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DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGG.

CS-405 ANALOG & DIGITAL COMMUNICATION - LAB

LIST OF EXPERIMENT

1. STUDY OF SAMPLING PROCESS & SIGNAL RECONSTRUCTION & ANALIASING.

2. STUDY OF PAM, PPM, & PWM.

3. TO STUDY THE OPERATION OF A DSB AM MODULATOR & SINGLE SIDE BAND GENERATION.

4. STUDY OF CARRIER MODULATION TECHNIQUES BY ASK, PSK & FSK.

5. TO STUDY FREQUENCY MODULATION USING REACTANCE MODULATOR.

6. STUDY OF SENSITIVITY AND SELECTIVITY OF A RADIO RECEIVER.

7. TO PLOT THE CHARACTERISTICS OF THE PRE-EMPHASIS AND DE- EMPHASIS CIRCUIT.

8. STUDY OF AVC AND AFC.

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EXPERIMENT NO:-1

OBJECTIVE:

To study different types of signal samplings and its reconstruction.
1) Natural Sampling. 2) Sample and Hold. 3) Flat top sampling.

EQUIPMENTS:

- Connecting Chords.
- Power supply.

THEORY:

It basically consists of functional blocks, namely Function Generator, Sampling Control Logic, Clock section, Sampling Circuitry and Filter Section.

SAMPLING CONTROL LOGIC:

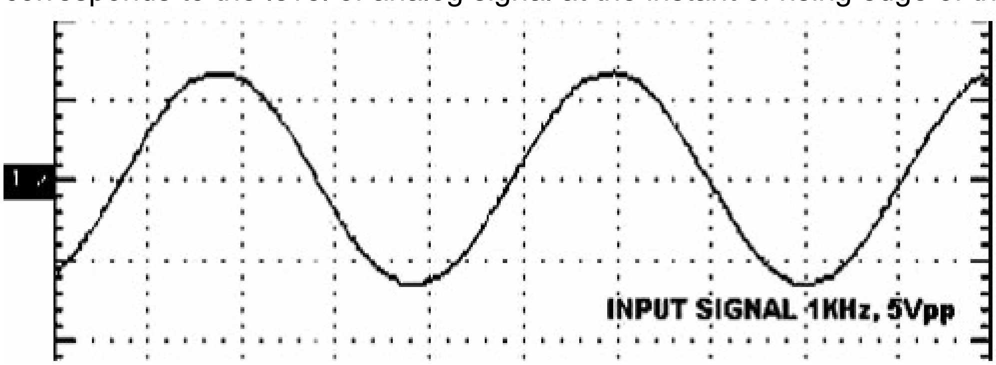
This unit generates two main signals used in the study of Sampling Theorem, namely the analog signals (5V pp, frequency 1 KHz and 2 KHz) and sampling signal of frequency 2 KHz, 4 KHz, 8 KHz, 16 KHz, 32 KHz, and 64 KHz.
The 6.4 MHz Crystal Oscillator generates the 6.4 MHz clock.

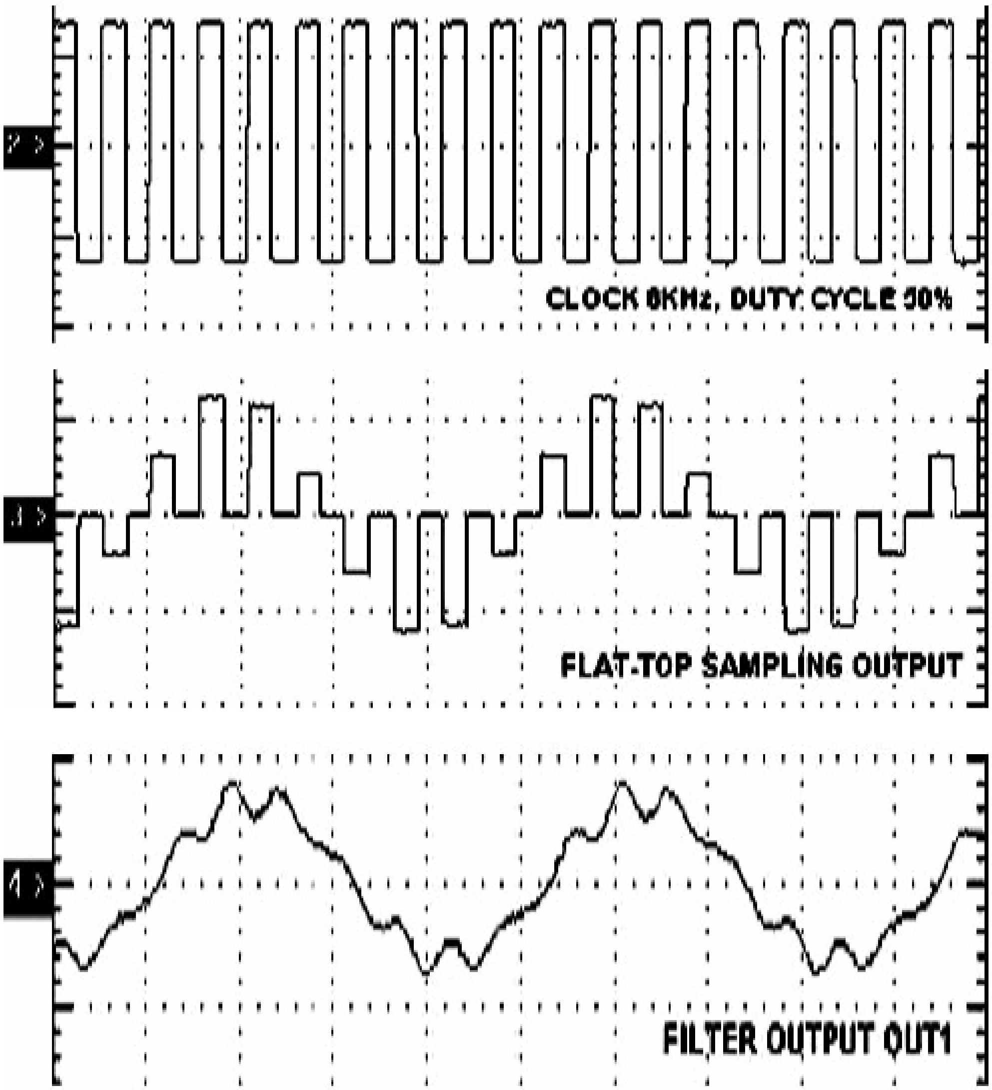
CLOCK SECTION:

This section facilitates the user to have his choice of external or internal clock feeding to the sampling section by using a switch.

SAMPLING CIRCUITRY:

The unit has three parts namely, Natural Sampling Circuit, Flat top Sampling Circuit, and Sample and Hold Circuit. The Natural sampling section takes sine wave as analog input and samples the analog input at the rate equal to the sampling signal.
For sample and hold circuit, the output is taken across a capacitor, which holds the level of the samples until the next sample arrives. For flat top sampling clock used is inverted to that of sample and hold circuit. Output of flat top sampling circuit is pulses with flat top and top corresponds to the level of analog signal at the instant of rising edge of the clock signal.





PROCEDURE:

- Connect power supply in proper polarity & switch it on.
- Connect the 1 KHz, 5Vpp Sine wave signal, generated onboard, to the BUF IN post of the BUFFER.
- Keep the sampling frequency clock in the internal mode INT CLK using switch (SW4).
- Using clock selector switch (S1) select 8 KHz sampling frequency.
- Using switch SW2 select 50% duty cycle.
- Connect BUF OUT post of the BUFFER to the IN post of the Natural Sampling block by means of the Connecting chords provided.

OBSERVATION:

- 1 KHz Analog Input waveform.
- Sampling frequency waveform.

RESULT:

VIVA QUESTIONS

1. What is the sampling theorem?
2. What is natural sampling?
3. Types of sampling?
4. What is flat-top sampling?
5. What is Nyqist theorem?

EXPERIMENT NO-2

OBJECTIVE:

Study of PAM/ PPM/ PWM

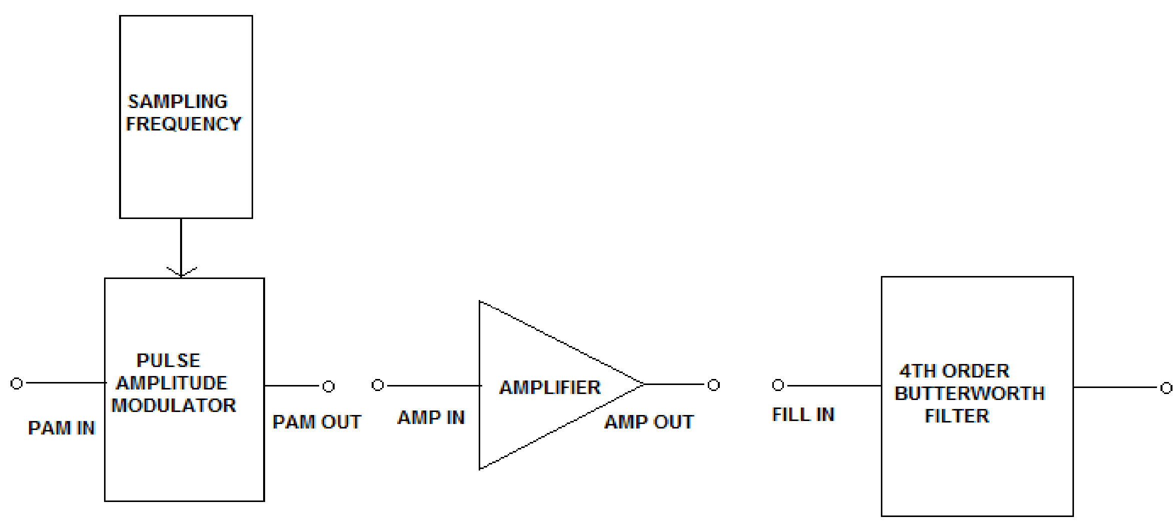
EQUIPMENTS:

- Connecting chords
- Power supply.
- 20 MHz Dual Trace Oscilloscope.

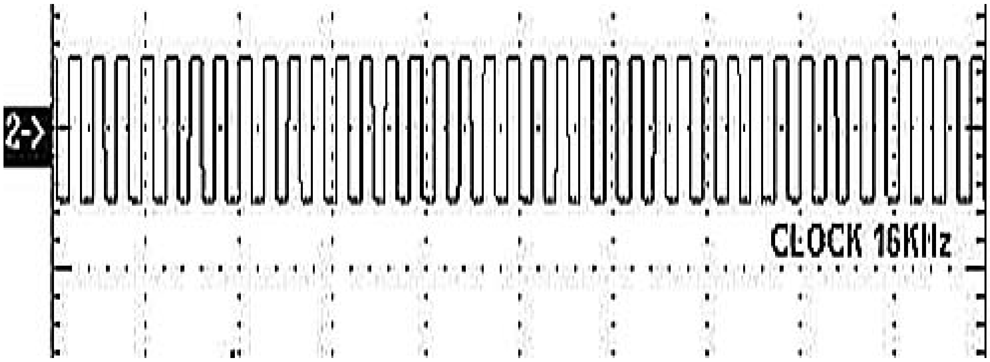
THEORY:

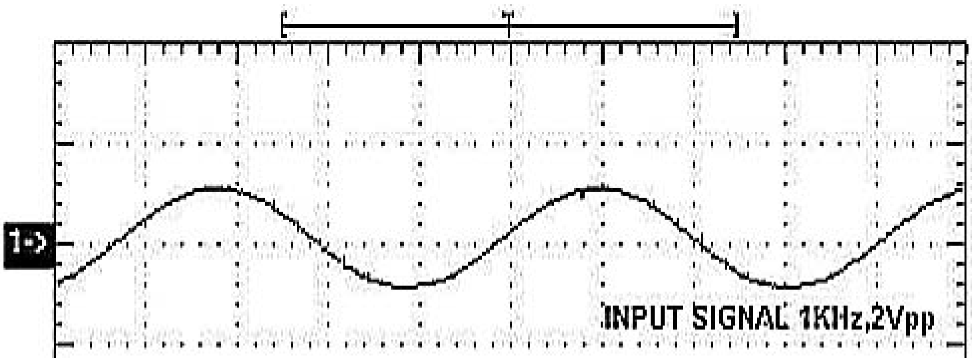
PAM:

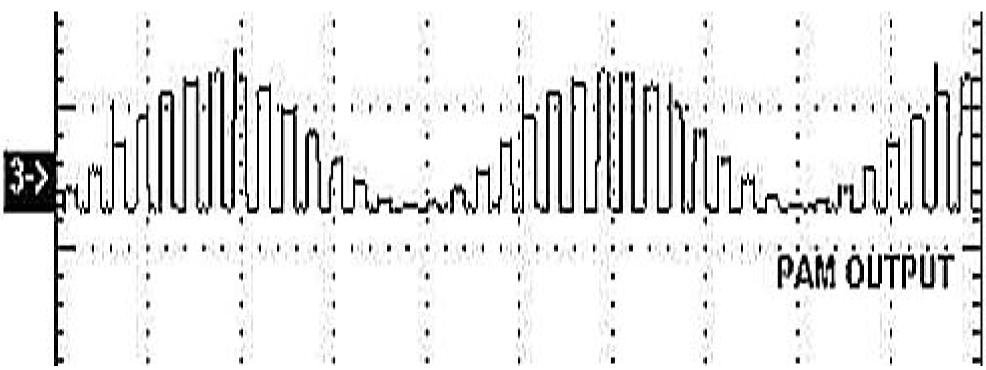
In Pulse Amplitude Modulation, the signal is sampled at regular intervals and the amplitude of each sample is made proportional to the amplitude of the signal at that instant of sampling. This amplitude of each sample is hold for the sample duration to make pulses flat top. It filters out the sampling frequency and their harmonics from the modulated signal and recovers the base band by integrated action.



BLOCK DIAGRAM FOR AMPLITUDE MODULATION AND DEMODULATION







PROCEDURE FOR PAM:

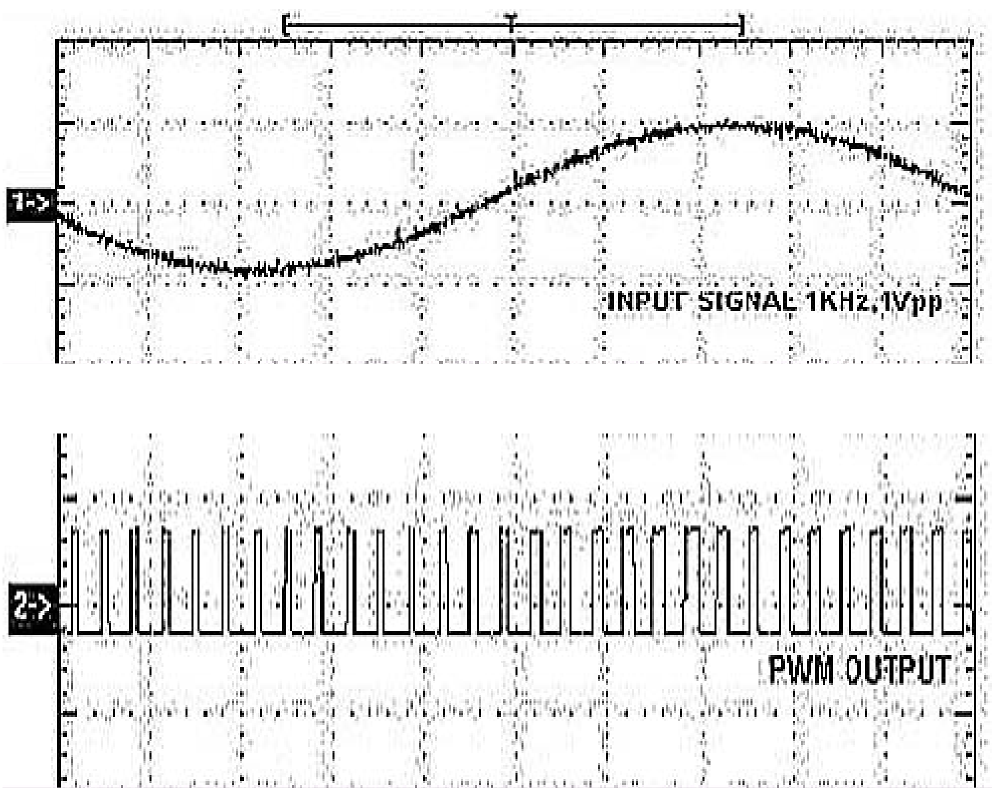
- Connect the power supply with the proper polarity and switch it ON.
- Select 16 KHz sampling frequency by jumper JP1.
- Connect the 1 KHz sine wave signal generated onboard 1 KHz post to PAM IN Post.
- Adjust the amplitude at 2Vp-p using pot P3.
- Observe the Pulse Amplitude Modulation output at PAM OUT Post.
- Connect PAM OUT post to AMP IN post.
- Connect AMP OUT post to FIL IN post.
- Keep the amplifier gain control potentiometer P5 to maximum completely clockwise.
- Observe the Amplified signal at AMP OUT Post.

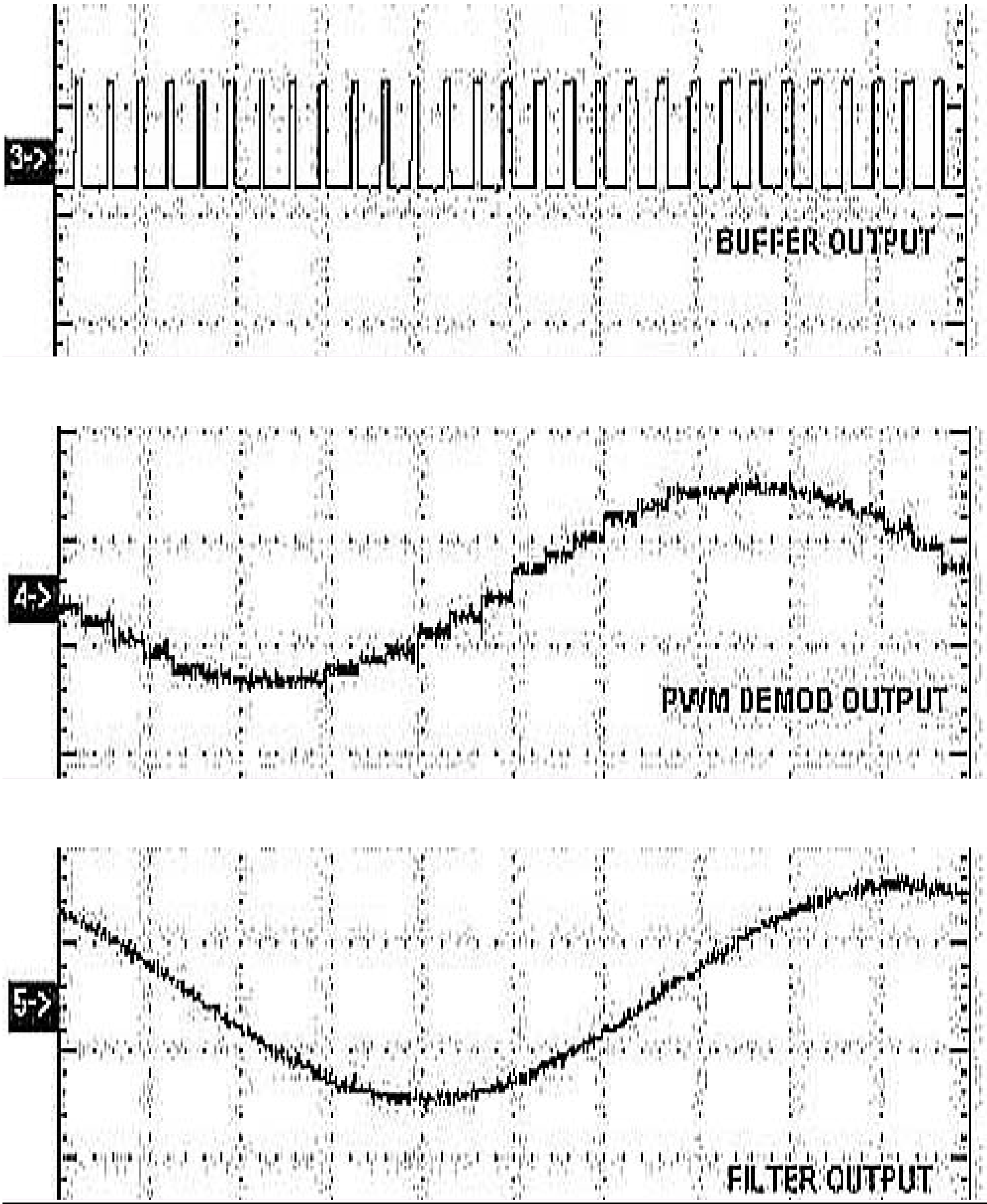
- Observe the Pulse Amplitude Demodulated signal at FIL OUT, which is same as the input signal

PWM:

This technique of modulation controls the variation of duty cycle of the square wave the amplitude variation of the modulation signal is reflected in the ON period variation of square wave. Hence, it is a technique of V to T conversion.

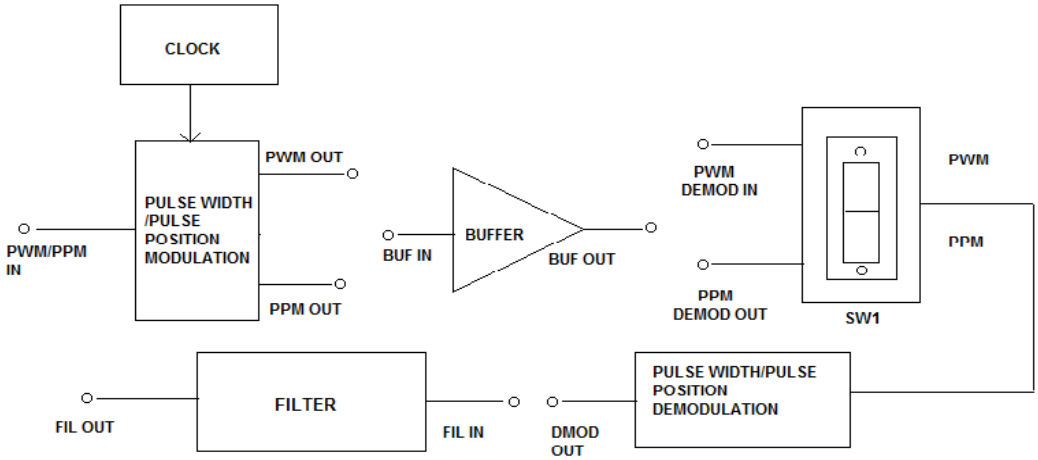
PULSE WIDTH DEMODULATION: The input signal is Pulse Width Modulated, so the ON time of the signal is changing according to the modulating signal. In this demodulation technique during the ON time of PWM signal one counter is enabled. At the end of ON time, counter gives a particular count, which directly corresponds to the amplitude of input signal. Then this count is fed to a DAC. The output of DAC corresponds to the amplitude of input signal. Thus train of varying pulse widths gives varying count values and accordingly DAC give outputs, which is directly proportional to amplitude of input signal. This is then filtered to get original signal. Thus at the output we get the original modulating signal extracted from PWM wave.



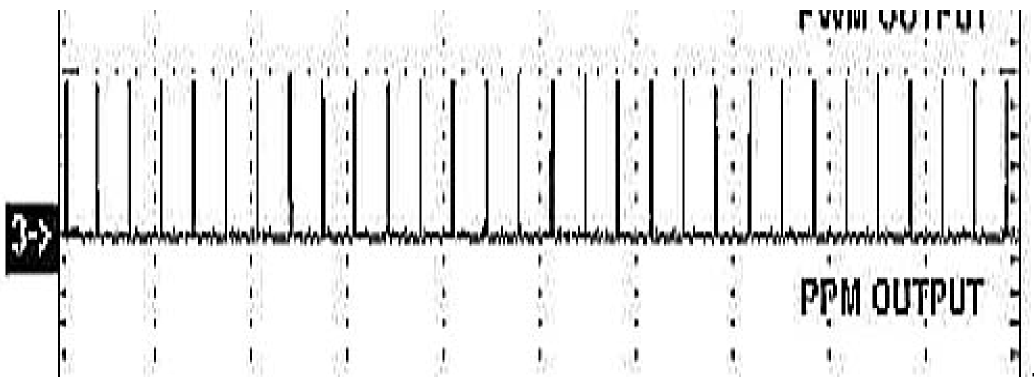


PPM:

This pulse position modulated signal is converted into PWM pulse form using Monostable multivibrator. This signal is then demodulated using the same technique of PWM demodulation. In this demodulation technique during the ON time of PWM signal one counter is enabled. At the end of ON time, counter gives a particular count, which directly corresponds to the amplitude of input signal. Then this count is fed to a DAC. The output of DAC corresponds to the amplitude of input signal. Thus train of varying pulse widths gives varying count values and accordingly DAC gives outputs, which is directly proportional to amplitude of input signal. This is then filtered to get original signal. Thus at the output we get the original modulating signal extracted from PWM wave.



BLOCK DIAGRAM FOR PPM/ PWM MODULATION AND DEMODULATION



PROCEDURE FOR PWM & PPM:

- Connect the Power Supply with proper polarity and switch it on.
- Put jumper JP3 to 2nd position.
- Connect the 1 KHz sine wave signal generated onboard 1 KHz post to PWM / PPM IN.
- Adjust the amplitude at 1Vp-p using pot P3.
- Connect PPM OUT post to BUF IN post.
- Connect BUF OUT post to PPM DMOD IN post.
- Keep Switch SW1 to PPM position.
- Connect DMOD OUT post to FIL IN post.
- Observe the Pulse Width Demodulated output at FIL OUT.

RESULT:

VIVA QUESTIONS

1. What do you mean by NYQUIST rate?
2. What is PPM?
3. Drawbacks of PAM signal?
4. Advantage and disadvantage of PWM?
5. What is the pulse amplitude modulation?

EXPERIMENT NO:

OBJECTIVE:-

To study the operation of a DSB AM Modulator & Single Side Band Generation.

EQUIPMENT:-

- Power supply.
- 20MHz Oscilloscope.
- Connective links
- Frequency Counter

THEORY:-

AMPLITUDE MODULATION

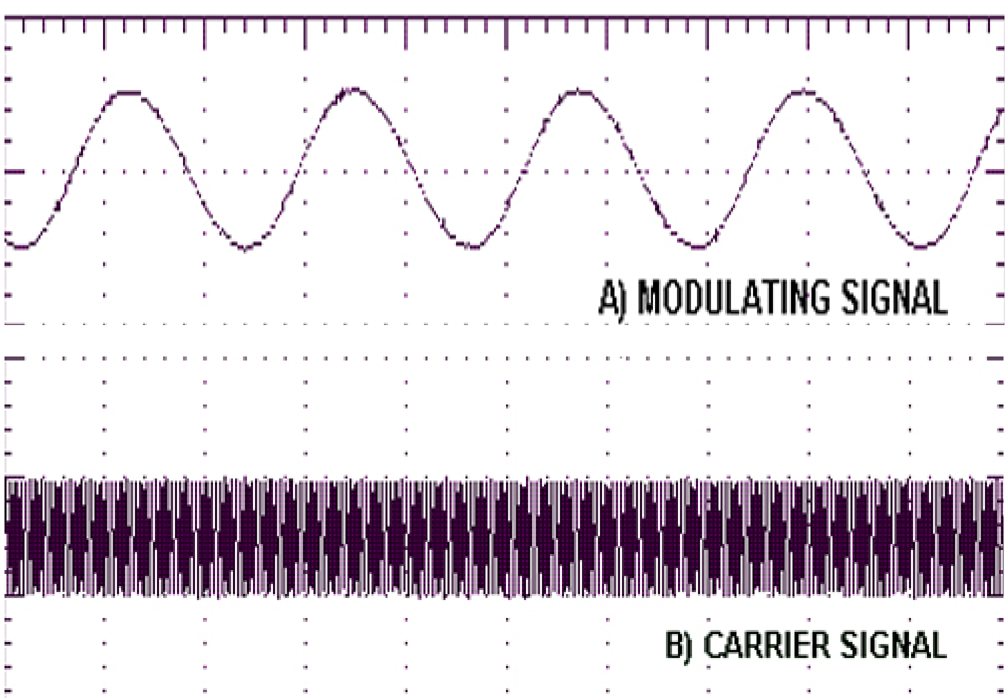
In Amplitude Modulation the amplitude of high frequency sine wave (carrier) is varied in accordance with the instantaneous value of the modulating signal. Refer FIG.6.
Consider a sine signal Vm(t) with frequency f(FIG.7).
$Vm(t)=B\bullet\sin(2\pi f\bullet t)$
And another sine signal vc(t) is called modulating signal, the signal Vc(t) is called carrier signal.
$Vc(t) = A\bullet Sin\ (2\pi F\bullet t)$
The signal Vm(t) is called modulating signal; the signal Vc(t) is called carrier signal.
Vary the amplitude of the carrier Vc(t) adding the modulating signal Vm(t) to A.
You obtain a signal vM(t) amplitude modulated, which can be expressed by:
$vM(t)=[A+k)\ \bullet B\bullet\sin(2\pi f\bullet t)]\ \bullet\sin(2\pi F\bullet t) = A\bullet\ [1+m)\ \bullet\sin(2\pi f\bullet t)]\ \bullet\sin(2\pi F\bullet t)$
With k=constant of proportionality.
Percentage modulation signal is defined as
$m=((K\cdot B)/A)\cdot 100$

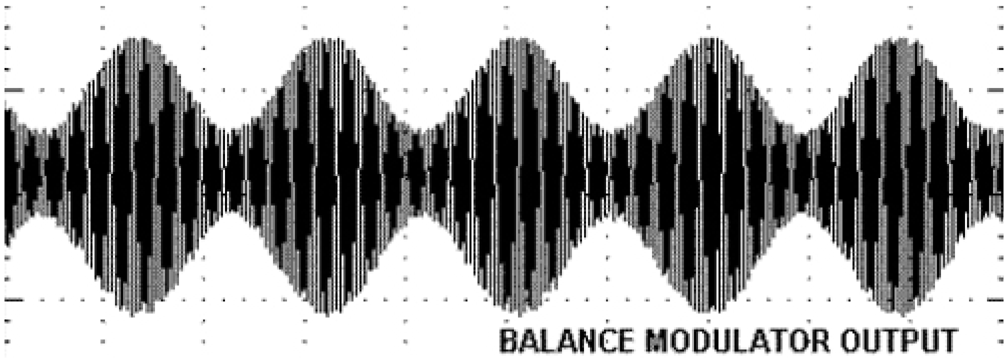
SPECTRUM OF THE MODULATED SIGNAL

With simple trigonometric passages, the relation expressing the modulated signal vM becomes:
$vM(t)=A\bullet\sin(2\pi F\bullet t)+m\bullet A/2\bullet\cos\ [(2\pi\ (F-f)\ \bullet t]-m\bullet A/2\bullet\cos[(2\pi\ (F+f)\ \bullet t)]$
From which we can deduce that the signal modulated in amplitude by a sine modulator consists of three sine components:
$A\cdot\sin(2pF\cdot t)$ Carrier
$m\bullet A/2\bullet\cos[(2\pi\ (F-f\bullet\cdot t)]$
Lower side band
$m\bullet A/2\bullet\cos[(2\pi\ (F+f)\ \bullet t)]$

PROCEDURE FOR DSB :-

- Connect SINE OUT post of FUNCTION GENERATOR SECTION to the i/p of Balance Modulator1 SIGNAL IN Post.
- To connect o/p of VCO RF OUT post to the input of Balance modulator 1 CARRIER IN post
- Connect the power supply with proper polarity while connecting this; ensure that the . power supply is OFF.
- Keep switch SW1 towards 1-10 KHz position.
- Keep Out post LEVEL about 0.5Vpp; FREQ. About 1 KHz.
- Keep switch SW2 towards 500 KHz position.
- Keep RF out LEVEL about 1 Vpp; FREQ. about 450 KHz, Switch on 500 KHz.
- BALANCED MODULATOR1: CARRIER NULL completely rotated clockwise or counter clockwise, so as “unbalance” the modulator and to obtain an AM signal with not suppressed carrier across the output; OUT LEVEL in fully clockwise.
- Observe the AM Modulator wave.
- Move the probe from post SIG to post OUT (output of the modulator), where signal modulated in amplitude is detected. Note that the modulated signal envelope corresponds to the wave form of the DSB AM modulating signal.
- Vary the amplitude of the modulating signal and check the 3 following conditions: Modulation percentage lower than the 100%, equal to the 100% , superior to 100% (over modulation).
- Vary the frequency and amplitude of the modulating signal, and check the corresponding variations of the modulated signal.
- Vary the amplitude of the modulating signal and note that the modulated signal can result saturation or over modulation.





PROCEDURE FOR SSB :-

- Refer to the FIG and Carry out the following connections.
- Keep all the switch faults in OFF position.
- Connect o/p of SINEWAVE section (ACL-01) OUT post to the i/p of Balance Modulator (ACL-01) SIGNAL post.
- Connect o/p of VCO (ACL-01) OUT post to the input of Balance modulator CAR. (ACL-01) post.
- Connect power supply with proper polarity to the kit ACL-01, while connecting this, and ensure that the power supply is OFF.
- Switch on the power supply.
- Keep switch SW1 towards 1-10 KHz position.
- Keep sine level about 1 Vpp, Freq. about 1 KHz.
- Keep switch SW2 towards 500 KHz position.
- Keep VCO level about 1 Vpp , freq. about 450 KHz.
- Keep Balanced Modulator Carrier null in central position, so that the modulator is" balanced" and obtain an AM signal across the output with suppressed carrier. OUT LEVEL in fully clockwise position.
- Connect the oscilloscope to the inputs of the modulator (posts SIG and CAR.) and detect the modulating signal and the carrier signal
- Move the probe from post SIG. to post OUT. where the modulated signal is detected
- Vary the amplitude of the modulating signal and check the corresponding variation of the modulated signal amplitude. Note that, differently from the AM modulation where the modulated signal is never null, the modulated signal annuls when the modulating signal is null.
- Vary frequency and waveform of the modulating signal, and check the corresponding variations of the modulated signal.

RESULT:

VIVA QUESTIONS

Q1. What is the BW for AM wave?
Q2. Define Modulation index for AM wave in AM system?
Q3. Define transmission efficiency in AM wave?
Q4. How can you obtain a DSB-SC signal?
Q5. What are the demodulation method for DSB- SC signal?
Q6. What are the generating method for SSB-SC signal?

EXPERIMENT NO:

OBJECTIVE:

Study of Carrier Modulation Techniques by ASK/FSK/PSK.

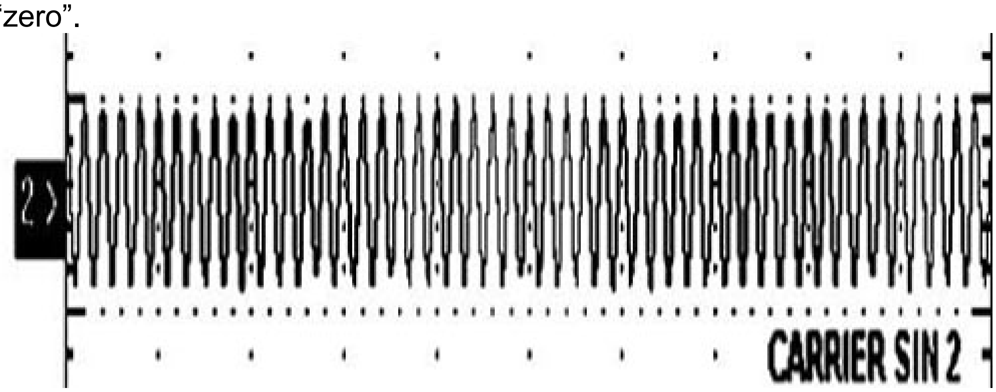
EQUIPMENTS:

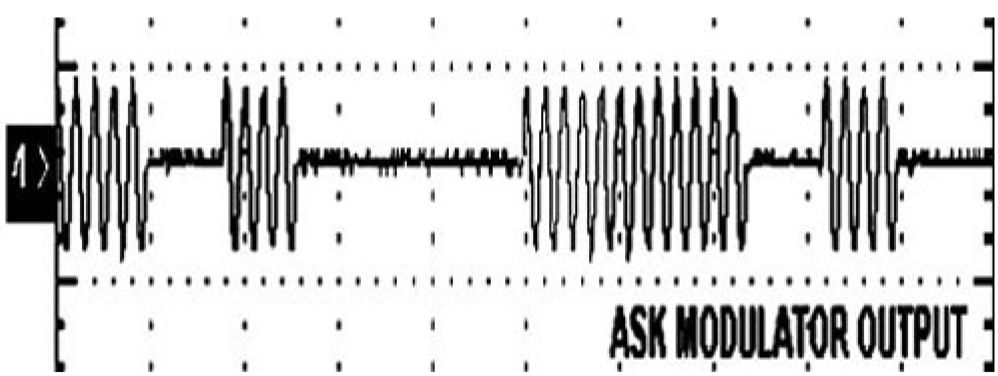
- Connecting chords
- Power supply.

THEORY:

ASK:

Carrier modulation is a technique by which digital data is made to modulate a continuous wave (sine wave) carrier. In Amplitude shift keying, the carrier is transmitted when the modulating data is `one' and the carrier is rejected from transmission when the data is “zero “The carrier frequency chosen for ASK modulation is 1 MHz ASK DEMODULATOR block on DCL-06 employs an envelope detector to recover the data from the modulated carrier. The ASK modulated input is fed to the half wave rectifier. The rectified input is fed to the filter, where the original data is recovered. The threshold detector is used to recover the original amplitude levels of the data. So whenever the sine wave is transmitted, the detector identifies it as a `one' and whenever the carrier is absent, the detector identifies it as a “zero”.





PROCEDURE:

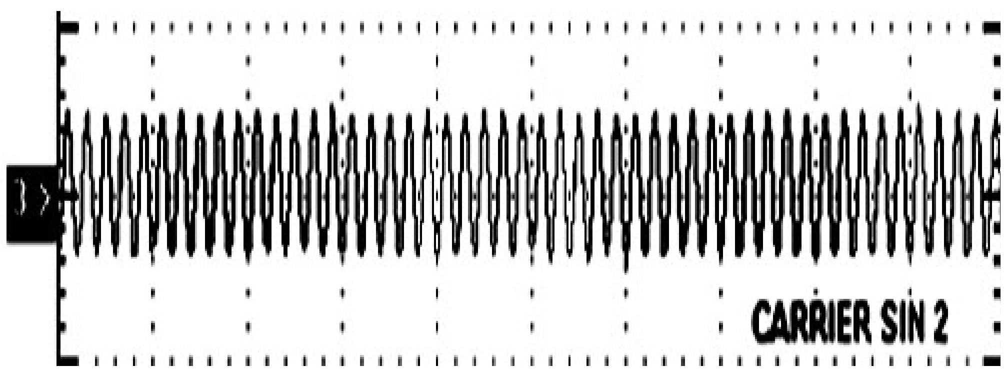
- Connect carrier component SIN2 to INPUT1
- Connect GROUND to INPUT2 of the Carrier Modulator Logic.
- Observe the modulated output at modulator

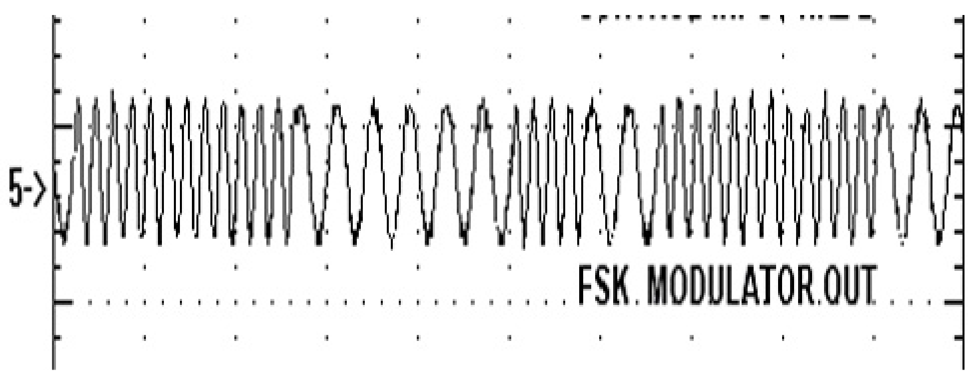
OBSERVATION:

- Input NRZ-L Data at CONTROL INPUT.
- Carrier frequency SIN 2.
- ASK modulated signal at MODULATOR OUTPUT.
- ASK demodulated signal at ASK OUT

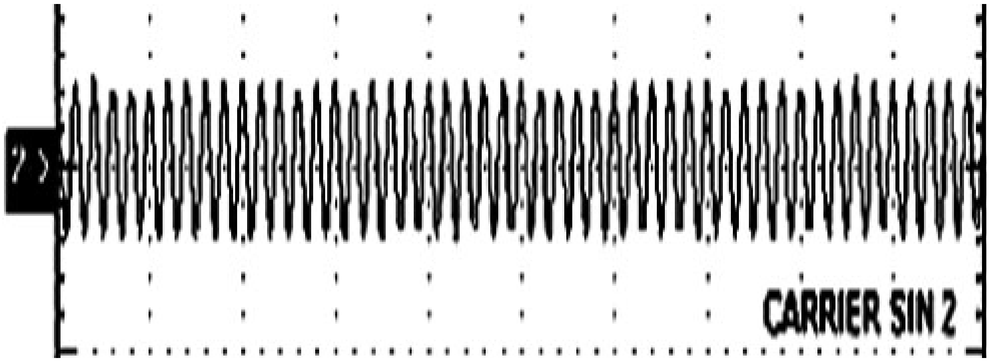
FSK AND PSK:

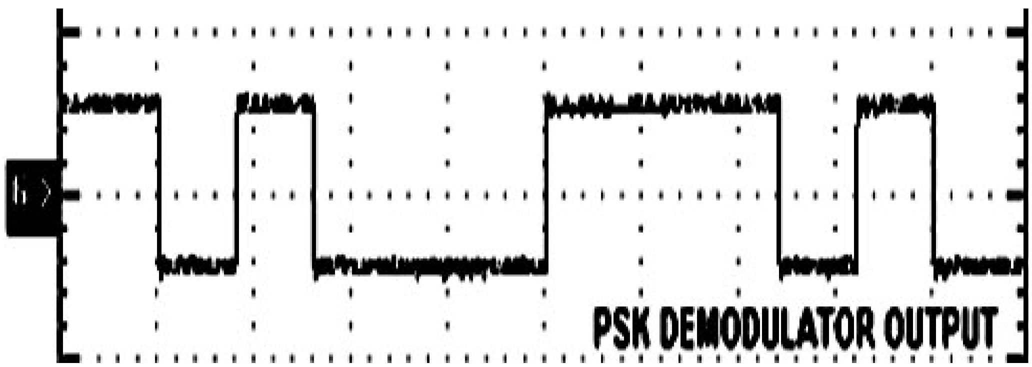
In Frequency Shift keying modulation techniques, the modulated output shifts between two frequencies for all `one' (mark) to `zero' (space) transitions. The carrier frequency chosen for FSK modulation are 500 KHz and 1MHz. Note that the above frequencies are greater than twice the modulating frequency. Note that the FSK may be thought of as an FM system in which the carrier frequency is midway between the mark and space frequencies, and modulation is by a square wave





psk





PROCEDURE:

- Connect power supply in proper polarity to the kits DCL-05 and DCL-06 and switch it on.
- Connect CLOCK to CODING CLOCK IN by means of the patch-chords provided.
- Connect DATA generated on DCL-05 to DATA INPUT by means of the patch-chords provided.
- Connect the NRZ-L data input to the CONTROL INPUT of the Carrier Modulator logic.
- Connect carrier component SIN1 to INPUT 2
- Connect SIN 2 to INPUT 1 of the Carrier Modulator Logic.
- Observe the modulated output at modulator
- Connect FSK modulated signal MODULATOR OUTPUT on DCL-05 to the FSK IN of the
- FSK DEMODULATOR on DCL-06.

OBSERVATION:

- Input NRZ-L Data at CONTROL INPUT.
- Carrier frequency SIN 1
- Carrier frequency SIN2
- FSK modulated signal at MODULATOR OUTPUT.
- FSK Demodulated signal at FSK OUT.

- Carrier frequency SIN3
- PSK modulated signal at MODULATOR OUTPUT.
- PSK Demodulated signal at PSK OUT

RESULT:

CONCLUSION:

A small phase lag exists between the modulating data and the recovered data because of the limitation of tracking ability and the time response of PLL

VIVA QUESTIONS

1. What do you mean by ASK?
2. What do you mean by PSK?
3. What do you mean by FSK?

EXPERIMENTS NO:

OBJECTIVE:-

To study frequency modulation using reactance modulator

EQUIPMENT:-

- ACL-03 Kit.
- Power supply.
- Connective links.
- Frequency meter.

THEORY:-

FREQUENCY MODULATION GENERATION:-

The circuits used to generate a frequency modulation must vary the frequency of a high frequency signal (carrier) as function of the amplitude of a low frequency Signal (modulating signal). In practice, there are two main methods used to Generate the FM:

DIRECT METHOD:-

An oscilloscope is used in which the reactance of one of the elements of the resonant circuit depends on the modulating voltage. The most common device with variable reactance is the Varactor or Varicap, which is a particular diode whose capacity varies as a function of the reverse bias voltage.

INDIRECT METHOD:-

In this case, FM is done by Phase Modulation, after the modulating signal has been integrated. In the phase modulator the carrier can be generated by quartz Oscillator and so its frequency stabilization is easier.
In the circuit used for the exercise, the frequency modulation is generated by a Hartley oscillator, whose frequency is determined by a fixed inductance and by a Capacitance (variable) supplied by Varicap diodes.

AN FM TRANSMITTER:-

The audio oscillator supplies the information signal and could, if we wish, be Replaced by a microphone and AF amplifier to provide speech and music instead of the sine wave signals that we are using with ACL-03.
The FM modulator is used to combine the carrier wave and the information Signal in the same way as in the AM transmitter. The only difference in this case is that the generation of the carrier wave and the modulation process is carried out in the same block. It doesn’t have to be, but in our case, it is.

The only real difference between the AM and FM transmitters are the Modulators, so we are only going to consider this part of the transmitter. We are going to investigate three types of modulator, they are called the **VARACTOR MODULATOR, REACTANCE MODULATOR** and the FM is obtained in this case by a Phase Modulation.

<u>THE VARICAP DIODE:-</u>

The Varicap (or Varactor) is a diode whose terminals are supplied with a capacity depending on the applied reverse voltage. The symbol and the equivalent circuit of the varicap diode are shown in FIG.4, where:

· Cj = junction capacity

· Rs= series resistance (it drops as the applied reverse voltage increases)

The junction capacity Cj depends on the reverse voltage VR applied to the diode, according to the relation:

Cj = C0 (1 + VR)/VD X h

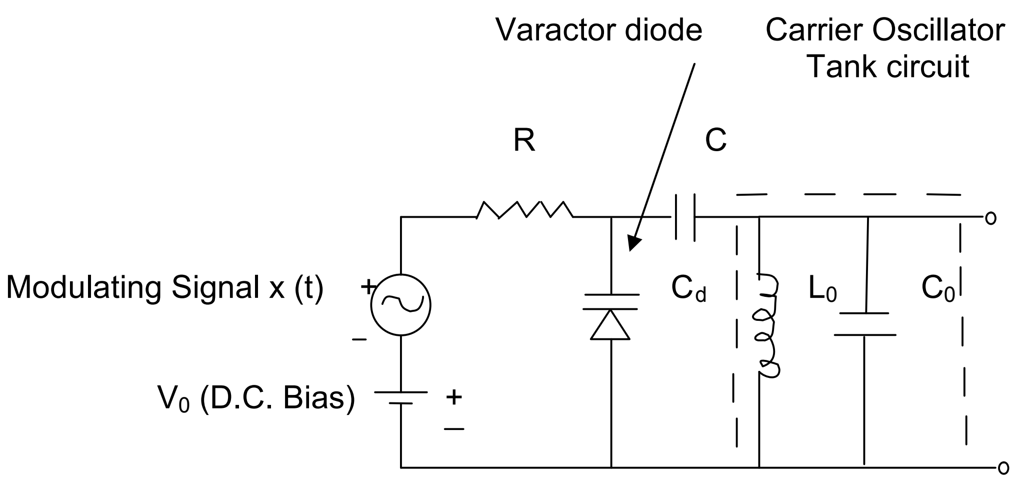
where:

VR = reverse voltage applied to Varicap

C0 = junction capacity for VR=0

VD = junction potential (0.6 V in the silicon diodes)

h = it depends on the manufacturing process; it ranges between 0.3 and 0.6approx.



<u>Varactor diode method of FM generation</u>

PROCEDURE:-

- Connect the power supply with proper polarity and Switch it on.
- Keep all Switch Faults in OFF position.
- Keep switch at 1500 KHz position.
- Using pot P5 keep frequency at minimum and using pot P6 keep Amplitude at 2Vpp.
- Connect the oscilloscope and frequency meter to the output of the Modulator FM/RF OUT.
- Connect the voltmeter to the cursor of the frequency regulation Potentiometer post V_f below SWITCH
- Vary the voltage in steps of 0.5 Volt and fill a table with the voltage values And the corresponding frequencies.
- Plot a graph with the measured voltage and frequency values.

RESULT:

VIVA QUESTIONS

1. What do you mean by angle modulation?

2. Define phase modulation

3. Define frequency modulation?

4. What are the disadvantages of FM modulation?

5. What is Carson rule?

EXPERIMENTS NO:

OBJECTIVE:-

Study of Sensitivity and Selectivity of a Radio receiver.

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EQUIPMENT:-

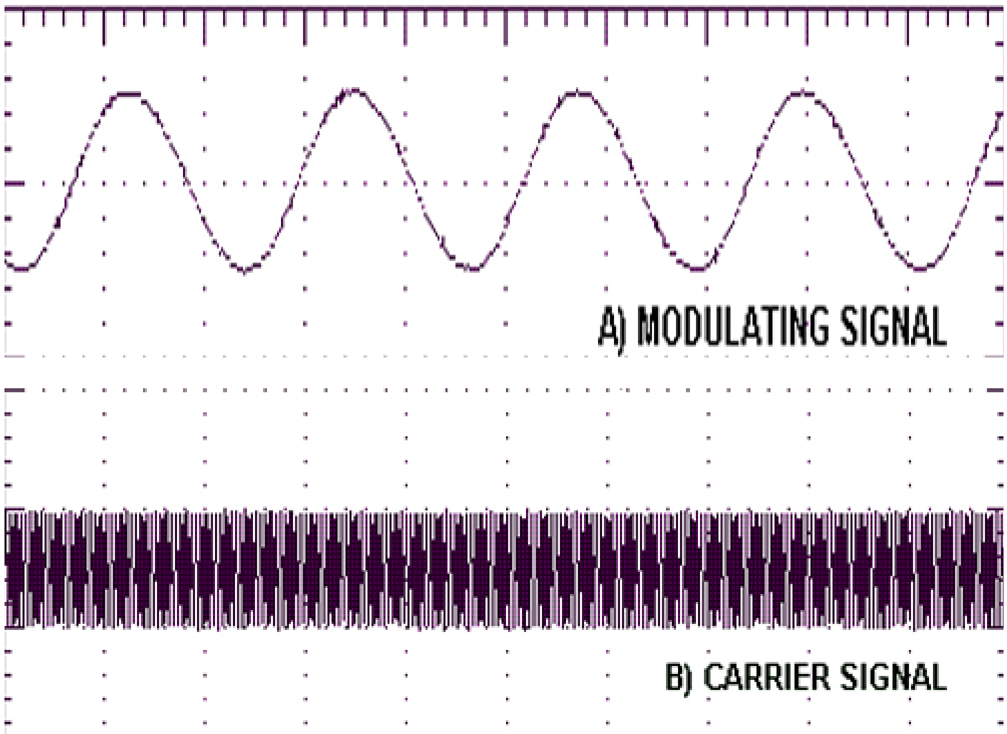
- Power supply GND, +5, +/-12V
- Connecting Links.
- Frequency Counter

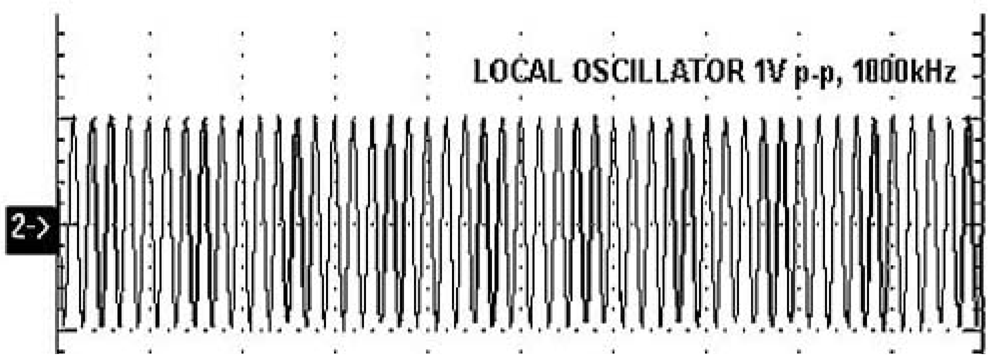
PROCEDURE:-

- Connect o/p of FUNCTION GENERATOR section (ACL-01) OUT post to the i/p of Balance Modulator1 (ACL-01) SIGNAL IN post.
- Connect o/p of FUNCTION GENERATOR section (ACL-01) OUT post to the i/p of Balance Modulator1 (ACL-01) SIGNAL IN post.
- Connect power supply with proper polarity.
  While connecting this, ensure that the power supply is OFF.
- Switch on the power supply.
- Keep switch SW1 towards 1-10KHZ position.
- Keep Sine out LEVEL about 0.5 Vpp; FREQ. About 1 KHZ.
- Keep switch SW2 towards 1500KHz position
- Keep LEVEL about 1.5Vpp; FREQ. About 600 KHz.
- BALANCED MODULATOR 1: 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 whose amplitude is about 200mVpp.
- OUTPUT AMPLIFIER (ACL-01): LEVEL fully clock wise.
- Keep LOCAL OSCILLATOR (ACL-02)signal at 1050KHz,1 v
- Connect local oscillator OUT post to LO IN of the mixer section.
- Connect balance modulator1 out to RF IN of mixer section in ACL-02.
- Connect mixer OUT to IF IN of 1st IF AMPLIFIER in ACL-02.
- Connect IF OUT1 of 1st IF to IF IN 1 and IF OUT2 of 1st IF to IF IN 2 of 2ND IF AMPLIFIER.
- Connect OUT post of 2nd IF amplifier to IN post of envelope detector.
- Connect post AGC1 to post AGC 2 and jumper position as per diagram.
- Observe the modulated signal envelope, which corresponds to the wave form of the modulating signal at OUT post of the balanced modulator1 of ACL-01. Connect the oscilloscope to the IN and OUT post of ENVELOPE DETECTOR and detect the AM signal.
- 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.
- Adjust the input to RF IN post by vaying the output of BM1 in such a way

that you should get minimum detected output of about 0.3Volt at the output of Envelope detector.

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





RESULT:

VIVA QUESTIONS

1. Write the drawback of tuned radio frequency (TRF) receiver?
2. What do you mean by sensitivity?
3. Define selectivity for a receiver?
4. Write the advantages of a RF amplifier?

EXPERIMENT NO:

OBJECTIVE:-

To plot the characteristics of the pre-emphasis and de-emphasis circuit.

EQUIPMENT:-

- ACL-03 Kit.
- Power supply.

THEORY:-

PRE-EMPHASIS CIRCUIT:-

The problem in FM broadcasting is that noise and hiss tend to be more Noticeable, especially when receiving the weaker stations. To reduce this effect, The treble response of the audio signal is artificially boosted prior to transmission. This is known as pre-emphasis
The pre-emphasis is obtained by using the simple audio filter, even simple RC filter will do the job. The pre-emphasis circuit produces higher output at higher frequencies Because the capacitive reactance is decreased as the frequency increases.

The response of the pre-emphasis circuit will be as follows:
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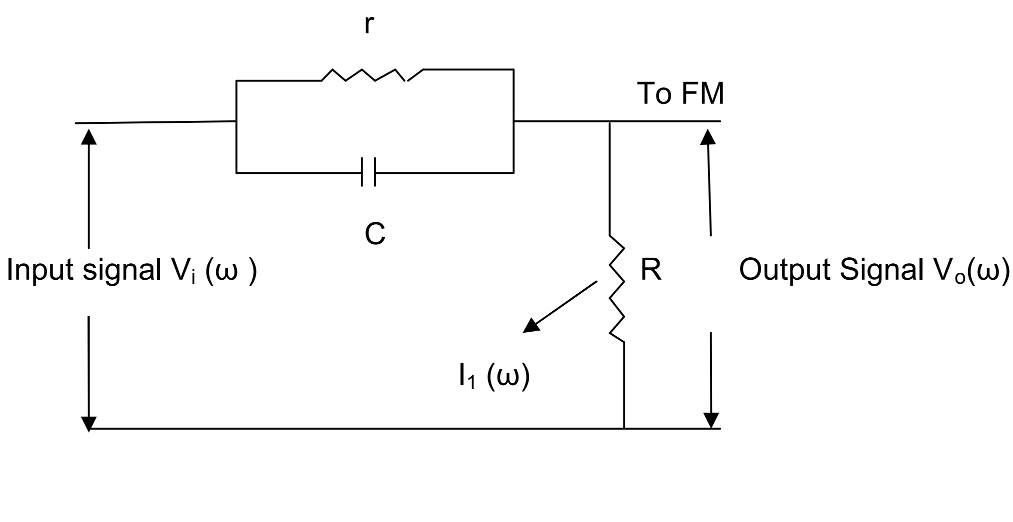
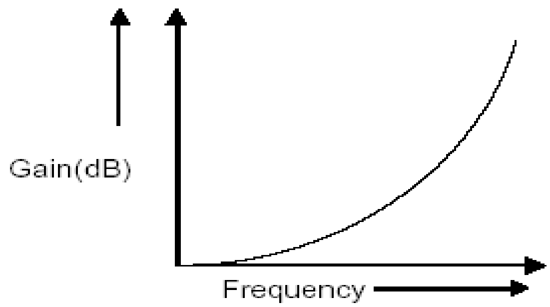


Pre-Emphasis Circuit

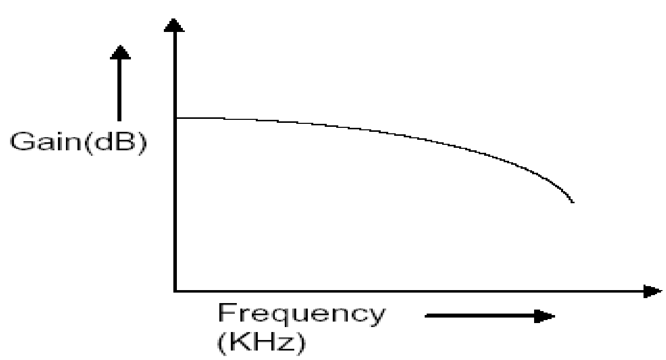


PROCEDURE:-

- Connect the power supply with proper polarity to the kit ACL-03 and switch it on.
- Keep all Switch Faults in OFF position.
- Select frequency range 1-10 KHz using JP4.
- Using pot P1 keep frequency at 1 KHz and using pot P2 keep amplitude at 0.1Vpp.
- Connect the output of function generator to the IN post of pre-emphasis circuit.
- Observe output voltage at the OUT post of pre-emphasis circuit.
- Vary the frequency in steps of 500Hz and note down the output voltage at
- The OUT post of pre-emphasis circuit.
- Plot the graph of output voltage v/s input frequency on graph paper.

DE-EMPHASIS CIRCUIT:-

At the receiver side a corresponding filter or “de-emphasis” circuit is required to reduce the treble response to correct level. Since most noise and hiss tend to be at the higher frequencies, the de-emphasis removes a lot of this. Pre-emphasis and de-emphasis thus allow an improved signal to noise ratio to be achieved while maintaining the frequency response of the original audio signal. The de-emphasis Stage is used after the detector stage.
The response of the emphasis circuit can be understood from the following Graph:



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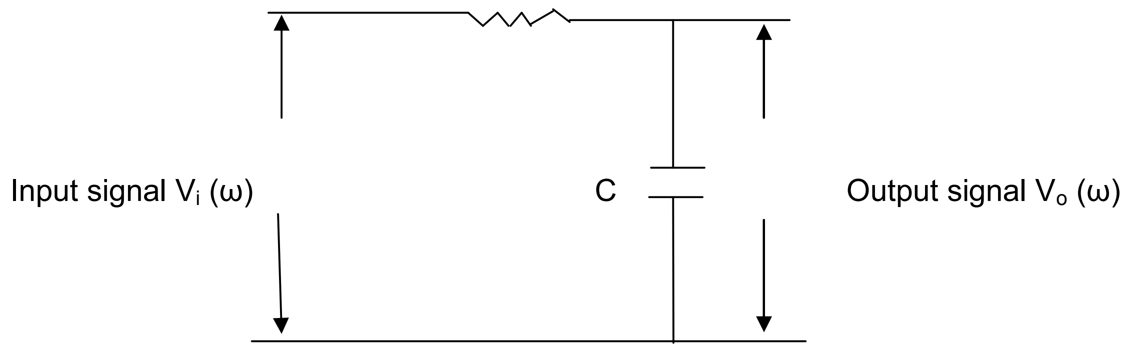


De-Emphasis Circuit

PROCEDURE:-

- Connect the power supply with proper polarity and switch it on.
- Select Sine wave signal using jumper JP1 shorted.
- Select frequency range 1-10 KHz using JP4.
- Using pot P1 keep frequency at 1 KHz and using pot P2 keep amplitude at 0.1 Vpp.
- Connect the output of function generator to the IN post of De-emphasis circuit.
- Observe output voltage at the OUT post of De-emphasis circuit.
- Plot the graph of output voltage v/s input frequency on graph paper.

RESULT:

VIVA QUESTIONS

1. What is pre-emphasis?

2. Which type of filter is used?

3. What is De-emphasis?

4. Which type of filter is used?

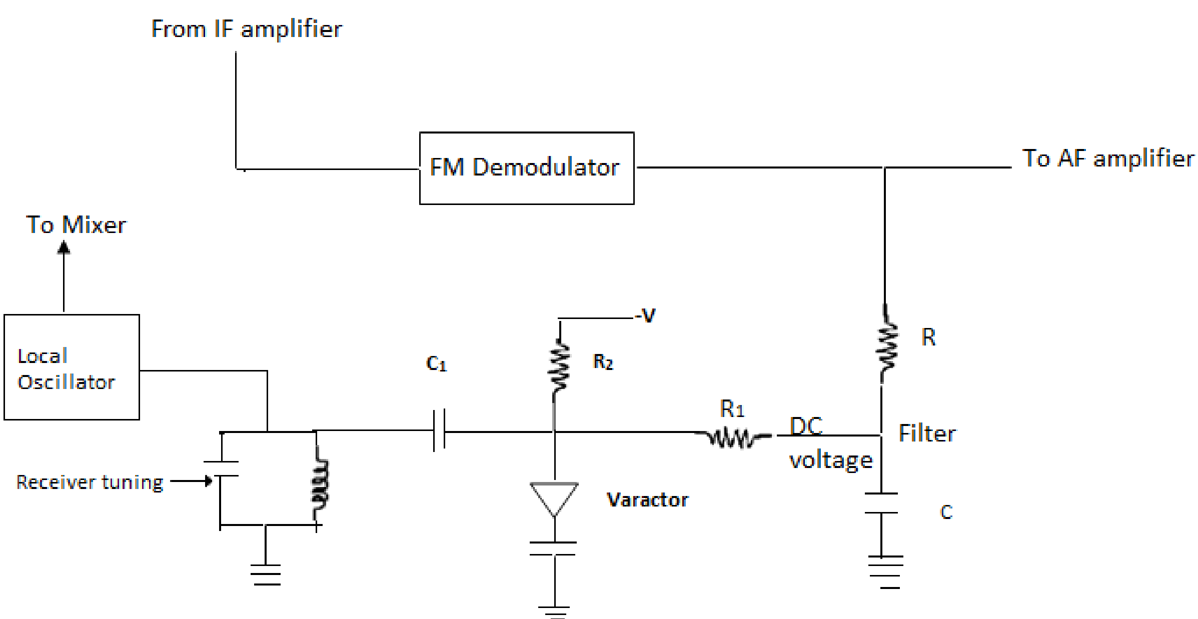
EXPERIMENT NO:-

AIM :– To study of AFC & AVC.

Apparatus Required: –

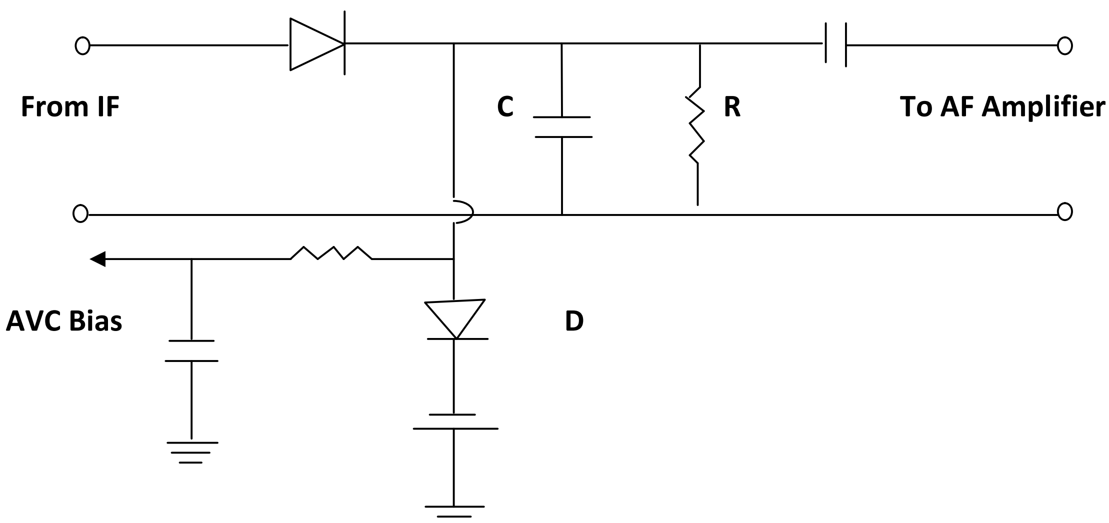
Theory: –

AFC- In AFC, some of the signal from the output of demodulator is filtered to get a D.C. voltage. This D.C. voltage is then used to control a varactor diode. As shown in fig. , the D.C. bias applied to the varactor will vary with the drift in frequency. It can be positive or negative. This D.C. voltage will then vary the capacitance of the varactor diode, which is connected across the oscillator tank circuit. Thus, the local oscillator frequency will be changed automatically to reduce the error to zero. If the local oscillator frequency increases above the desired frequency, then IF will increase. This produces a positive D.C. voltage at the output of the demodulator. This will cause the capacitance of varactor diode to increase and the local oscillator frequency to decrease. Thus, automatic frequency control is achieved.



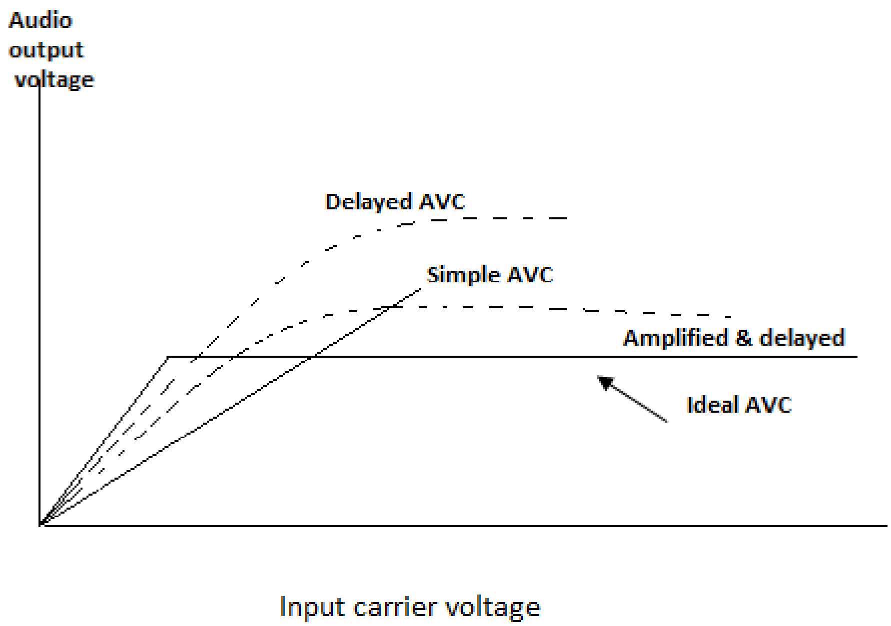
Automatic frequency control (AFC)

AVC- The automatic volume control (AVC) bias is obtained from this stage in order to keep the receiver output substantially constant with time for any variations in receiver input voltage . The magnitude of the receiver input voltage varies with time due to fading, or when the receiver is tuned from one station to another having different signal strength. The AVC eliminates the effect of these variations. Fig. provides a "simple AVC" bias if diode D in AVC circuit is removed. The AVC circuit samples a fraction of the detector output and converts it to AVC bias voltage. The AVC bias is applied to RF and IF stages to provide them a negative bias. As the input of receiver signal increases, the AVC bias voltage also increases, and in turn, the negative bias to RF and IF amplifiers is increased, thereby reducing their gain. The output of the receiver is, thus maintained constant The AVC is operative only if the signal is more than the diode bias voltage. The AVC characteristics can be further improved by amplifying the delayed AVC bias with the help of D.C. amplifier. The is known as amplified and delayed AVC.



<u>A Detector Circuit with AVC</u>

(a)


Audio
output
voltage
Delayed AVC
Simple AVC
Amplified & delayed
Ideal AVC
Input carrier voltage

(b)