency deviations at three different amplitudes.

FM RECEPTION (DEMODULATION) in KIT ME742

Connection Diagram:

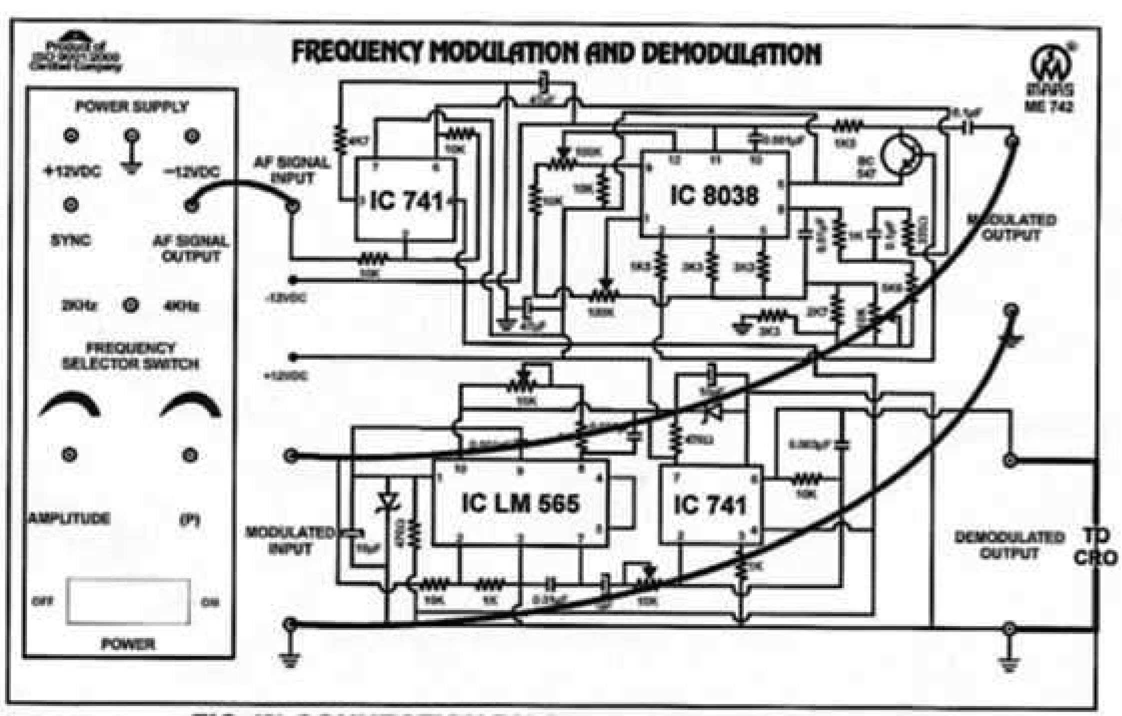


Fig. 4.18

Procedure:

1. Carry out the connections and presetting as under:
    a. Function generator: Frequency at 2-3 KHz and amplitude 1-2 V.
    b. Connect between AF Signal output and AF Signal input (Input of Modulator) as in figure 4.18
    c. Output of Modulator to input of Demodulator (PLL detector)
2. Observe the output of Demodulator (PLL detector).
3. Note that the sine wave observed on the CRO resembles the modulating signal. Vary the modulating signal's frequency and amplitude to confirm that it is the demodulated output.

Experimental Data:

Sl. No.	Modulating Signal		Carrier frequency	Modulated Output		Frequency Deviation	Modulation Index	Constant proportionality
	Freq. **(f_m)**	Ampl. **(A_m)**	**(f_c)**	**f_{max}**	**f_{min}**	**$\Delta f = (f_{max} - f_{min})/2$**	**$m = \Delta f/f_m$**	**$K_f = \Delta_f/A_m$**
1.								
2.								
3.								

LPF Response:

Table 2

Sl. No	Input Signal		Output Signal	
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)
1.	f_1	A_{in}	f_{o1}	A_1
2.	f_2	A_{in}	f_{o2}	A_2
3.	f_3	A_{in}	f_{o3}	A_2
n.	f_n	A_{in}	f_{o4}	A_n

Linearity:

Table 3

Sl. No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_{o1}	A_{o1}	M_1
2.	f_1	A_2	f_{o1}	A_{o2}	M_2
3.	f_1	A_2	f_{o1}	A_{03}	M_3
n.	f_1	A_n	f_{o1}	A_{on}	M_n

Table 4

Sl. No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_{o1}	A_{o1}	M_1
2.	f_2	A_1	f_{o2}	A_{o2}	M_2
3.	f_3	A_1	f_{o3}	A_{03}	M_3
n.	f_n	A_1	f_{on}	A_{on}	M_n

Data Analysis:

1. Calculate Carrier swing (CS), modulation index and constant proportionality from the modulating and modulated waveforms.
2. Calculate the ratio of amplitudes of input and output signals and show the linearity of LPF.
3. Find the cut off frequency of LPF. Tally the frequency of input and output signal of modulator and demodulator.

Discussion:

Write briefly your comments about the above experiment.

Precautionary Measure to be taken:

1. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
2. For connecting any signal from one equipment to equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.

3. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.
4. For unusual spark or burning smell, immediately switch off the main supply to the kit.
5. Ensure that, a signal is connected to an appropriate junction destined for it.
6. Always cover the equipment for dust protection after the experiment.

Some Sample questions:

1. Find the frequency deviation at three different amplitudes.

2. Design a FM system taking modulation frequency 2 KHz. Calculate the leading time and fall time for the VCO output. Draw the demodulated output signal.

EXPERIMENT # 5

Title: STUDY OF PULSE MODULATION & DEMODULATION

Objective:

- Study Of Pulse Amplitude Modulation / Demodulationusing Natural & Flat Top Sampling And Sample & Hold Circuit

Theory:

1. **Pulse Amplitude Modulation (PAM):**
Pulse Amplitude Modulation (PAM) is an analog pulse modulation scheme where a rectangular pulse train is used as a carrier waveform and the analog modulating signal is used to change the height of the carrier pulses at discrete instants. Nyquist's sampling theorem forms the basis of the principle of PAM. The analog modulating signal can be satisfactorily recovered from the modulated signal by using an appropriately designed low pass filter. However, the demodulated signal may have low level of distortion compared to the original modulating signal.
In pulse amplitude modulation system the amplitude of the pulse is varied in accordance with the instantaneous level of the modulating signal. Now days, the PAM system is not generally used, but it forms the first stage of the other types of pulse modulation.
There are two types of sampling techniques for transmitting a signal using PAM. They are:
   - Flat Top PAM
   - Natural PAM

Flat Top PAM: The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. The amplitude of the signal cannot be changed with respect to the analog signal to be sampled. The tops of the amplitude remain flat.

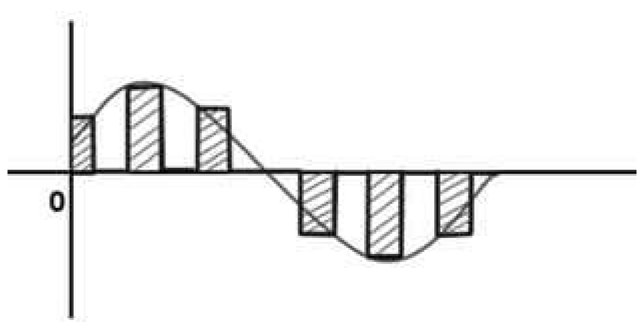


Fig. 5.1

Natural PAM: The amplitude of each pulse is directly proportional to modulating signal amplitude at the time of pulse occurrence. Then follows the amplitude of the pulse for the rest of the half cycle.

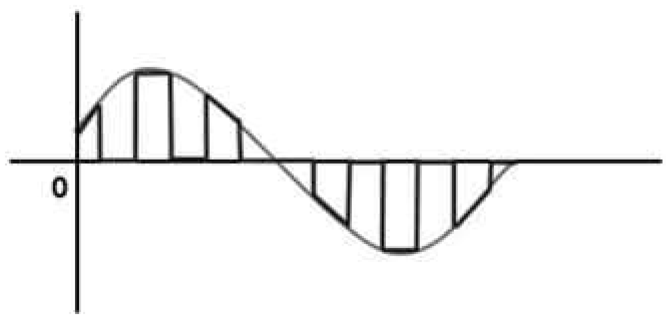


Fig. 5.2

Sample & Hold circuit

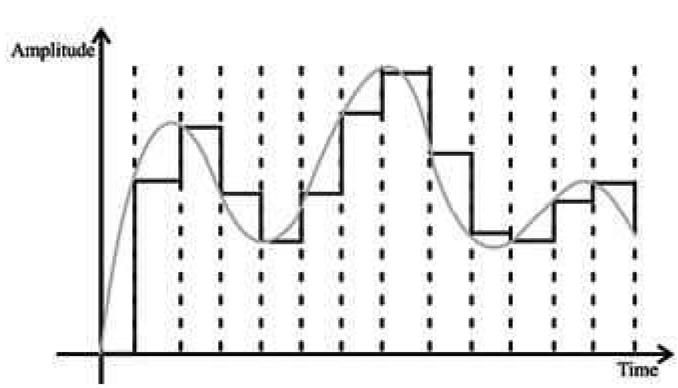
In electronics, a sample and hold circuit is used to interface real-world signals, by changing analogue signals to a subsequent system. The purpose of this circuit is to hold the analogue value steady for a short time while the converter or other following system performs some operation that takes a little time.

Sampling mode:

In this mode, the switch is in the closed position and the capacitor charges to the instantaneous input voltage.

Hold mode:

In this mode, the switch is in the open position. The capacitor is now disconnected from the input. As there is no path for the capacitor to discharge, it will hold the voltage on it just before opening the switch. The capacitor will hold this voltage till the next sampling instant



Sample and Hold Waveform - Fig. 5.3

2. **Pulse Width Modulation (PWM):**
   In PWM system the width of the pulse is varied in accordance with the instantaneous level of the modulating signal.
3. **Pulse Position Modulation (PPM):**
   In PPM System, the position of the pulse relative to the zero reference level is varied in accordance with the instantaneous level of the modulating signal.

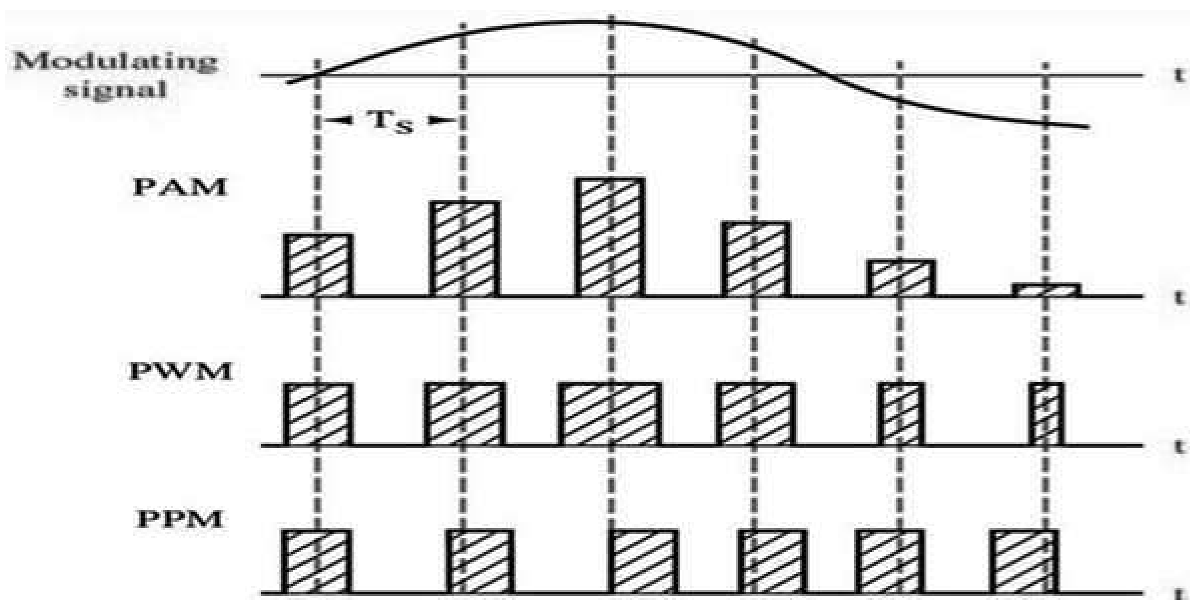


Fig. 5.4

Apparatus Required:

- PAM, PPM, PWM Mod. /Demod. Experiment Kit.
- Function Generator.
- A CRO / DSO

❖ STUDY OF PULSE AMPLITUDE MODULATION USING NATURAL & FLAT TOP SAMPLING AND SAMPLE & HOLD CIRCUIT

Connection Block Diagram:

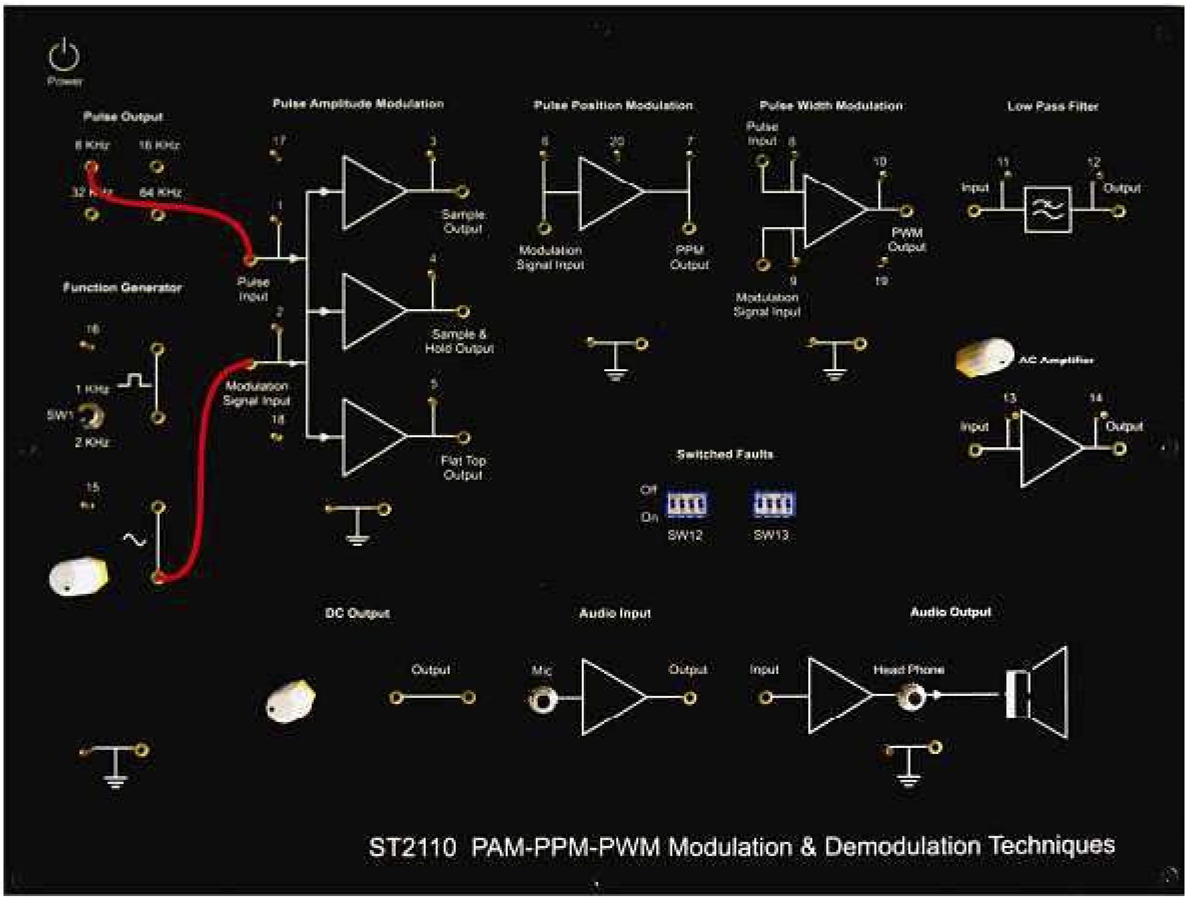


Fig. 5.5

Procedure:

1. Connect the circuit as shown in Figure 5.
   a. Output of sine wave to modulation signal input in PAM block keeping theswitch in 1 KHz position.
   b. 8 KHz pulse output to pulse input.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the outputs at TP (3 & 5) these are natural & flat top outputsrespectively.
4. Observe the difference between the two outputs.

5. Observe the output of sample & hold circuit at TP4
6. Vary the amplitude potentiometer and frequency change over switch & observethe effect on the outputs.
7. Vary the frequency of pulse, by connecting the pulse input to the 4 frequenciesavailable i.e. 8, 16, 32, 64 kHz in Pulse output block.
8. Switch 'On' fault No. 1, 2, 3, 4 one by one & observe their effect on PulseAmplitudeModulation output and try to locate them.
9. Switch 'Off' the power supply.

❖ **<u>Study of Pulse Amplitude Modulation & Demodulation with Sample, Sample & Hold & Flat Top</u>**

Connection Block Diagram:

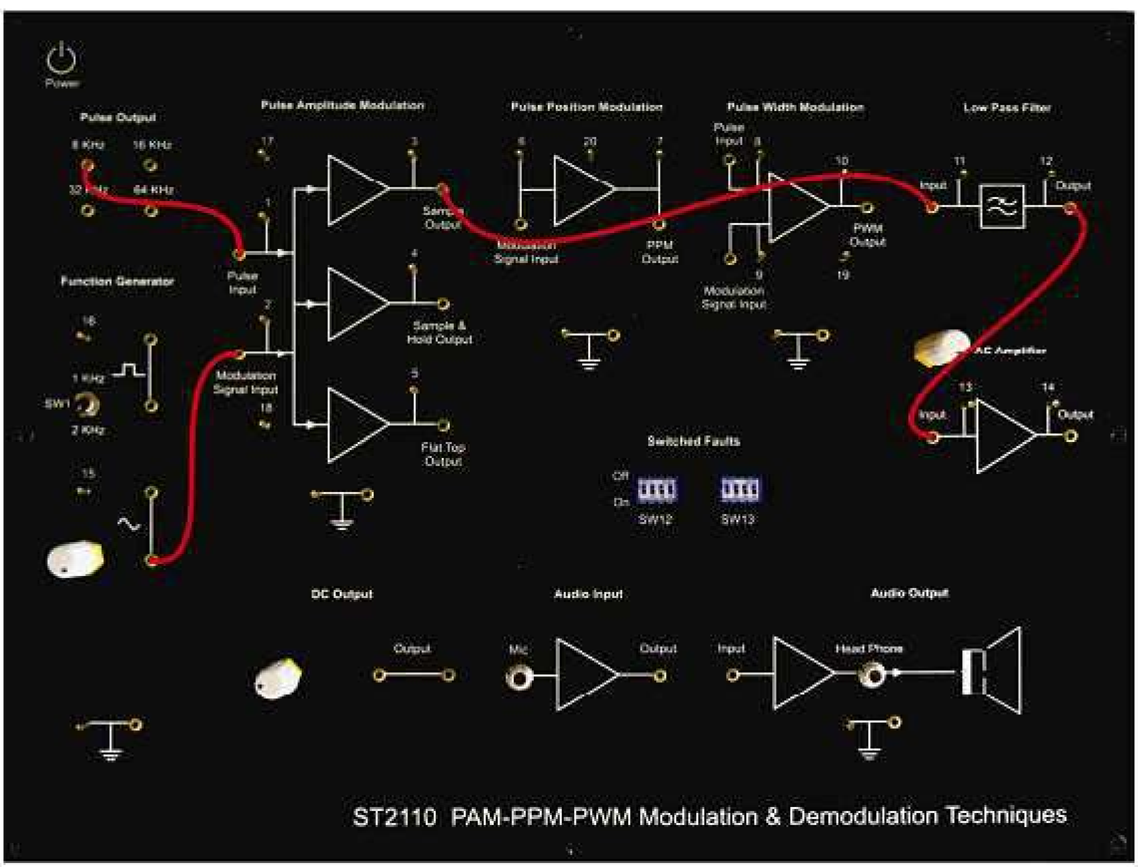


Fig. 5.6

Procedure:

1. Connect the circuit as shown in Figure 5.6.
   a. Output of sine wave to modulation signal IN in PAM block keeping the switch in 1 KHz position.
   b. 8 KHz pulse output to pulse input.
   c. Connect the sample output to low pass filter input.
   d. Output of low pass filter to input of AC amplifier. Keep the gain pot in AC amplifier block in anti-clock wise position.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the outputs at TP (3 & 5) these are natural & flat top outputs respectively.
4. Observe the difference between the two outputs.
5. Vary the amplitude potentiometer and frequency change over switch & observe the effect on the two outputs.
6. Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.
7. Switch 'On' fault No. 1, 2, 3, 4 one by one & observe their effect on Pulse Amplitude Modulation output and try to locate them.
8. Monitor the output of AC amplifier. It should be a pure sine wave similar to input.
9. Vary the amplitude of input, the amplitude of output will vary.
10. Similarly connect the sample & hold & flat top outputs to low pass filter and see the demodulated waveform at the output of AC amplifier.
11. Switch 'On' the switched faults No. 1, 2, 3, 4, 5 & 8 one by one and see their effects on output.
12. Switch 'Off' the power supply.

Experimental Data:

Table 1

Sl.No.	Modulating Signal		Sampling Pulse		Modulated (Sampled) Output		Demodulated (Reconstructed) Output	
	Freq. (f_m)	Ampl. (A_m)	Freq. (f_s)	Period T_s	Time between two sampled value (T_s')	Pulse Amplitude (A_s')	Freq. (f_{Dm})	Ampl. (A_{Dm})
1.	f_m	A_{m1}						
2.	f_m	A_{m2}						
3.	f_m	A_{m3}						

LPF Response:

Table 2

Sl.No	Input Signal		Output Signal	
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)
1.	f_1	A_{in}	f_o	A_1
2.	f_2	A_{in}	f_o	A_2
3.	f_3	A_{in}	f_o	A_2
n.	f_n	A_{in}	f_o	A_n

Linearity:

Table 3

Sl.No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_o	A_{o1}	M_1
2.	f_1	A_2	f_o	A_{o2}	M_2
3.	f_1	A_2	f_o	A_{03}	M_3
n.	f_1	A_n	f_o	A_{on}	M_n

Table 4

Sl.No	Input Signal		Output Signal		A_o/A_{in} (M)
	Freq. (f_{in})	Ampl. (A_{in})	Freq. (f_o)	Ampl. (A_o)	
1.	f_1	A_1	f_{o1}	A_{o1}	M_1
2.	f_2	A_1	f_{o2}	A_{o2}	M_2
3.	f_3	A_1	f_{o3}	A_{03}	M_3
n.	f_n	A_1	f_{on}	A_{on}	M_n

Data Analysis:

1. Calculate the ratio of amplitudes of input and output signals and show the linearity of LPF.
2. Find the cut off frequency of LPF.
3. Tally sampling time and time between two consecutive samples at modulator output.
4. Tally the frequency and amplitude of input signal and output signal of the system.

Some Sample questions:

1. Why flat top sampling is better than natural sampling?
2. What do you understand by sample and hold circuit?
3. What is the importance of doing sample and hold of signals?

EXPERIMENT # 5a

Objective: Study of PPM

Connection Block Diagram:

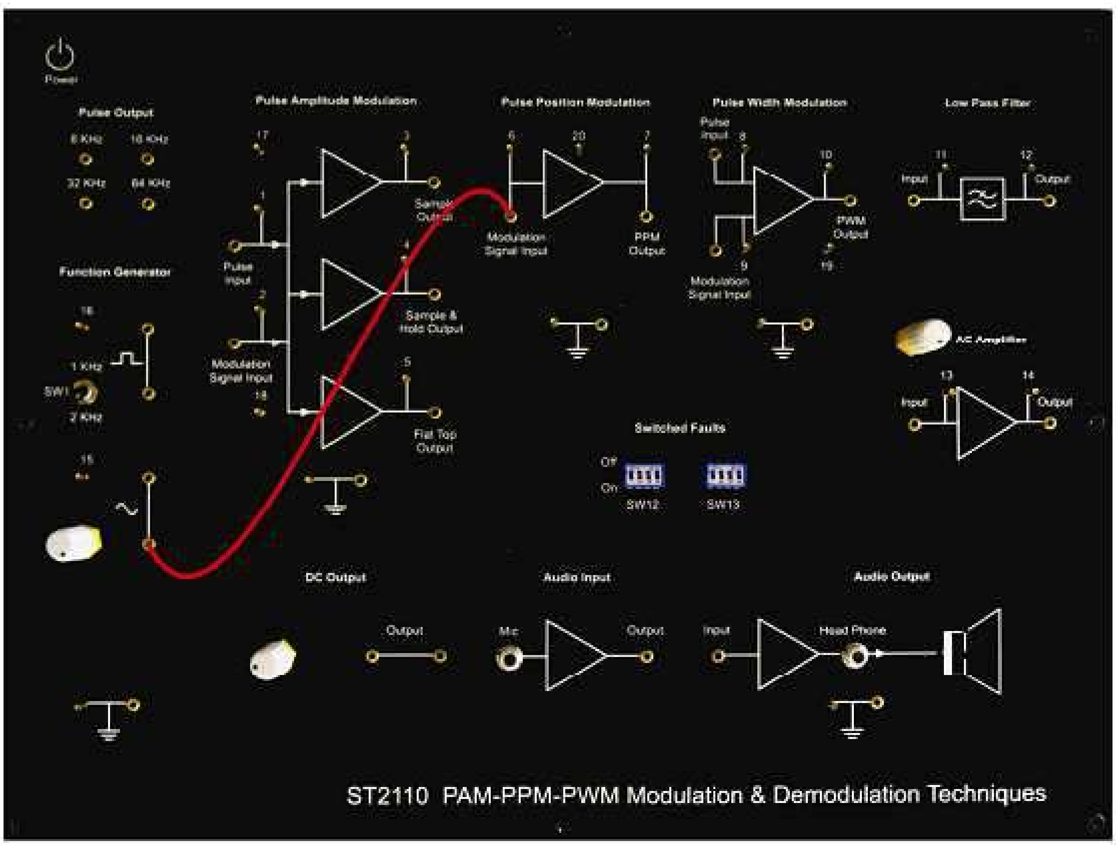


Fig. 5.7a

Procedure:

1. Connect the circuit as shown in Figure 5.7a and also described below for clarity.
    a. Input of pulse position modulation blocks to sine wave output of FG block.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the pulse position modulated waveform at the pulse position modulation block output.
4. Vary the amplitude of sine wave and observe the pulse position modulation, keep the amplitude preset in center. Here you can best observe the pulse modulation.
5. Switch 'On' fault No. 1, 2, & 6 one by one & observe their effects in pulse position modulation output and try to locate them.
6. Switch 'Off' the power supply.

PULSE POSITION DEMODULATION:

Connection Block Diagram:

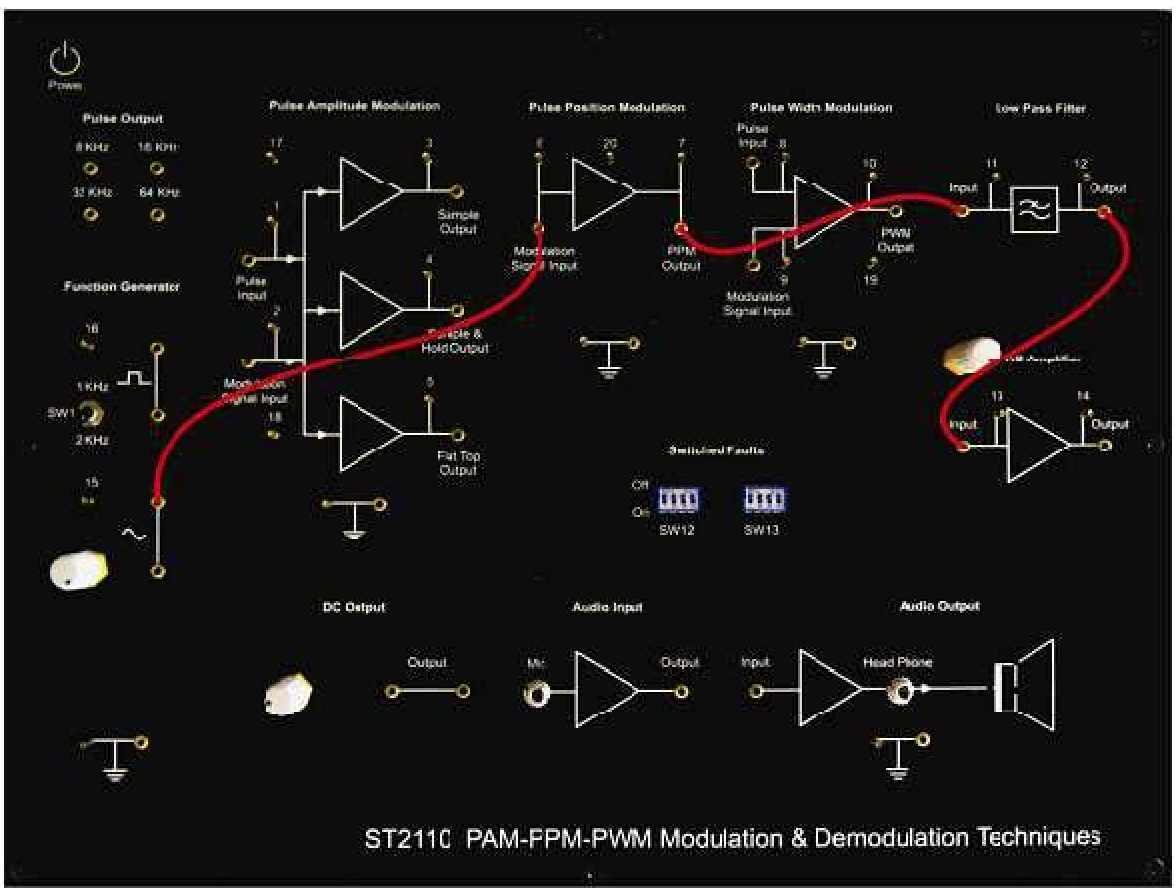


Fig. 5.8a

Procedure:

1. Connect the circuit as shown in Figure 5.8a and also described below for clarity.
   a. Sine wave of 1 KHz to input of PPM block.
   b. Output PPM block to input of low pass filter.
   c. Output of low pass filter to input of AC amplifier.
   d. Keep the gain potentiometer in amplifier block at maximum position.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the waveform at the TP12 output of low pass filter block.
4. Then observe the demodulated output at TP14 output of AC amplifier.
5. Switch 'On' fault No. 1, 2, 6 & 8 one by one & observes their effect on demodulated waveform & tries to locate them.
6. Switch 'Off' the power supply.

Some Sample questions:

1. What do you understand by PPM modulation and demodulation?
2. Why modulation is required in digital communication?
3. Why low pass filters are required in PPM demodulation process?
4. Why PPM is not generally used in digital communication system?

EXPERIMENT # 5b

Objective: Study of PWM using different Sampling Frequency

Connection Block Diagram:

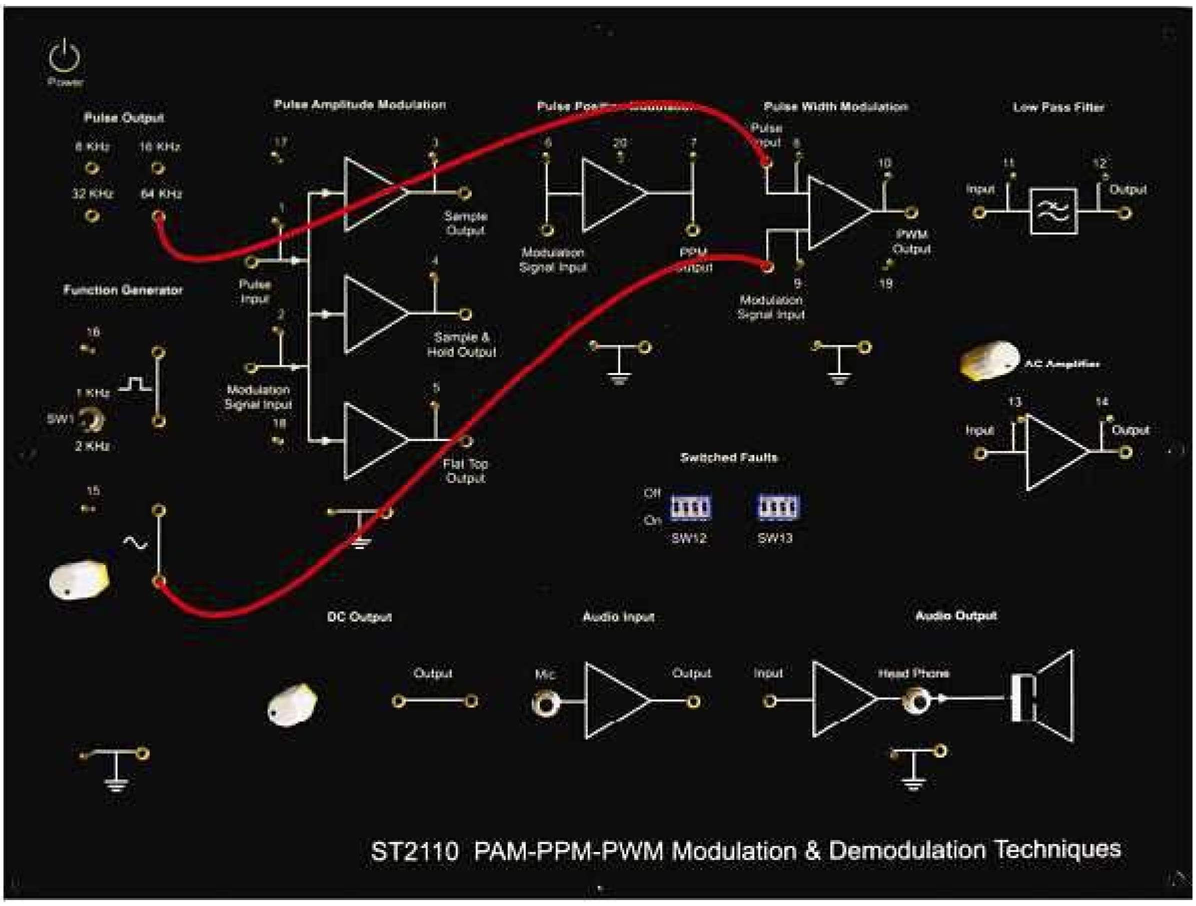


Fig. 5.9b

Procedure:

1. Connect the circuit as shown in Figure 9b and also described below for clarity.
   a. 1 KHz sine wave output of function generator block to modulation input of PWM block.
   b. 64 KHz square wave output to pulse input of PWM block.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the output of PWM block.
4. Vary the amplitude of sine wave and see its effect on pulse output.

5. Vary the sine wave frequency by switching the frequency selector switch to 2 KHz.
6. Also, change the frequency of the pulse by connecting the pulse input to different pulse frequencies viz. 8 KHz, 16 KHz, 32 KHz and see the variations in the PWM output.
7. Switch 'On' fault No. 1, 2, & 5 one by one & observes their effect on PWM output and tries to locate them.
8. Switch 'Off' the power supply.

PULSE WIDTH DEMODULATION

Connection Block Diagram:

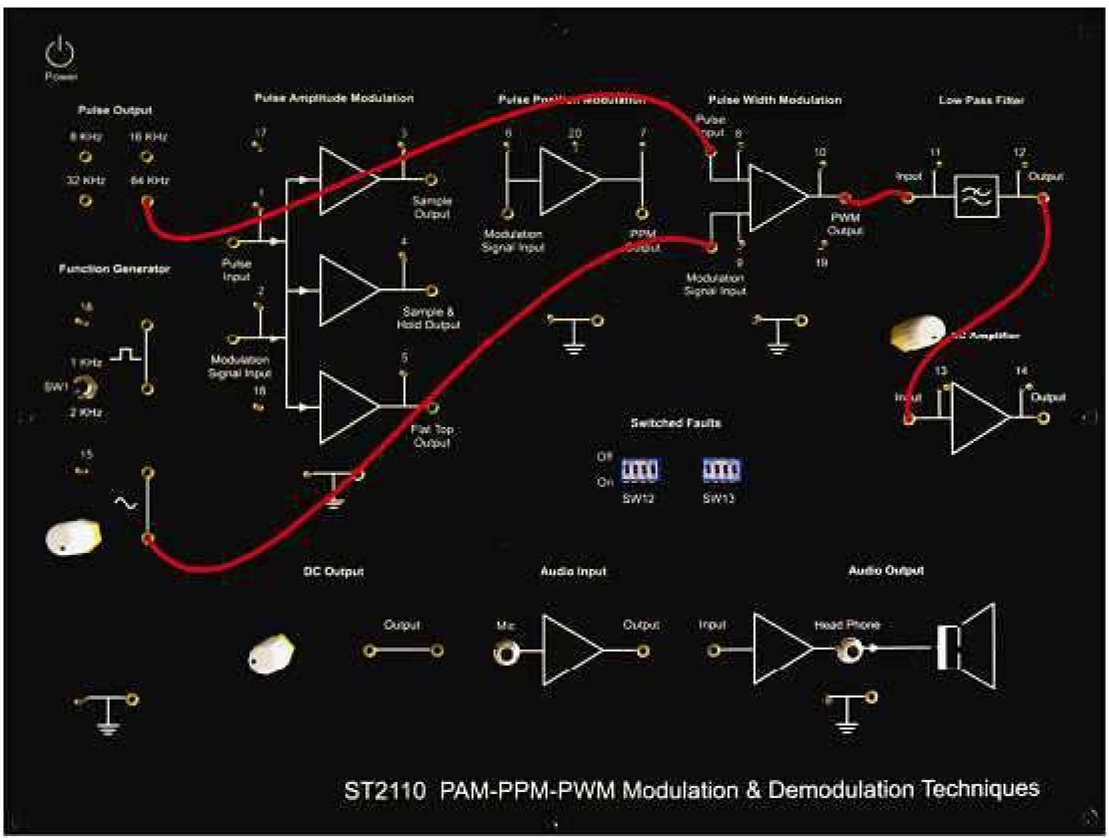


Fig. 5.10b

Procedure:

1. Connect the circuit as shown in Figure 5.10b and also described below for clarity.
   a. 1 KHz sine wave output of function generator block to modulation input of PWM block.
   b. 64 KHz square wave output to pulse input.
   c. Output of PWM to input of low pass filter.
   d. Output of low pass filter to input of AC Amplifier.
2. Switch 'On' the power supply & oscilloscope.
3. Observe the output of low pass filter and AC amplifier respectively to understand the demodulation of pulse width demodulation waveform in detail.
4. Vary the amplitude and frequency of sine wave and observe its effect on the demodulated waveform.
5. Now, connect the pulse input in the pulse width modulation block to the different frequencies available on board viz. 8, 16, 32 KHz and observe their demodulated waveforms.
6. Try varying the amplitude of sine wave signal; you will observe that the output signal varies similarly.
7. Switch 'On' fault no, 1, 2, 5 & 8 one by one at a time. Observe their effects on final output and try to locate them.
8. Switch 'Off' the power supply.

Experimental Data:

<table>
<tr><td rowspan="3">Sl. No.</td><td colspan="3">Modulating Signals</td><td colspan="2">Modulated Output</td><td colspan="2">Ampl. to Width Ratio</td><td colspan="2">Demodulated Output</td></tr>
<tr><td rowspan="2">Freq. (f_m)</td><td colspan="2">Ampl. (A_m)</td><td colspan="2">Width of Rectangular Pulse</td><td rowspan="2">Time</td><td rowspan="2">Ratio</td><td rowspan="2">Freq. (f_D)</td><td rowspan="2">Ampl. (A_D)</td></tr>
<tr><td>Time</td><td>Ampl.</td><td>Time</td><td>Width</td></tr>
<tr><td rowspan="3">1.</td><td rowspan="3"></td><td>T_1</td><td>M_1</td><td>T_1</td><td>W_1</td><td>T_1</td><td>M_1/ W_1</td><td rowspan="3"></td><td rowspan="3"></td></tr>
<tr><td>T_2</td><td>M_2</td><td>T_2</td><td>W_2</td><td>T_2</td><td>M_2/ W_2</td></tr>
<tr><td>T_3</td><td>M_3</td><td>T_3</td><td>W_3</td><td>T_3</td><td>M_3/ W_3</td></tr>
<tr><td rowspan="3">2.</td><td rowspan="3"></td><td></td><td></td><td></td><td></td><td></td><td></td><td rowspan="3"></td><td rowspan="3"></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
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<tr><td rowspan="3">3.</td><td rowspan="3"></td><td></td><td></td><td></td><td></td><td></td><td></td><td rowspan="3"></td><td rowspan="3"></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td></td><td></td><td></td><td></td><td></td><td></td></tr>
</table>

Discussion:

Write briefly your comments about the above experiment.

Some Sample questions:

1. What do you understand by pulse width modulation?
2. What is the effect of varying sampling frequency on pulse width modulation signal?
3. Design a P.W.M circuit and plot a graph for signal voltage vs Pulse width and draw P.W.M waveform for a sinusoidal modulating signal.
4. Design a P.W.M. circuit taking suitable signals and show that width of the output rectangular pulse varies linearly with the amplitude of modulating signal

Precautionary Measure to be taken:

4. Ensure that equipment or training kit switch is kept off While connect it to main power supply.
5. For connecting any signal from one equipment to equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.
6. Handle gently all the necessary button or knob in the equipment avoiding all other button or knobs which are not required to be adjusted for the equipment.
7. For unusual spark of burning smell, immediately switch off the main supply to the kit.
8. Ensure that, a signal is connected to an appropriate junction destined for it.
9. Always cover the equipment for dust protection after the experiment.

EXPERIMENT # 6

Title: STUDY OF TDM-PAM SYSTEM

Objective:

- Study of complete TDM-PAM System and the overall effect of theindividualparameter/ mode on the communication system (Transmitting).
- Study the working of a TDM-PAM Transmitter and Receiver (Receiving)

Theory:

Time Division Multiplexing Pulse Amplitude Modulation is communication system where the message signal modulates using Pulse Amplitude Modulation and multiple accesses is provided using Time Division Multiplexing.Multiplexing is the process of combining signals fromdifferent information sourcesso that they can be transmitted over a common channel.

In this technique, signals from different sources are sampled sequentially and transmitted serially. In this case i.e output of each source is either a PAM signal or a digital signal that means the signals are discrete in nature. So signals from each source are generated in a particular frequency. The sequential sampling implies that bits are put for transmission in a single channel from one source after another. For example let us consider that first bit from each of the sources are transmitted one after another until the last source is sampled and then is transmitted the second bit of first source and so on. Selection of sources sequentially is established by a digital multiplexer.

In the receiving end, on the other hand, the serial data are sent to the specific destinations sequentially. This is achieved using a digital multiplexer.

A common clock synchronizes all operations in the transmitter and receiver. The block diagram of a Time Division Multiplexer is shown in fig. 6.1.

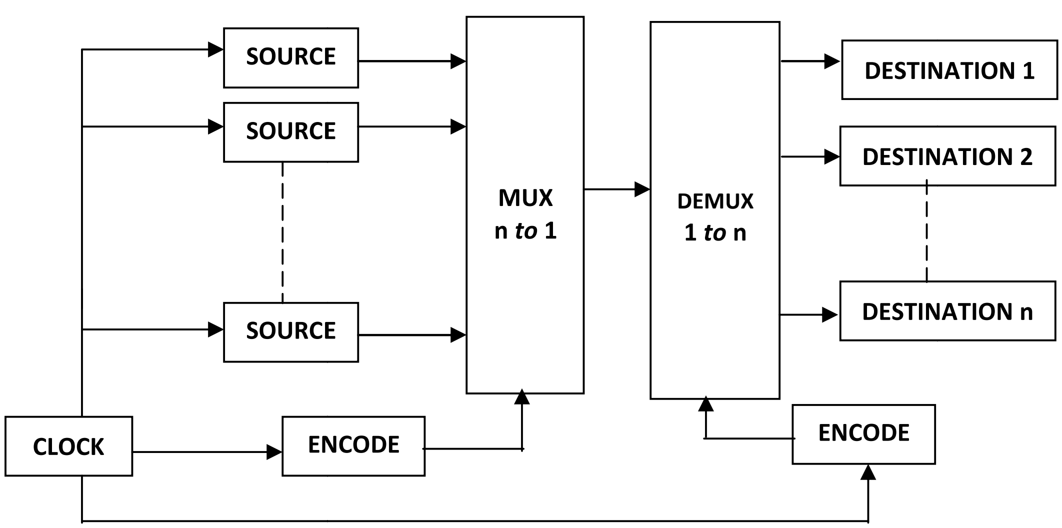
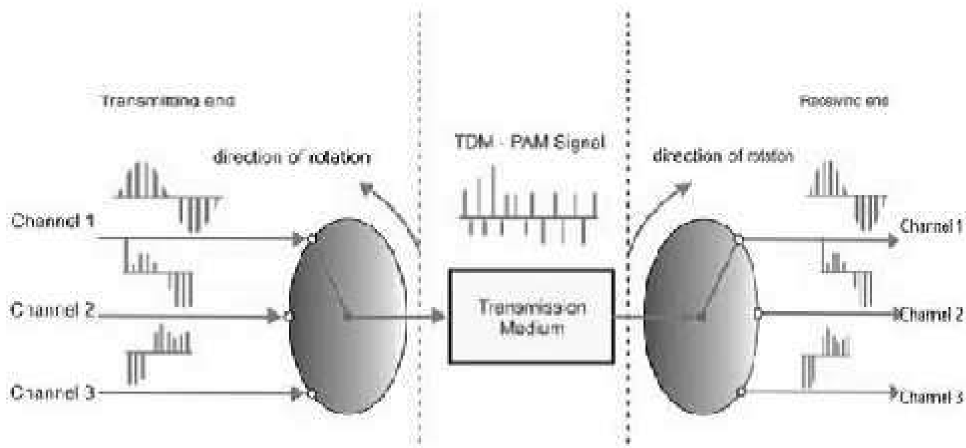


Fig. 6.1.

The fact utilized in TDM technique is that there are large intervals between the message samples. The samples from the other sources can be placed within these time intervals. Thus every sample is separated from other in time domain. The time division multiplexing system can be simulated by two rotating switches, one at transmitter and the other at receiver. (See figure 6.2) The two wipers rotate and establish electrical contact with one channel at a time.

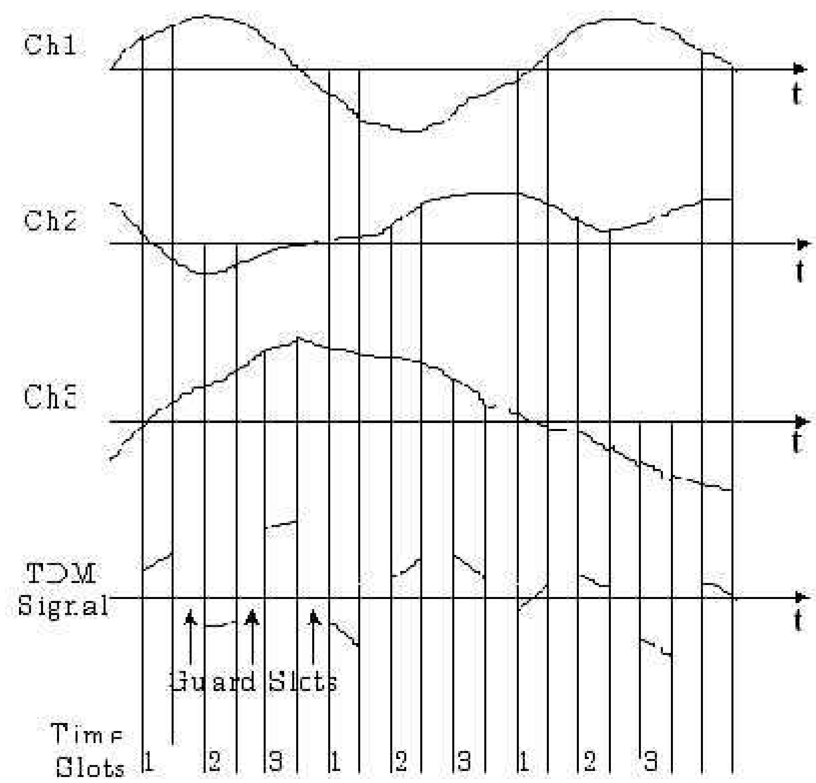


Principle operation of TDM-PAM system

Fig. 6.2.

Each signal is sampled over one sampling interval and transmitted one after the other along a common channel. Thus part of message 1 is transmitted first followed by part of message 2, message 3 and then again message 1 so on.
The switches connect the transmitter and the receiver to each of the channels in turn for a specific interval of time. In effect each channel is sampled and the sample is transmitted when the switches are in the channel 1 position, channel 1 forms a PAM channel with an LPF for reconstruction, and so on for channels 2 and 3. The result is that the amplitudes samples from each channel share the line sequentially, becoming interleaved to form a complex PAM wave, as shown above.
A major problem in any TDM system is the synchronization of the transmitter and receiver timing circuits. The transmitter and receiver must switch at the same time and frequency. Also SW1 must be in the channel 1 position when SW2 is in the channel 1 position, so that the switches must be synchronized in position also.
In a system that uses analogue modulation (PAM) the time slots are separated by guard slots to prevent crosstalk between channels.

This figure 6.3 shows the waveforms produced during the operation of the PAM-TDM system.



Timing waveform of TDM-PAM system

Fig. 6.3.

It can be anticipated from above process that the receiver switch has to follow two constraints:

1. It must rotate at the same rate as the transmitter switch.
2. It must start at the same time as the transmitting switch and it must establish electrical contact with the same channel no. as that of the transmitter. If these two conditions are met, the receiver is said to be in synchronization with transmitter. If constraint one is not met, the samples of different sources would get mixed at the receiver. If constraint two is not met, the information from source 1 will be received by some other channel which is not intending to accept the information from that particular channel. To establish synchronization, the receiver needs to know:
    a. Frequency/ rate of operation at transmitter.
    b. Sample identification.

Apparatus Required:

- TDM PAM Transmitter and Receiver ST2152.
- DC Power Supply for Kit.
- A CRO / DSO

MODULATION:

Connection Diagram:

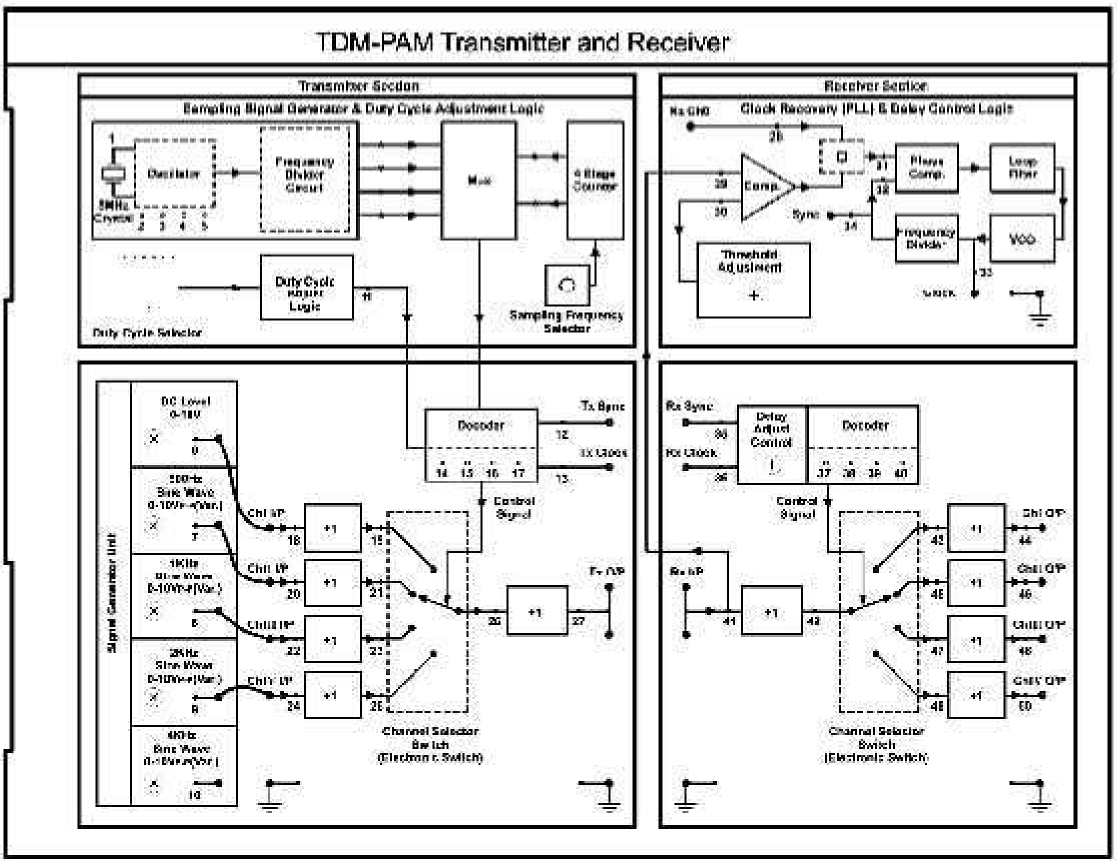


Fig. 6.2

Procedure:

Initial setup of Tech book:

Function Generator pot direction	:	Anti-clock wise
Duty cycle Position	:	5
Delay control	:	Anti-clock wise
Comparator Threshold level	:	Clockwise
Frequency Divider circuit O/P	:	Highest frequency

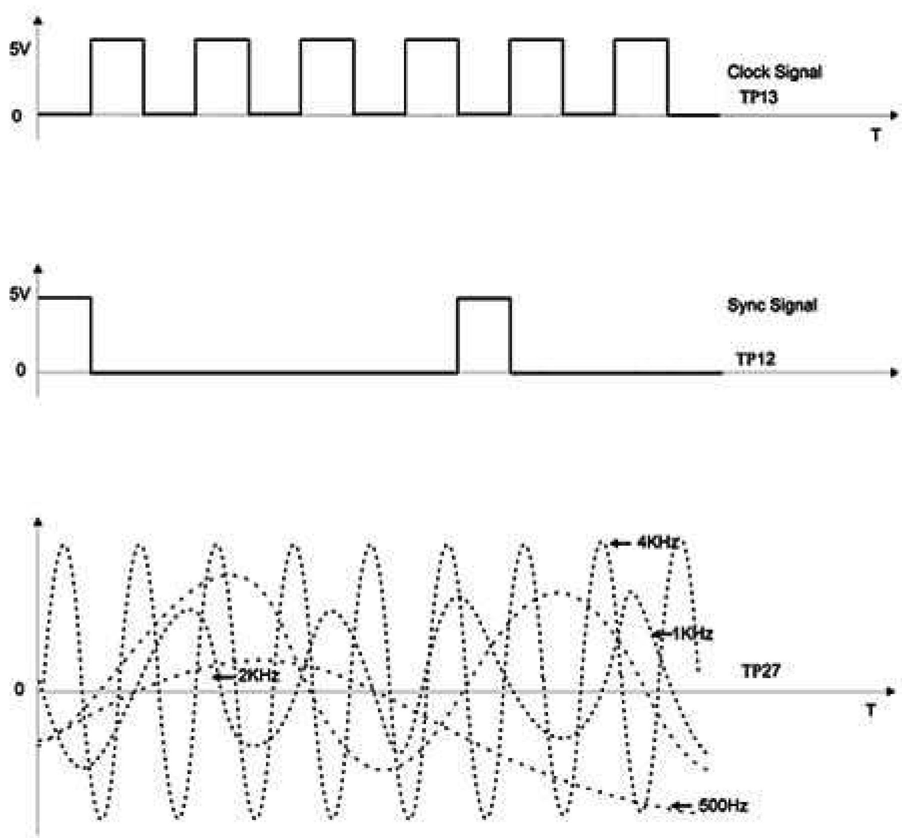
1. Connect the power cord to the tech book. Keep the power switch in 'Off' position.
2. Switch 'On' the tech book's Power Supply & Oscilloscope.
3. Connect BNC connector to the CRO and to the tech book's output port.
4. Make following connections with patch chords:

    a. 500Hz to CH1 I/P socket : 2 V
    b. 1 KHz to CH2 I/P socket : 3 V
    c. 2 KHz to CH3 I/P socket : 4 V
    d. 4 KHz to CH4 I/P socket : 5 V

5. Connect the Oscilloscope channel CH1 at TP27, draw the waveforms.
6. Connect the Oscilloscope channel CH1 at TP13 clock signal and channel CH2 at TP27, draw the waveforms.

Observations /Analysis:

1. The output waveform coming at Tx. O/P (TP27) is sampled input signal. The Transmitter Circuit samples all channels at different time intervals. The time division multiplexed samples appear at the Tx. O/P (TP27) as shown in the figure 6.3.



Clock, Sync and Input Sampled Signals

Fig. 6.3

2. Vary the amplitude of the input sine-waves by varying the potentiometers in the Signal Generator Unit. This will help in identifying which sample belongs to which input channel.

Conclusion:

In TDM-PAM system signals send as samples, control signals decided when to take samples, Sync signal synchronizes transmitter and receiver. The waveform shown in the figure 6.3 is Time Division Multiplexed Pulse Amplitude Modulated signal.

DEMODULATION:

I. Study the working of a TDM-PAM Transmitter and Receiver (Receiving):

Connection Diagram:

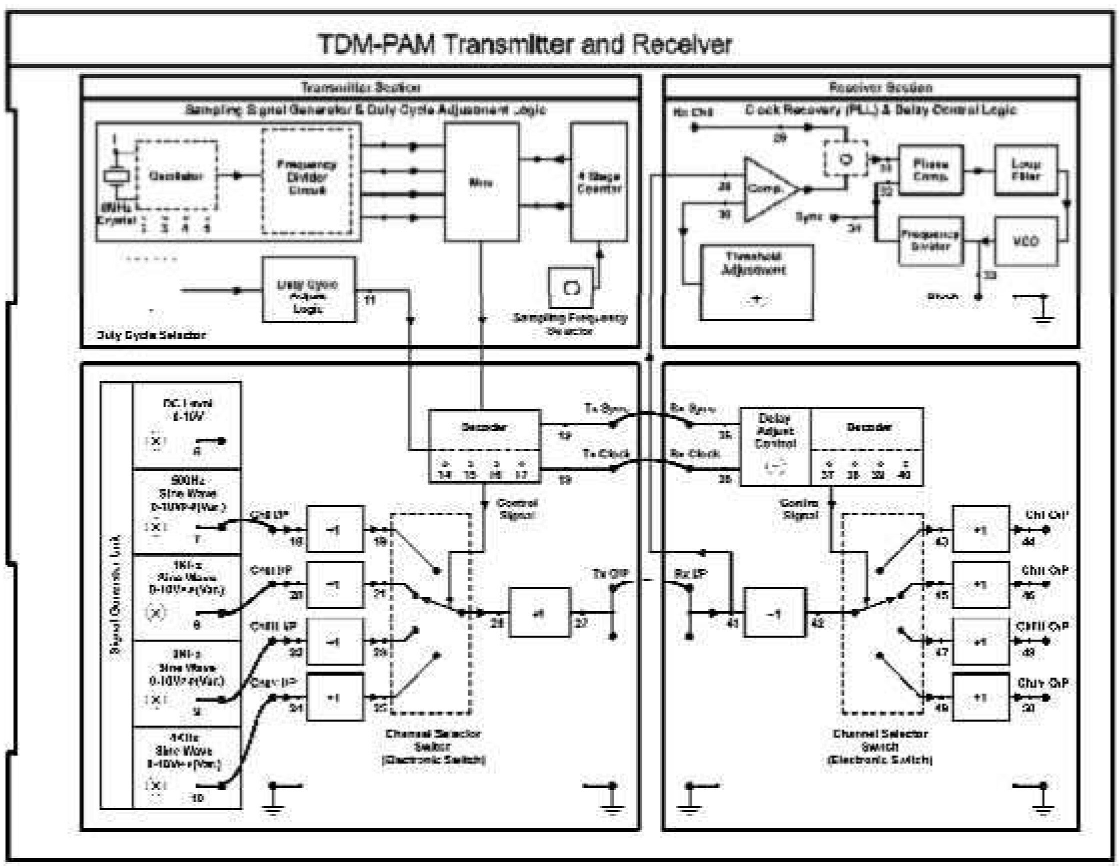


Fig. 6.4

Procedure:

Initial setup of Techbook:

Function Generator pot direction	:	Anti-clock wise
Duty cycle Position	:	5
Delay control	:	Anti-clock wise
Comparator Threshold level	:	Clockwise
Frequency Divider circuit O/P	:	Highest frequency

1. Connect the power cord to the techbook. Keep the power switch in 'Off' position.
2. Switch 'On' the techbook's Power Supply & Oscilloscope.
3. Connect BNC connector to the CRO and to the techbook's output port.
4. Make following connections with patch chords:

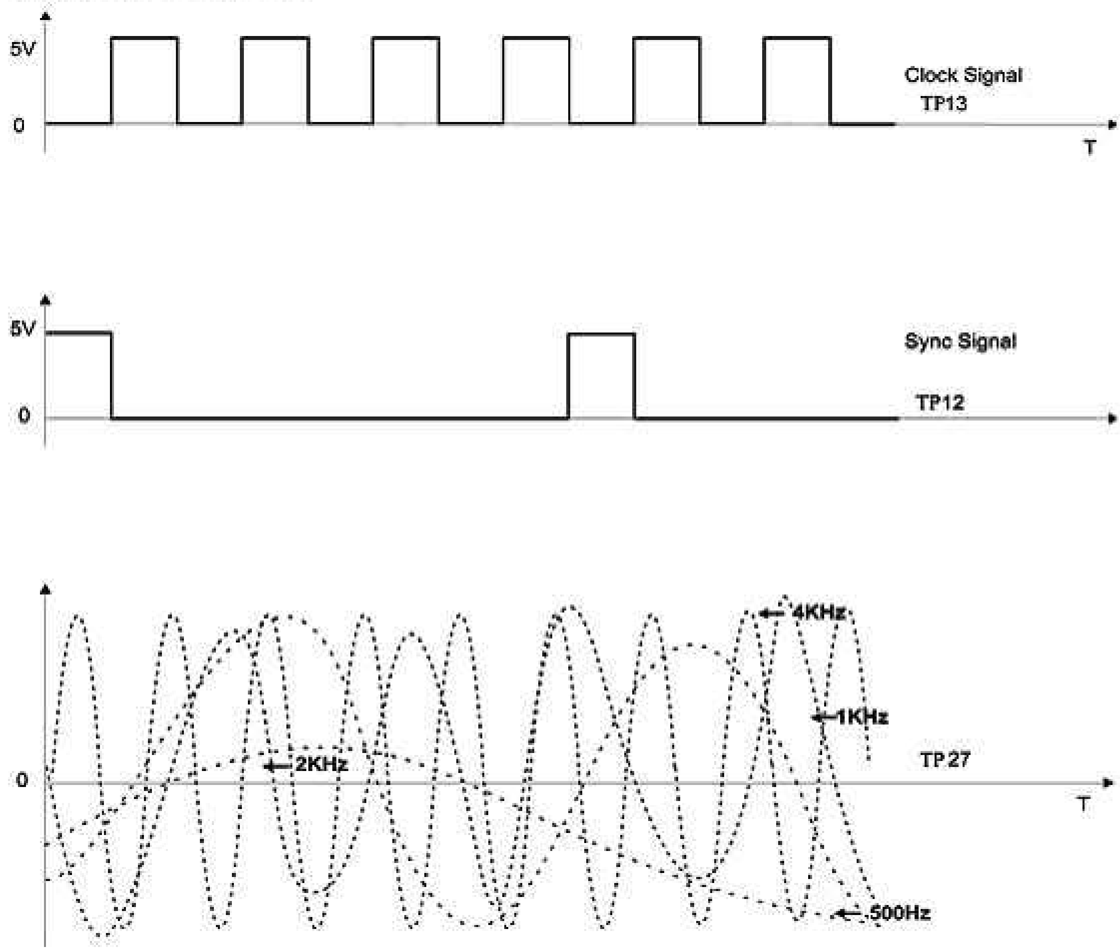
   a. 500Hz to CH1 I/P socket.
   b. 1 KHz to CH2 I/P socket.
   c. 2 KHz to CH3 I/P socket.
   d. 4 KHz to CH4 I/P socket.
   e. Tx. O/P (TP27) to Rx. I/P (TP41)
   f. Tx. Sync (TP12) to Rx. Sync (TP35)
   g. Tx. Clock (TP13) to Rx. Clock (TP36)

5. Connect the Oscilloscope channel CH1 at Tx. O/P (TP27), draw the waveforms.
6. Connect the Oscilloscope channel CH1 at TP43, TP45, TP47 and TP49 one by one and channel CH2 at TP44, TP46, TP48 and TP50 one by one respectively, and note amplitude, frequency and draw the waveforms.
7. Now vary the Duty Cycle Selector switch. Notice the effect of variation Duty Cycle Selector switch on Tx. O/P & amplitude of the Receiver's output.
8. Set the 'Duty cycle Selector' switch again in '5' position.
9. Push the Sampling Frequency Selector button at Transmitter block and observe the wave shape change in output signal.

Observations:

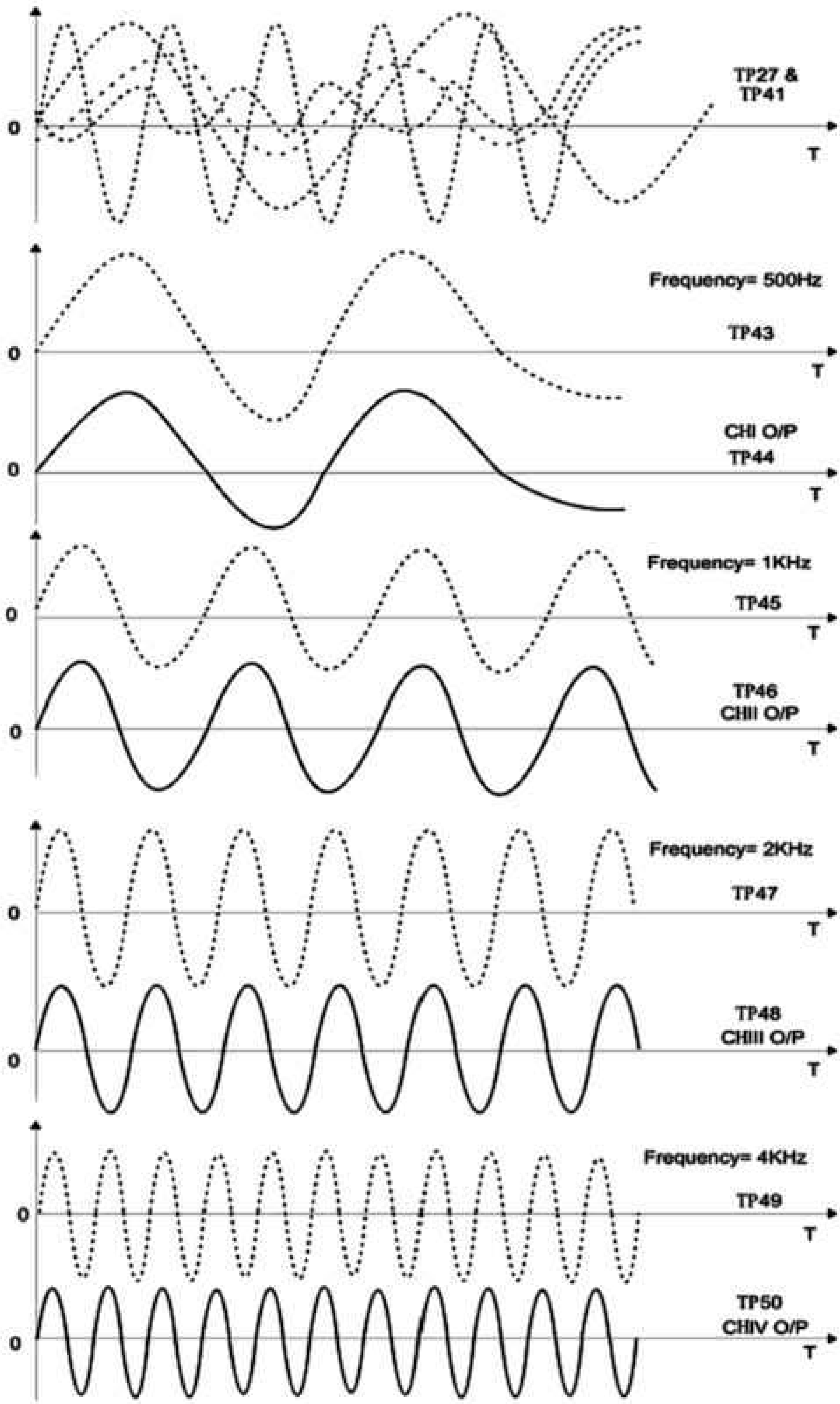
1. Tx. Clock signal is used by the receiver to synchronize its activity &Tx. Sync signal is used by the receiver to know which sample belongs to which channel.
2. The waveform at TP27 is time division multiplexed samples of each input channel as shown in the figure 6.5.
3. The output at TP43, TP45, TP47 and TP49 are Receiver's Low pass filter's inputs and TP44, TP46, TP48 and TP50 are CH1, CH2, CH3 and CH4 Outputs respectively as shown in the figure 6.6.

Observation diagram:



Clock, Sync and Input Sampled Signals

Fig. 6.5



Sampled Input and Output of Various Transmission Channels

Fig. 6.6

Conclusion:

1. *At 50% duty cycle and highest sampling frequency, the output at Receivers channel is in same order as it is in Transmitter's input.*
2. *Output voltage at Receiver will be attenuated due to Low pass filters'.*
3. *As we change the duty cycle the wave shape of the Receiver's output will be distorted or have very less amplitude due to change in On and Off time of its sampling Frequency.*
4. *The wave shape of output at Receiver's channel will be distorted due to fewer samples for better reconstruction of received signal.*

Experimental Data:

Signal	Amplitude(Peak to Peak)	Frequency
Tx Sync (TP12)		
Tx Clock (TP13)		
Transmitter output(TP27)		
Receiver input(TP41)		
CH1 I/P (TP18)		
CH1 O/P (TP44)		
CH2 I/P (TP20)		
CH2 O/P (TP46)		
CH3 I/P (TP22)		
CH3 O/P (TP48)		
CH4 I/P (TP25)		
CH4 O/P (TP50		

II. Study the working of a TDM-PAM Transmitter and Receiver at 2 Link Communication Mode i.e. Mode 2 (Receiving):

Connection Diagram:

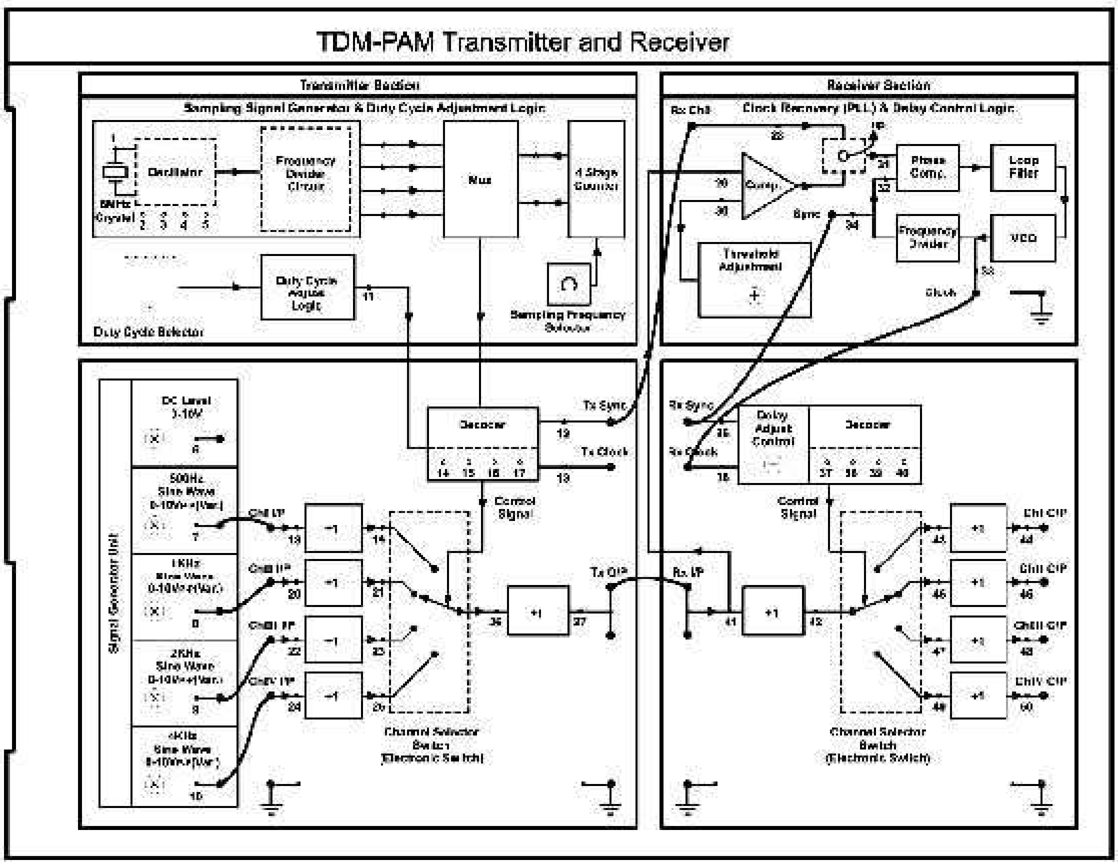


Fig. 6.7

Procedure:

Initial setup of Techbook:

Function Generator pot direction	:	Anti-clock wise
Duty cycle Position	:	5
Delay control	:	Anti-clock wise
Comparator Threshold level	:	Clockwise
Frequency Divider circuit O/P	:	Highest frequency

1. Connect the power cord to the techbook. Keep the power switch in 'Off' position.
2. Switch 'On' the techbook's Power Supply & Oscilloscope.

3. Connect BNC connector to the CRO and to the techbook's output port.
4. Make following connections with patch chords:

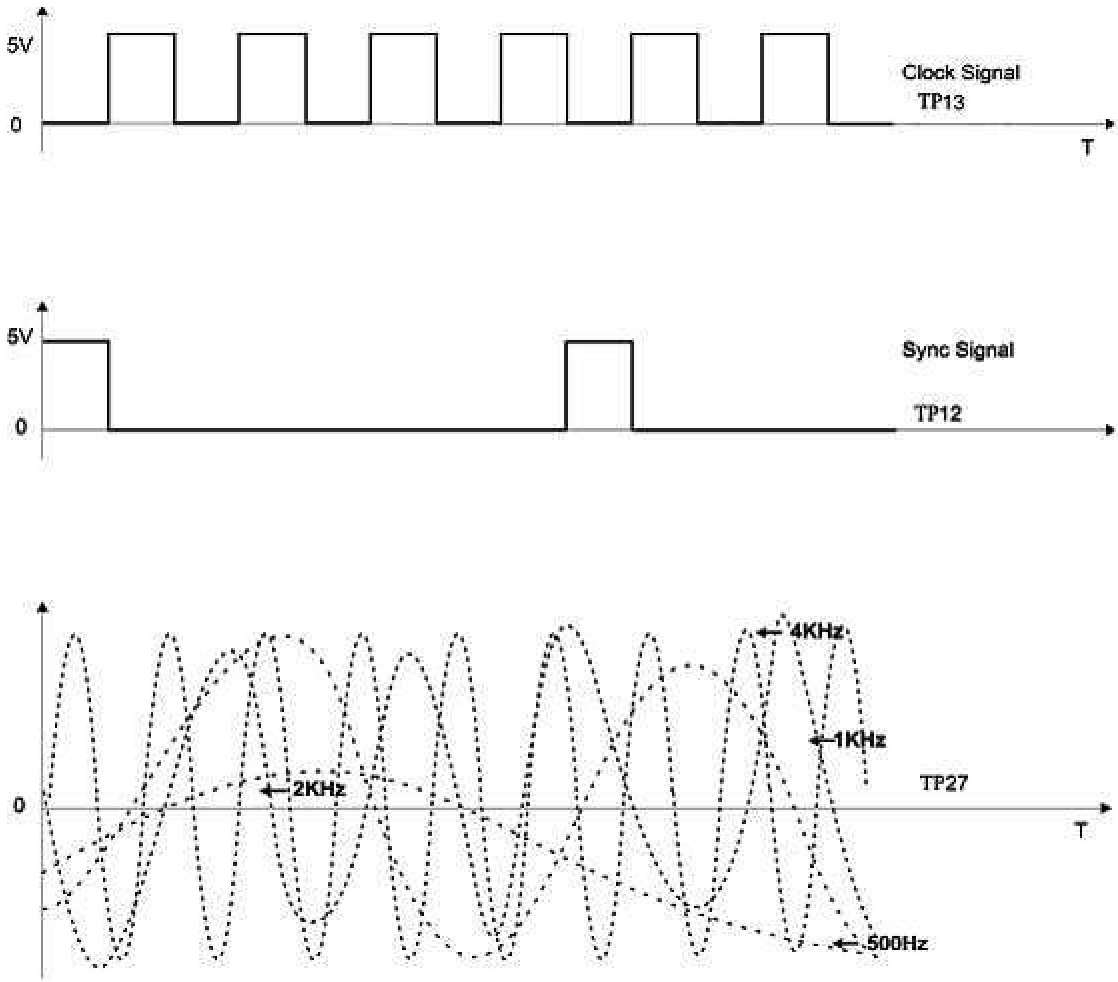
    a. 500Hz to CH1 I/P socket.
    b. 1 KHz to CH2 I/P socket.
    c. 2 KHz to CH3 I/P socket.
    d. 4 KHz to CH4 I/P socket.
    e. Tx. O/P (TP27) to Rx. I/P (TP41)
    f. Tx. Sync (TP12) to Rx. CH0
    g. Clock (TP33) to Rx. Clock (TP36)
    h. Sync (TP34) to Rx. Sync (TP35)

5. Set the level of toggle switch in Clock Recovery (PLL) & Delay Control Logic is in upward position.
6. Connect the Oscilloscope channel CH1 at TP27, draw the waveforms.
7. Connect the Oscilloscope channel CH1 at TP13 and other channel at TP12, TP31, TP32, TP33, measure amplitude, frequency and draw the waveforms.
8. Connect the Oscilloscope channel CH1 at TP43, TP45, TP47 and TP49 one by one and channel CH2 at TP44, TP46, TP48 and TP50 one by one respectively, and note amplitude, frequency and draw the waveforms.
9. Now vary the Duty Cycle Selector switch. Notice the effect of variation Duty Cycle Selector switch on Tx. O/P & amplitude of the Receiver's CH1 Output.
10. Set the 'Duty cycle Selector' switch again in '5' position.
11. Push the Sampling Frequency Selector button at Transmitter block and observe the wave shape change in output signal.

Observations:

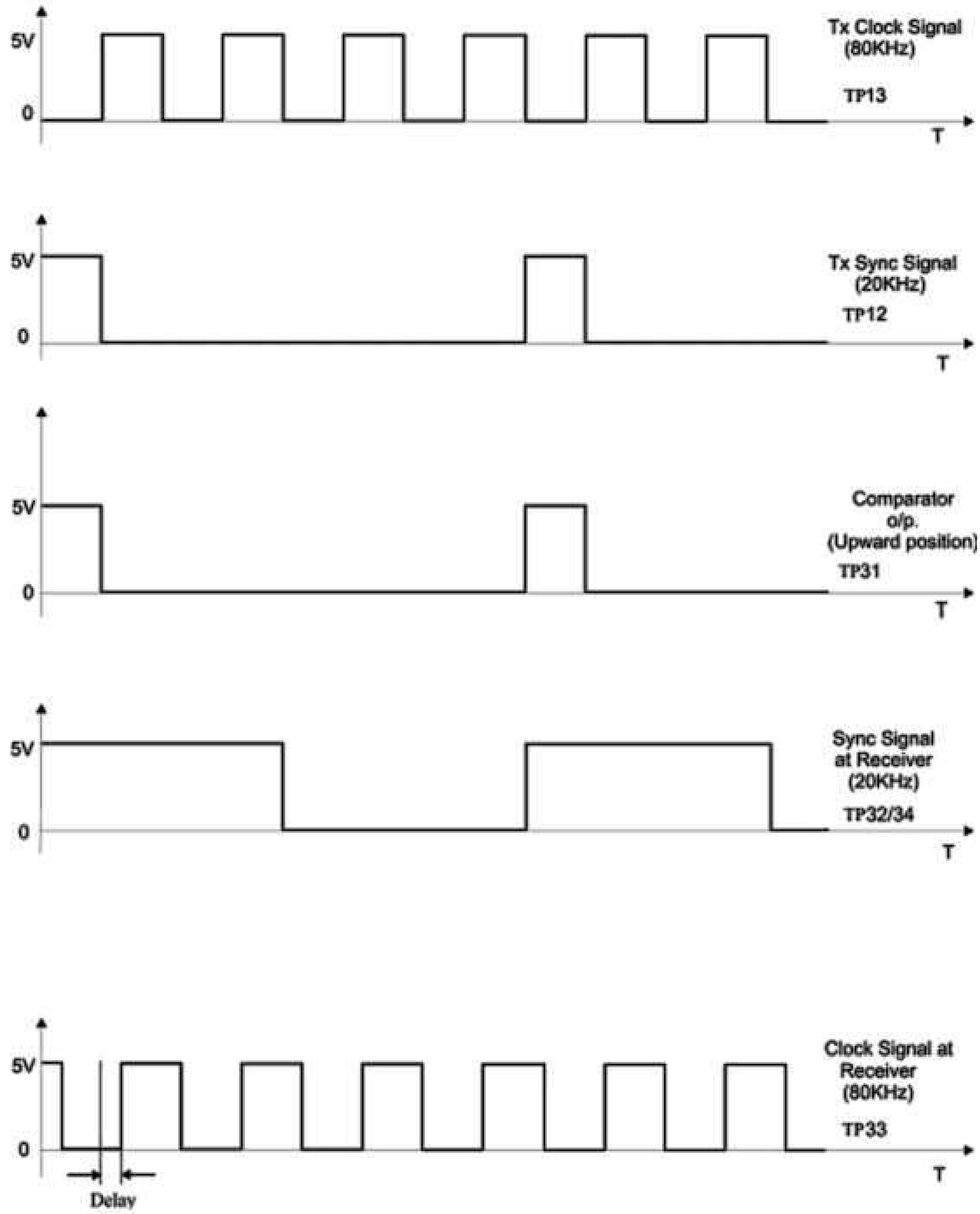
1. Tx. Clock signal is used by the receiver to synchronize its activity &Tx. Sync signal is used by the receiver to know which sample belongs to which channel and these signal are generated at the receiver using Phase Lock Loop.
2. The waveform at TP27 is time division multiplexed samples of each input channel.
3. The waveforms of transmitter clock TP13, PLL clock at TP33, transmitter Sync (receiver) TP12 and PLL (receiver) sync at TP34. Comparison among these signals.
4. The output at TP43, TP45, TP47 and TP49 are Receiver's Low pass filter's inputs and TP44, TP46, TP48 and TP50 are CH1, CH2, CH3 and CH4 outputs respectively.

Observation diagram:



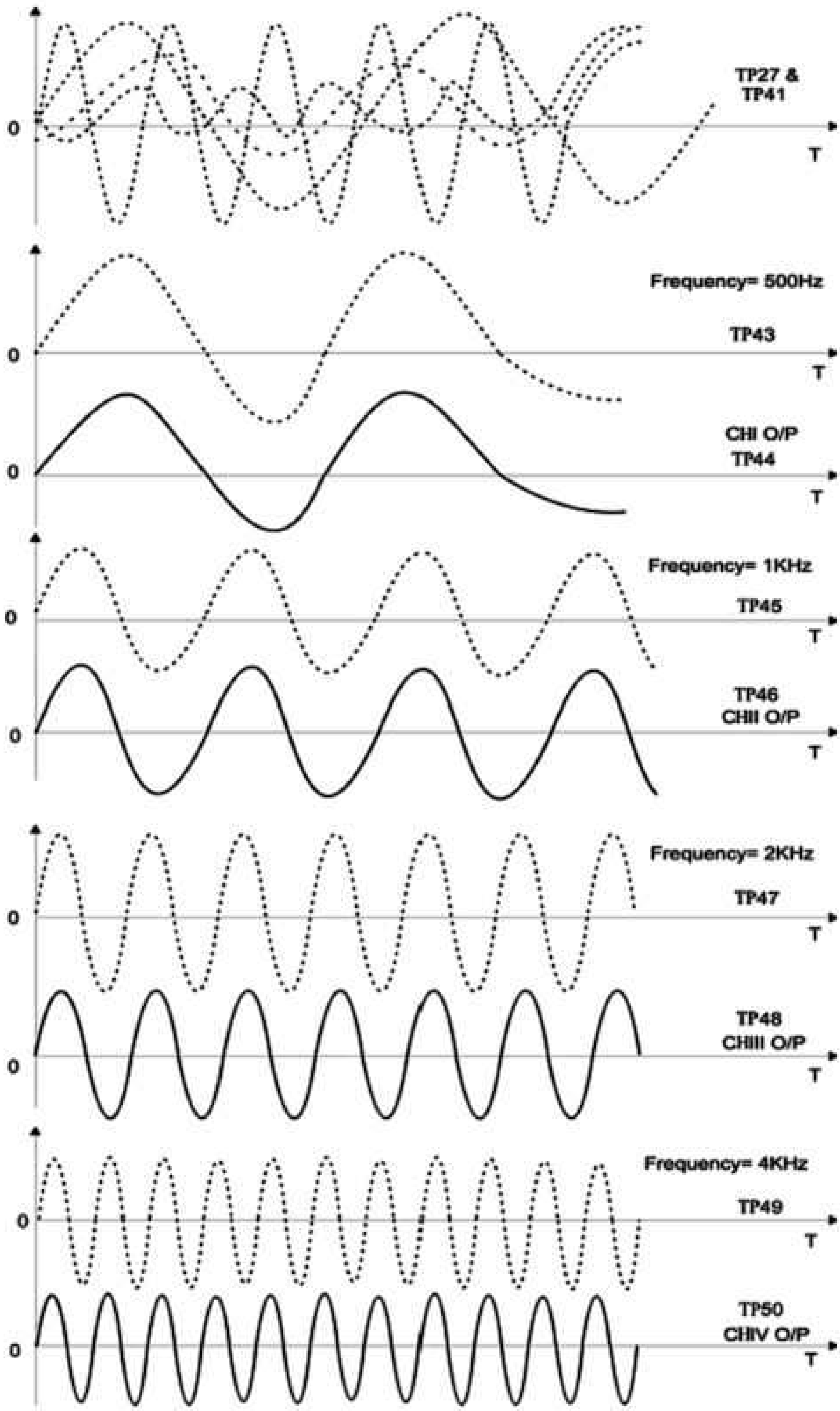
Clock, Sync and Input Sampled Signals

Fig. 6.8



Control signal, Clock signal, Sync Signal of Transmitter and Receiver

Fig. 6.9



Sampled Input and Output of Various Transmission Channels

Fig. 6.10

Experimental Data:

Signal	Amplitude(Peak to Peak)	Frequency
Tx Sync (TP12)		
Sync (TP13)		
Receiver Clock (TP36)		
Receiver Sync (TP35)		
Transmitter output(TP27)		
Receiver input(TP41)		
CH1 I/P (TP18)		
CH1 O/P (TP44)		
CH2 I/P (TP20)		
CH2 O/P (TP46)		
CH3 I/P (TP22)		
CH3 O/P (TP48)		
CH4 I/P (TP25)		
CH4 O/P (TP50		

Conclusion:

1. *The Transmitter and Receiver's Clock and Sync pulses are similar.*
2. *At 50% duty cycle and highest sampling frequency, the output at Receivers channel is in same order as it is in Transmitter's input.*
3. *Output voltage at Receiver will be attenuated due to Low pass filters'.*
4. *As we change the duty cycle the wave shape of The Receiver's output will be distorted or have very less amplitude due to change in on and off time of its sampling Frequency.*
5. *The wave shape of output at Receiver's channel will be distorted due to fewer samples for better reconstruction of received signal.*

III. <u>Study the working of a TDM-PAM Transmitter and Receiver at 1 Channel Communication Mode i.e. Mode 3 (Receiving):</u>

Connection Diagram:

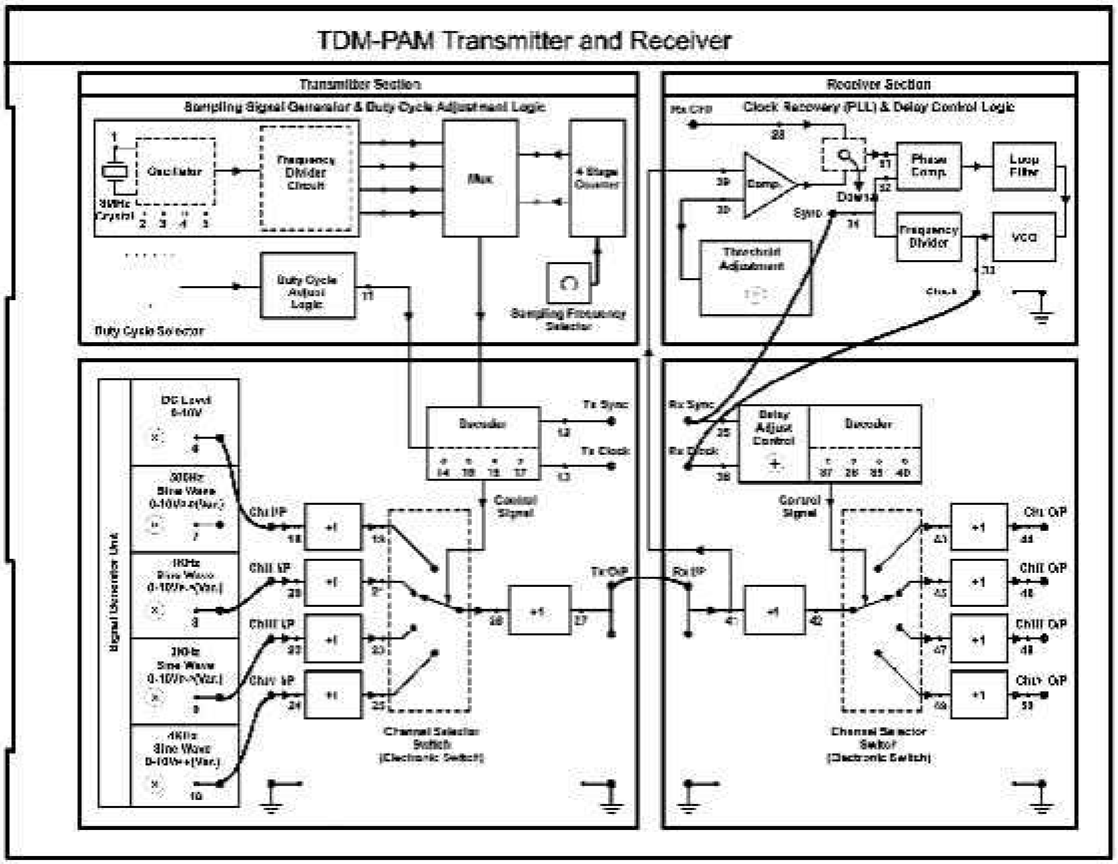


Fig. 6.11

Procedure:

Initial setup of Techbook:

Function Generator pot direction	:	Anti-clock wise
Duty cycle Position	:	5
Delay control	:	Anti-clock wise
Comparator Threshold level	:	Clockwise
Frequency Divider circuit O/P	:	Highest frequency

1. Connect the power cord to the techbook. Keep the power switch in 'Off' position.
2. Switch 'On' the techbook's Power Supply & Oscilloscope.

3. Connect BNC connector to the CRO and to the techbook's output port.
4. Ensure that the toggle level in Clock Recovery (PLL) & Delay Control Logic block is in down position.
5. Make following connections with patch chords:

    a. DC Level to CH1 I/P socket.
    b. 1 KHz to CH2 I/P socket.
    c. 2 KHz to CH3 I/P socket.
    d. 4 KHz to CH4 I/P socket.
    e. Tx. O/P (TP27) to Rx. I/P (TP41)
    f. Sync (TP34) to Rx. Sync (TP35)
    g. Clock (TP33) to Rx. Clock (TP36)

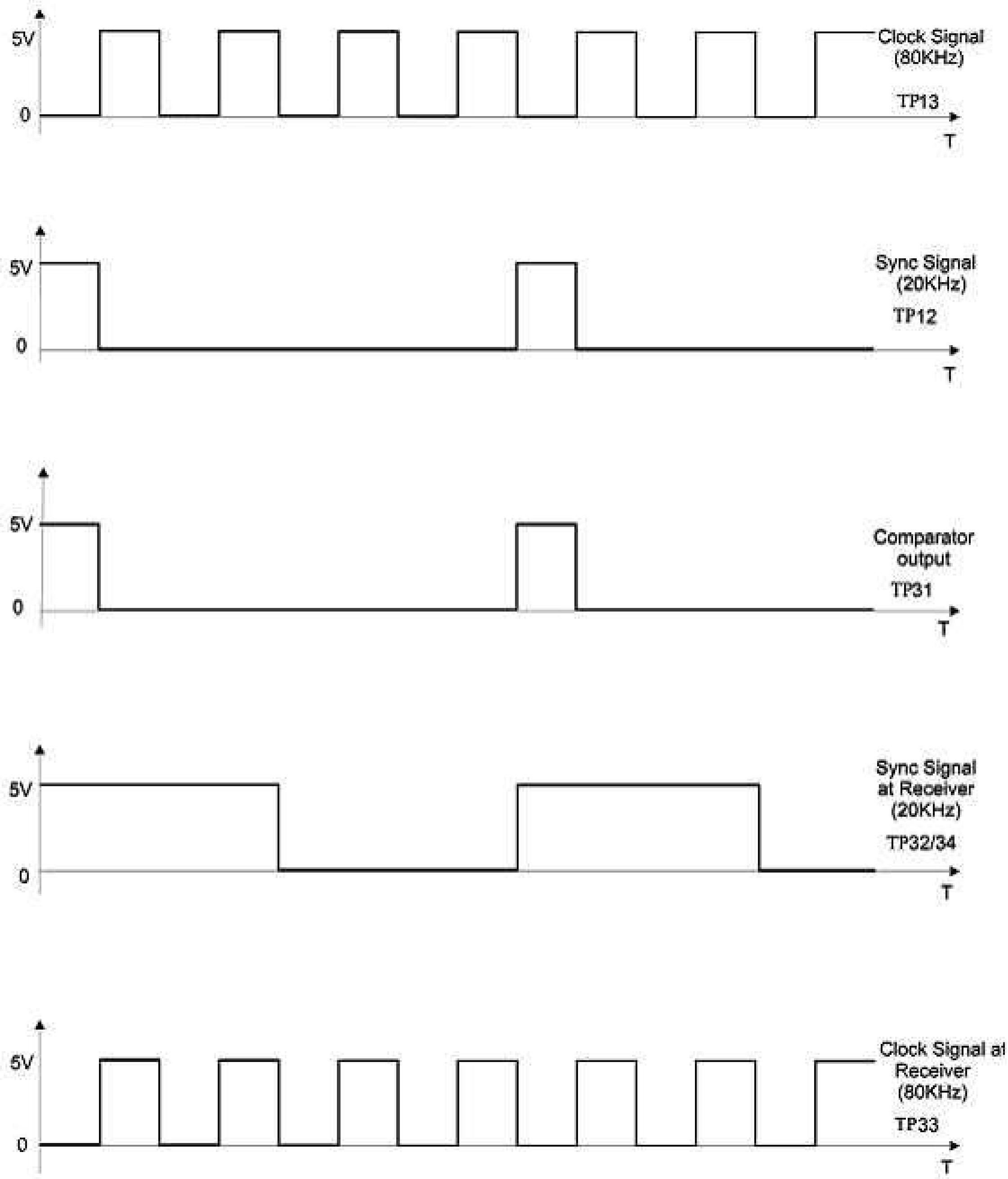
6. Ensure following peak voltage levels at the described test-points by varying thecorrespondingpotentiometers in the Signal Generator Unit.
    a. DC Level : 7 V (at TP18)
    b. ~500Hz : 4 V(at TP20)
    c. ~1 KHz : 3 V(at TP22)
    d. ~2 KHz : 2 V (at TP24)
7. Connect the Oscilloscope channel CH1 at TP27, draw the waveforms.
8. Connect the Oscilloscope channel CH1 at TP13 and other channel at TP12, TP31, TP32, TP33, measure amplitude, frequency and draw the waveforms.
9. Connect the Oscilloscope channel CH1 at TP43, TP45, TP47 and TP49 one by one and channel CH2 at TP44, TP46, TP48 and TP50 one by one respectively, and note amplitude, frequency and draw the waveforms.
10. Now vary the Duty Cycle Selector switch. Notice the effect of variation Duty Cycle Selector switch on Tx. O/P & amplitude of the Receiver's CH1 Output.
11. Set the 'Duty cycle Selector' switch again in '5' position.
12. Push the Sampling Frequency Selector button at Transmitter block and observe the wave shape change in output signal.

Observations:

1. Tx. Clock signal is used by the receiver to synchronize its activity &Tx. Sync signal is used by the receiver to know which sample belongs to which channel and these signal are generated at the receiver using Phase Lock Loop.
2. The waveform at TP27 is time division multiplexed samples of each input channel as shown in the figure 9.2.
3. The waveforms of transmitter clock TP13, PLL clock at TP33, transmitter Sync (receiver) TP12 and PLL (receiver) sync at TP34. Comparison among these signals is shown in the figure 6.12.

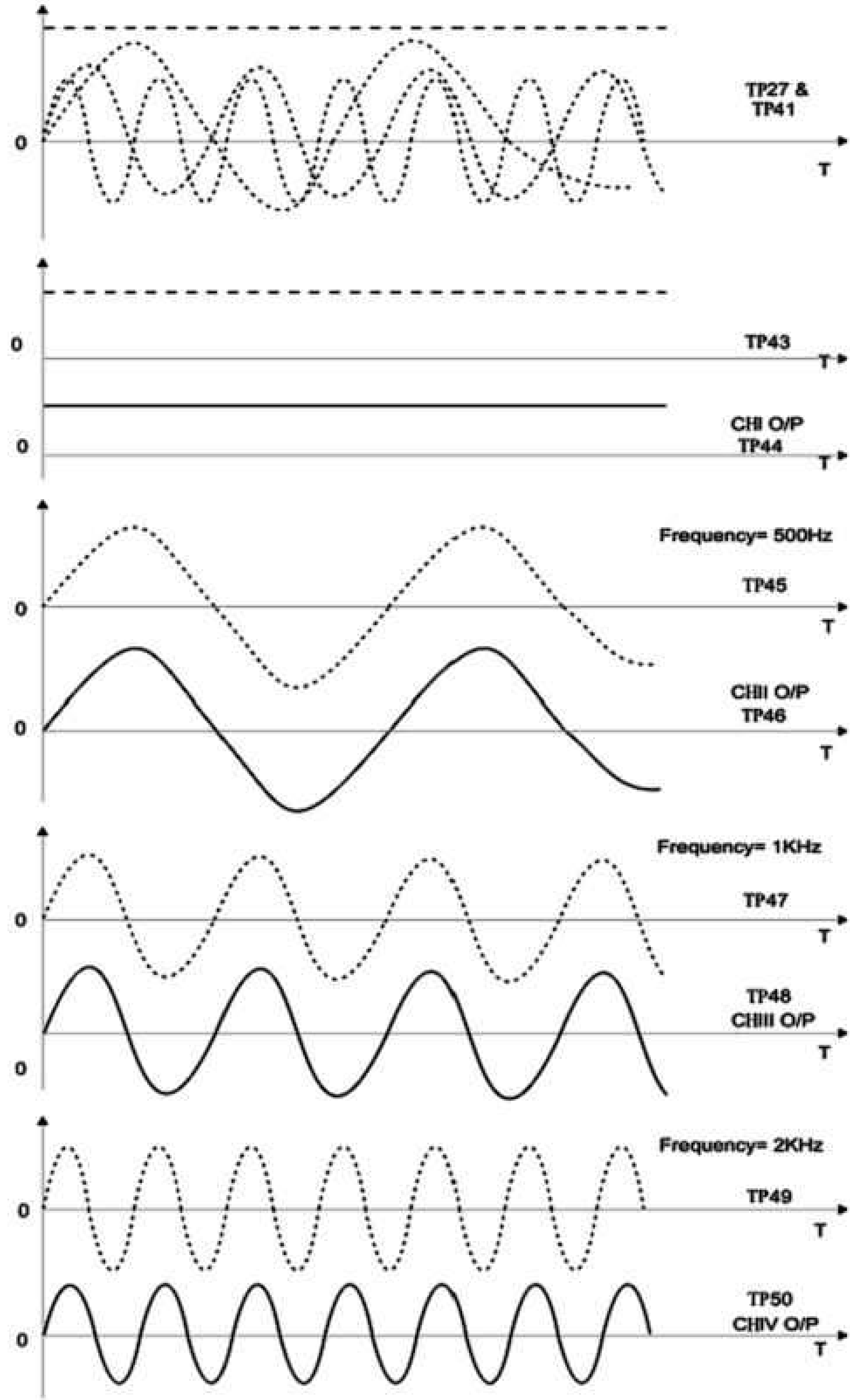
4. The output at TP43, TP45, TP47 and TP49 are Receiver's Low pass filter's inputs and TP44, TP46, TP48 and TP50 are CH1, CH2, CH3 and CH4 Outputs respectively as shown in the figure 6.13.

Observation diagram:



Control signal, Clock signal, Sync Signal of Transmitter and Receiver

Fig. 6.12



Sampled Input and Output of Various Transmission Channels

Fig. 6.13

Conclusion:

1. *In Mode-3 of communication there is only one link between transmitter and receiver and receiver extract Clock signal and Synchronizing signal from this time multiplexed pulse amplitude modulated signal.*
2. *At 50% duty cycle and highest sampling frequency, the output at Receivers channel is in same order as it is in Transmitter's input.*
3. *Output voltage at Receiver will be attenuated due to Low pass filters'*
4. *As we change the duty cycle the wave shape of The Receiver's output will be distorted or have very less amplitude due to change in 'On' and 'Off' time of its sampling Frequency.*
5. *The wave shape of output at Receiver's channel will be distorted due to fewer samples for better reconstruction of received signal.*

Experimental Data:

Signal	Amplitude(Peak to Peak)	Frequency
Tx Sync (TP12)		
Sync (TP13)		
Receiver Clock (TP36)		
Receiver Sync (TP35)		
Transmitter output(TP27)		
Receiver input(TP41)		
CH1 I/P (TP18)		
CH1 O/P (TP44)		
CH2 I/P (TP20)		
CH2 O/P (TP46)		
CH3 I/P (TP22)		
CH3 O/P (TP48)		
CH4 I/P (TP25)		
CH4 O/P (TP50		

Precautionary Measure to be taken:

1. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
2. For connecting any signal from one equipment to another equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.
3. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.
4. For unusual spark or burning smell, immediately switch off the main supply to the kit.

5. Ensure that, a signal is connected to an appropriate junction destined for it.
6. Always cover the equipment for dust protection after the experiment.

Some Sample questions:

1. What do you understand by PAM?
2. What do you understand by guard time?
3. Why Synchronization is required for TDM-PAM system?
4. Draw the block diagram of TDM-PAM system?
5. Why high frequency signals can not be transmitted through TDM system?

EXPERIMENT # 7

Title: Design and set up a PLL using VCO & to measure the lock frequency

Objective:

- Measurement of Capture Range
- Measurement of Lock Range

Theory:

ThePhase locked Loop is a feedback system consisting of the following units:

- Phase Comparator
- Low Pass Filter
- Error amplifier
- Voltage Controlled Oscillator(VCO)

These units are interconnected as shown in fig. 7.1 with no input signal applied, the VCO runs at a predetermined centre frequency called f_0. When an input signal of a particular frequency is applied to the phase comparator input, this compares the phase and hence the frequency of the input signal with that of the VCO output and generates an error voltage depending upon their phase difference. This error voltage is filtered by LPF(Low Pass Filter), amplified by error amplifier and feedback to the VCO. This control signal changes the VCO output frequency to that of the input signal frequency. Thus the feedback loop forces the VCO frequency to lock with the input signal frequency. Thus the feedback loop forces the VCO frequency to lock with the input signal frequency.

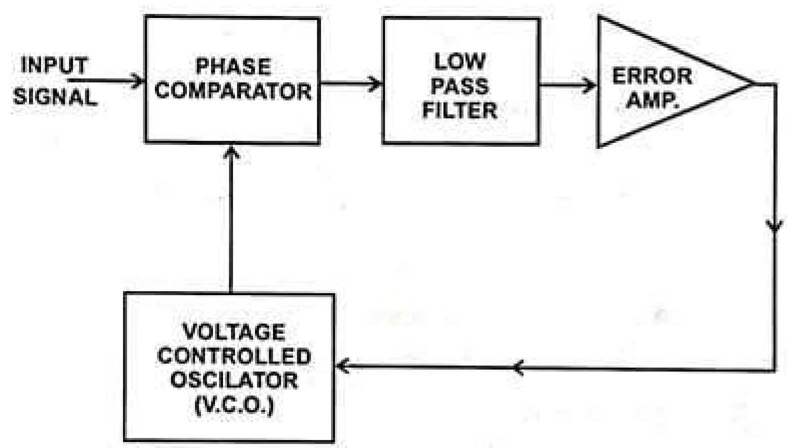


Fig:7.1 Block Diagram of PLL

CAPTURE RANGE

This is the range of frequencies about the centre frequency f_0 over which the PLL will acquire lock with an input signal initially starting from out of lock condition.

LOCK RANGE

Once PLL is locked to the input signal, the VCO output will follow the changes in input signal frequency over a range of frequencies about f_0. This range is called lock range. Lock range is always greater than the capture range.

Design Formula

The free running frequency of VCO

$f_0 = \frac{1.2}{4RC}$ in Hz…………………………………(1)

Where R and C are the external resistance & capacitor added to IC 565 ref. to Fig. 2

Lock range $f_L = \pm \frac{8fo}{Vcc}$ in Hz ………………………………………(2)

Take V_{cc-} as 12V (± 6V DC)

Capture range $f_0 = \pm \frac{1}{2\pi}\sqrt{\frac{2\pi f L}{T}}$

Where T = the time constant = $R_{inf}\, C_4$

Where R_{inf} = 3K6 (internally connected)

Thus T= $3.6 \times 10^3 \times C_4$

Where C_4 is connected as shown in fig 7.4

PANEL DESCRIPTION

The panel of this Electronic Training Board has been designed keeping in mind the case of use and clarity.

The circuits can be assembled using Patch-Cords and thus no soldering is required. The various inputs/outputs and controls have been described below. The IC NE-565 is placed almost in the middle of the panel. All the required pins have been brought out through sockets. The mains power switch and fuse is provided at the extreme left hand bottom part.

± 6V DC supply is provided in the middle top part of panel. On the extreme right hand top is a D.C. voltmeter switchable to 10V or 1V range with a switch provided on its left side. This meter is for voltage measurements. There is a linear potentiometer of 10K value on extreme left hand top side. Two sets of three sockets each are provided around the IC NE-565 for linking to other sockets when required. Along the bottom side of the panel there are selectable values of fixed value resistances and capacitors.

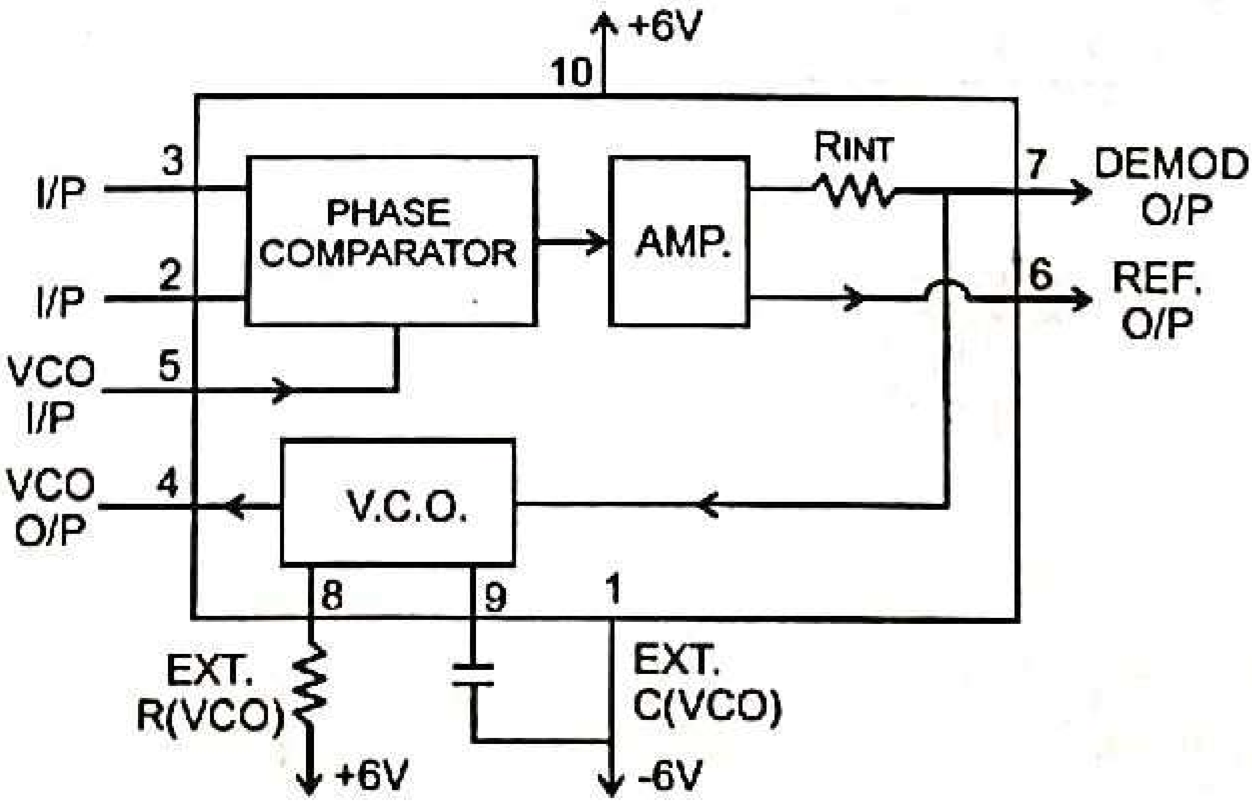


FIG – 7.2 BLOCK DIAGRAM OF IC NE-565

VCO CHARACTERISTICS:

Objective:

To study and measure the Free running frequency or centre frequency of VCO

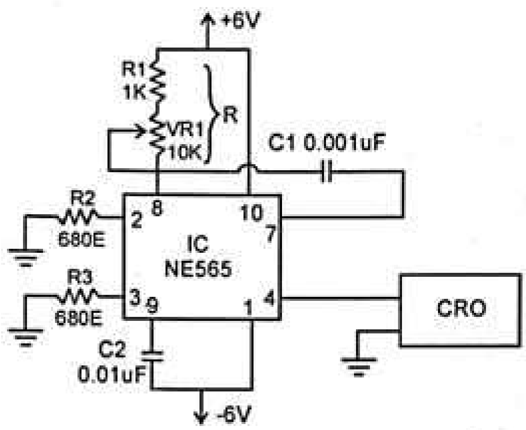


FIG – 7.3 MEASUREMENT OF CENTRE FREQUENCY f_0

Procedure:

1. Using Patch Cords assemble the circuit of fig. 7.3
2. Switch ON mains power
3. Observe the output at pin 4 on CRO. It will be square wave signal and is the free running frequency f_0.
4. Now rotate the knob of the potentiometer VR1 and observe that the frequency of the VCO signal varies. Thus the centre frequencies dependent on the value of resistance R (consisting of R1 + VR1) and capacitor C2. The frequency can be measured accurately by connecting a frequency counter in place of the CRO.

PLL Characteristics

To study and measure capture range and lock range

Block Diagram:

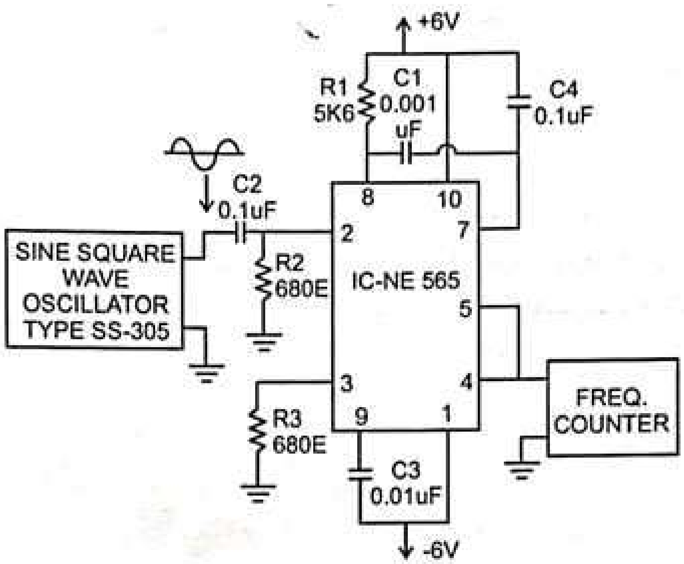


FIG:7.4 MEASUREMENT OF CAPTURE AND LOCK RANGE

Procedure:

1. Assemble the circuit of fig. 7.4.
2. Switch ON the mains power
3. Apply sine wave signal slowly increase the input signal frequency from 1 KHz note down the frequency at which the VCO output starts following the input signal frequency let it be f_1. Now go on increasing the input signal frequency further till the VCO output stops following the input signal frequency let it be f_2'.

   Now again start decreasing the input signal frequency and note down the frequency at which VCO output again starts following the input signal frequency let it be f_2. Go on decreasing the input signal frequency further till the VCO output stops following the input signal frequency, let it f_1'

   Lock range $=f_2' - f_1'$ and

   Capture range $= f_2 - f_1$

   Note that the capture range is similar than lock range.
4. Change the value of capacitor C4 to 0.01 µF and repeat the above steps. Note that the capture range increase.

Apparatus Required:

- ACL-NP Communication Trainer Kit
- Power Supply
- A CRO / DSO
- Amplitude Modulator Circuit

Other Accessories Required :

- Patch cords 4 mm length 50 cm Red…………08
- Patch cords 4 mm length 50 cm Black…………08

Discussion:

Write briefly your comments about the above experiment.

Precautionary Measure to be taken:

1. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
2. For connecting any signal from one equipment to equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.
3. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.
4. For unusual spark or burning smell, immediately switch off the main supply to the kit.
5. Ensure that, a signal is connected to an appropriate junction destined for it.
   Always cover the equipment for dust protection after the experiment
6. Power meter counts for first15seconds then shows stable output for 15seconds, this cycle repeats continuously .All measurements should be done when display isstable.

EXPERIMENT # 8

Title: MEASUREMENT OF SIGNAL TO NOISE RATIO OF A RF AMPLIFIER

Objective:

- Calculation of the Signal to Noise Ratio

Theory:

Signal- To- Noise Ratio

The calculation of the equivalent noise resistance of an amplifier, receiver or device may have one of two purposes or sometimes both. The first purpose is comparison of two kinds of equipment in evaluating their performance. The second is comparison of noise and signal at the same point. Therefore same point to ensure that the noise is not excessive. In the second instance, and also when equivalent noise resistance is difficult to obtain, the signal-to-noise ratio *S/N* is very often used. It is defined as the ratio of signal power to noise power. Signal to noise ratio is calculated using diff procedures for diff modulation techniques.

SNR = signal power / noise power

Theoretical Power Calculations

A. Sine Wave

Signal Power = $E^2/2$

$= (V_m \text{ Sin wt})^2/2$

$= V_m^2 \text{ Sin2wt12}$

$= V_m^2/2 \times (1\text{-Cos2wt})/2$

$= V_m^2/2$

$= 2^2/2$

= 2 Watt.

B. AM Signal

Let V_c (Carrier Amplitude) = 1.8Vpp

Let V_m (Modulating Signal Amplitude) = 0.5Vpp

Therefore P_c (Power of Carrier signal) = $V_c^2/2$ = 1.62

$m = V_m/ V_c$

= 0.5/1.8

= 0.28

Total Power (Pt) = $Pc\ (1 + m^2/2)$

$= 1.62 \times (1 + (0.27)^2/2)$

= 1.68W

Apparatus Required:

- ACL-NP Communication Trainer Kit
- Power Supply
- A CRO / DSO
- Amplitude Modulator Circuit

Block Diagram:

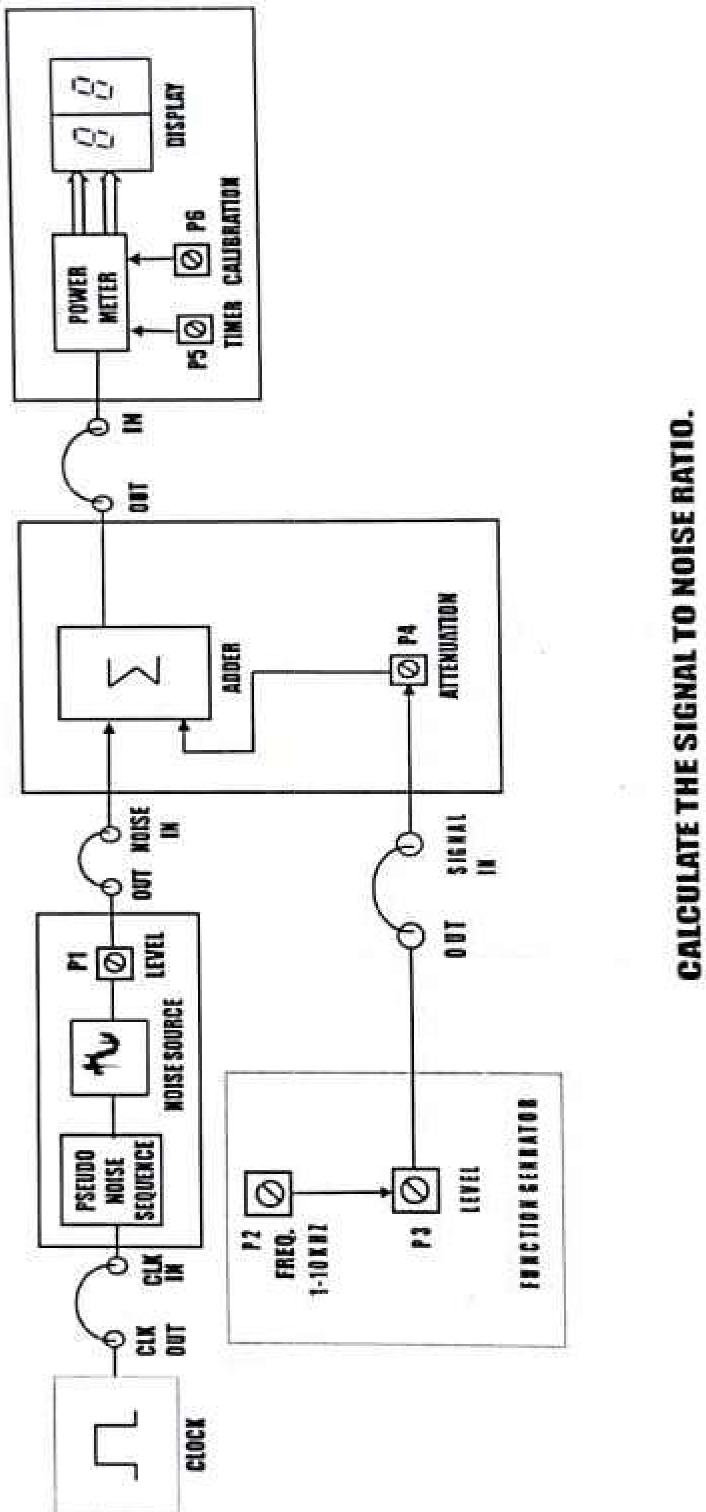


Fig:8.1

Procedure:

1. Connect the Power supply with proper polarity to the kit ACL-NP and Switch it **ON.**
2. Ref to the fig. 8.1 & carry out the following connections and settings.
3. Connect the **CLK OUT** post to **CLK IN** post of Noise generator.
4. Connect **OUT** post of noise generator to **NOISE IN** post os adder.
5. Keep the **LEVEL** pot P1 of noise generator to minimum position.
6. Connect **OUT** post of function generator to **SIGNAL IN** post of adder.
7. Keep freq. pot **P2** to minimum position i.e. at **1KHZ**.
8. Using pot **P3** keep signal amplitude at **2 Vpp.**
9. Keep Timer pot **P5** to maximum position.
10. Keep the signal attenuation at minimum using pot **P4**
11. Vary the calibration pot P6 in such a way that you should get power output approximates to the theoretical value of the power of the signal(i,e. Keep between 4 & 5).
12. Increase the noise level by using **LEVEL** pot **P1.**
13. Keep the attenuation pot **P4** to minimum position.
14. Measure noise power **Pn** on power meter.
15. Repeat the procedure by increasing the noise level in steps.

17. Calculate signal to noise ratio by using formula

    SNR = (Ps/Pn)

18. Similar observations and calculations can also be performed by using AM signal as input signal.
19. Vary the calibration **pot P6** in such a way that you should get power output, which approximates to the theoretical value of the power of the signal.(i.e. Keep between 7& 8)

Sr. No.	Psn	Pn	Ps = (Psn – Pn)	SNR
1.	2.1	0.1	2.0	20
2.	2.2	0.2	2.0	10
3.	2.4	0.3	2.1	7
4.	2.5	0.4	2.1	5.25
5.	2.6	0.5	2.1	4.2

(Table given for reference only actual values may differ from above)

Experimental Data:

Sr. No.	Psn	Pn	Ps = (Psn – Pn)	SNR
1.				
2.				
3.				
4.				
5.				

Data Analysis:

Calculate the ratio of Signal to Noise ratio.

Discussion:

Write briefly your comments about the above experiment.

Precautionary Measure to be taken:

1. Ensure that equipment or training kit switch is kept off. While connecting it to main power supply.
2. For connecting any signal from one equipment to equipment or from one section to other section of the same kit, ensure that signal is grounded properly and subsequently connect the signal-to-signal line.
3. Handle gently all the necessary button or knob in the equipment avoiding all other buttons or knobs which are not required to be adjusted for the equipment.
4. For unusual spark or burning smell, immediately switch off the main supply to the kit.
5. Ensure that, a signal is connected to an appropriate junction destined for it. Always cover the equipment for dust protection after the experiment
6. Power meter counts for first 15 seconds then shows stable output for 15 seconds, this cycle repeats continuously .All measurements should be done when display is stable.

Electronics and Communication Engineering Department

Continuous Lab assessment for 4ECE

Analog Communication Lab [EC 491]

EN: *Experiment No.*
FA: *File Assessment*
*PRFM: *Performance-*
E = Excellent (10), G = Good (8),
F = Fair (6), P = Poor (4),

SL.	Roll No.	Name	UNIV Roll No.	Week-			Week-			Week-			Week-			Week-		
				EN	FA	*PRFM	EN	FA	*PRFM	EN	FA	*PRFM	EN	FA	*PRFM	EN	FA	*PRFM
1	ECE/24/001	DURBADAL ROUTH	10700324010															
2	ECE/24/002	ANIKET PANDA	10700324009															
3	ECE/24/003	SHUBHANKAR BOSE	10700324008															
4	ECE/24/004	RATUL ROY	10700324007															
5	ECE/24/005	DEVDIPTA MONDAL	10700324006															
6	ECE/24/006	SRIJIT DAS	10700324005															
7	ECE/24/007	RIYA HEMBRAM	10700324004															
8	ECE/24/008	SUSMITA SENA	10700324003															
9	ECE/24/009	SOUMEN MANNA	10700324014															
10	ECE/24/010	SWARNAVA DAS	10700324001															
11	ECE/24/011	SAYAN SAMANTA	10700324002															
12	ECE/24/012	ANKAN MAITI	10700324013															
13	ECE/24/013	ARUP MAITY	10700324012															
14	ECE/24/015	NIBEDITA DAS	10700324011															
			Name of the Teacher															
			Signature with date															

Exp. No.	Experiment Details	Performed in

LABORATORYNAME: ………………………………………………………………………………..……..

NAME:…………………………………………COLLEGEROLLNO.……………………

IndexSheet

SL	TitleoftheExperiment	Exp.No.	Date ofExperiment	Date ofSubmission	Page No.	Grade	Teacher'sSignature
						☐E ☐G ☐F ☐P ☐NS	
						☐E ☐G ☐F ☐P ☐NS	
						☐E ☐G ☐F ☐P ☐NS	
						☐E ☐G ☐F ☐P ☐NS	
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						☐E ☐G ☐F ☐P ☐NS	
						☐E ☐G ☐F ☐P ☐NS	

** E: Excellent G: Good F: Fair P: Poor NS: Not Submitted