### INDEX

<table>
<thead>
<tr>
<th>Section</th>
<th>Page No.</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Introduction</td>
<td>4</td>
</tr>
<tr>
<td>2. List of experiments</td>
<td>4</td>
</tr>
<tr>
<td>3. General guidelines for conducting an experiment</td>
<td>5</td>
</tr>
<tr>
<td>3.1 Simulation</td>
<td></td>
</tr>
<tr>
<td>3.2 Hardware</td>
<td></td>
</tr>
<tr>
<td>3.3 Do’s and Don’ts</td>
<td></td>
</tr>
<tr>
<td>4. Experiments</td>
<td>6</td>
</tr>
<tr>
<td>4.1 Amplitude modulation and demodulation</td>
<td>7-12</td>
</tr>
<tr>
<td>4.1.1 AIM</td>
<td></td>
</tr>
<tr>
<td>4.1.2 Theory</td>
<td></td>
</tr>
<tr>
<td>4.1.3 MATLAB program and description</td>
<td></td>
</tr>
<tr>
<td>4.1.4 Hardware</td>
<td></td>
</tr>
<tr>
<td>4.2 DSB-SC modulation and demodulation</td>
<td>13-16</td>
</tr>
<tr>
<td>4.2.1 AIM</td>
<td></td>
</tr>
<tr>
<td>4.2.2 Theory</td>
<td></td>
</tr>
<tr>
<td>4.2.3 MATLAB program and description</td>
<td></td>
</tr>
<tr>
<td>4.2.4 Hardware</td>
<td></td>
</tr>
<tr>
<td>4.3 SSB-SC modulation and Demodulation (Phase Shift method)</td>
<td>17-24</td>
</tr>
<tr>
<td>4.3.1 AIM</td>
<td></td>
</tr>
<tr>
<td>4.3.2 Theory</td>
<td></td>
</tr>
<tr>
<td>4.3.3 MATLAB program and description</td>
<td></td>
</tr>
<tr>
<td>4.3.4 Hardware</td>
<td></td>
</tr>
<tr>
<td>4.4 Frequency modulation and demodulation</td>
<td>25-29</td>
</tr>
<tr>
<td>4.4.1 AIM</td>
<td></td>
</tr>
<tr>
<td>4.4.2 Theory</td>
<td></td>
</tr>
<tr>
<td>4.4.3 MATLAB program and description</td>
<td></td>
</tr>
<tr>
<td>4.4.4 Hardware</td>
<td></td>
</tr>
<tr>
<td>4.5 Study of spectrum analyzer and study of AM and FM signals</td>
<td>30-32</td>
</tr>
<tr>
<td>4.5.1 AIM</td>
<td></td>
</tr>
<tr>
<td>4.5.2 Theory</td>
<td></td>
</tr>
<tr>
<td>4.5.3 MATLAB program and description</td>
<td></td>
</tr>
<tr>
<td>Section</td>
<td>Title</td>
</tr>
<tr>
<td>---------</td>
<td>-------</td>
</tr>
<tr>
<td>4.5.4</td>
<td>Hardware</td>
</tr>
<tr>
<td>4.6</td>
<td>Pre-emphasis and De-emphasis</td>
</tr>
<tr>
<td>4.7</td>
<td>Time division multiplexing and de-multiplexing</td>
</tr>
<tr>
<td>4.8</td>
<td>Frequency division multiplexing and de-multiplexing</td>
</tr>
<tr>
<td>4.9</td>
<td>Verification of sampling theorem</td>
</tr>
<tr>
<td>4.10</td>
<td>Pulse Amplitude modulation and demodulation</td>
</tr>
</tbody>
</table>
4.11 Pulse width modulation and demodulation 57-60
  4.11.1 AIM
  4.11.2 Theory
  4.11.3 MATLAB program and description
  4.11.4 Hardware
    - Apparatus
    - Circuit diagram
    - Procedure
    - Expected waveform

4.12 Pulse position modulation and demodulation 61-64
  4.12.1 AIM
  4.12.2 Theory
  4.12.3 MATLAB program and description
  4.12.4 Hardware
    - Apparatus
    - Circuit diagram
    - Procedure
    - Expected waveform

4.13 Frequency synthesizer 65-68
  4.13.1 AIM
  4.13.2 Theory
  4.13.3 MATLAB program and description
  4.13.4 Hardware
    - Apparatus
    - Circuit diagram
    - Procedure
    - Expected waveform

4.14 AGC characteristics 69-73
  4.14.1 AIM
  4.14.2 Theory
  4.14.3 MATLAB program and description
  4.14.4 Hardware
    - Apparatus
    - Circuit diagram
    - Procedure
    - Expected waveform

4.15 PLL and FM demodulator 74-79
  4.15.1 AIM
  4.15.2 Theory
  4.15.3 MATLAB program and description
  4.15.4 Hardware
    - Apparatus
    - Circuit diagram
    - Procedure
    - Expected waveform
1. Introduction

Analog communications lab is for B.Tech III year ECE I semester. The students learn Analog Communications theory subject in the same semester. The lab experiments are meant to equip the students with firm practical knowledge of the concerned subject. As listed in section-2, there are total fifteen (15) experiments in this lab which are according to JNTUH R-13 syllabus, out of which minimum 12 experiments are to be conducted.

General procedure for conducting an experiment is described in Section – 3. Each experiment will first be simulated using MATLAB, a simulation software program, which the students have already learned in Basic Simulation lab in II year I semester. The same experiment will then be realized using proper hardware, as described in detail for each experiment in section -4.

2. List of Experiments

1. Amplitude modulation and demodulation
2. DSB-SC modulator and detector
3. SSB-SC modulator and detector
4. Frequency modulation and demodulation
5. Study of Spectrum Analyzer and Analysis of AM and FM Signals
6. Pre-emphasis and de-emphasis
7. Time division multiplexing and de-multiplexing
8. Frequency division multiplexing and de-multiplexing
9. Verification of sampling theorem
10. Pulse amplitude modulation and demodulation
11. Pulse width modulation and demodulation
12. Pulse Position modulation and demodulation
13. Frequency synthesizer
14. AGC Characteristics
15. PLL & FM demodulator
3.0 General Guidelines for conducting an experiment

Each experiment first has to be simulated using MATLAB and then be realized in hardware as described in Section -4.

MATLAB, short for MATrix LABoratory is a programming package specifically designed for scientific calculations and I/O. It has literally hundreds of built-in functions for a wide variety of computations and many tool boxes designed for specific research disciplines, including statistics, optimization, solution of partial differential equations and data analysis.

3.1 Simulation

Each experiment will first be simulated with MATLAB software package. The MATLAB program code for each experiment is given in the following section. The students have to write the programs in PC and execute them using MATLAB software and make sure that they get the expected waveforms as shown in this manual.

The students are also advised to change certain parameters in the MATLAB program and see their impact on the output waveforms.

3.2 Hardware

After completion of the simulation, each experiment has to be realized in hardware as described for all experiments in section – 4. Out of fifteen experiments, fourteen experiments are to be conducted with the PHYSITECH lab kits along with the additional instruments as mentioned for each experiment in section – 4. The fifth experiment, “Study of Spectrum Analyzer and Analysis of AM and FM Signals” does not require the PHYSITECH experiment kit.

3.3 Do’s and Don’ts

The students are to follow the given general Do’s and Don’ts for simulation lab.

Do’s:
1. Enter in to the simulation lab in time.
2. Wear student identity badges round your neck before entering the lab.
3. Keep silence in the lab
4. Follow the instructions of the lab in-charges and lab supervisor.
5. Always save your input files and results in the prescribed directory.

Don’ts:
1. Do not use internet or open any other programs other than MATLAB.
2. Do not mishandle or rough handle the keyboard of CPU.
3. Do not use pen drive or card reader without the permission of the lab in-charge.
4. Do not make noise in the lab.

The students are to follow the given general Do’s and Don’ts for hardware lab

Do’s
1. Enter the hardware lab in time.
2. Wear shoes/sandals in the lab.
3. Wear student identity badge before entering the lab
4. Follow the instructions of the lab in-charges and lab supervisor.

Don’ts:
1. Do not make noise in the lab.
2. Do not switch the power supply until you finish all the connections.
3. Do not remove connecting wires or probes when the power is ON.
4. Do not mishandle the kits and instruments.
5. Do not pluck the ICs or other components from the trainer kits.
6. Before turning on the power, show your experimental arrangement to the lab in-charges or lab supervisor.
4.0 Experiments
EXPERIMENT – 1

4.1 Amplitude modulation and demodulation

4.1.1 Aim: To study the function of Amplitude Modulation & Demodulation (under modulation, perfect modulation & over modulation) and also to calculate the modulation index.

4.1.2 Theory:

Amplitude modulation (AM) is defined as a process in which the amplitude of the carrier wave $c(t)$ is varied about a mean value, linearly with the base band signal $m(t)$. An AM wave may thus be described, in its most general form, as a function of time as follows.

$$S(t) = AC \cdot [1 + K_a m(t)] \cos(2 \pi f_c t)$$

Where $K_a$ Amplitude Sensitivity of the modulator

$S(t)$ Modulated signal

$AC$ Carrier Amplitude

$m(t)$ Message Signal

The amplitude of $K_a m(t)$ is always less than unity, that is $|K_a m(t)| < 1$ for all $t$

It ensures that the function $1 + K_a m(t)$ is always positive. When the amplitude sensitivity $K_a$ of the modulator is large enough to make $|K_a m(t)| > 1$ for any $t$, the carrier wave becomes over modulated, resulting in carrier phase reversals. Whenever the factor $1 + K_a m(t)$ crosses zero. The modulated wave then exhibits envelope distortion as shown in fig. below.

The absolute maximum value of $K_a m(t)$ multiplied by 100 is referred to as the percentage modulation.

$$\frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

or percentage modulation =

The carrier frequency $f_c$ is much greater than the highest frequency component $w$ of the message signal $m(t)$, that is $f_c >> W$ Where $W$ is the message bandwidth.

If this condition is not satisfied, envelope cannot be visualized (and therefore detected) satisfactorily.
PHYSITECH'S modulation and demodulation trainer has a carrier generator, which generates carrier wave of 100KHz when the trainer is switched on.

The blocks, carrier generator, modulator and demodulator are provided with built in supplies, no supply connections are to be given externally.

4.1.3 MATLAB Program and description:

Program:
A=input('enter the carrier signal peak')
B=input('enter the baseband signal peak')
f1=input('enter the baseband signal frequency')
f2=input('enter the carrier signal frequency')
fs=input('enter the sampling frequency')
t=0:0.001:1;
M=cos(2*pi*f1*t);
N=cos(2*pi*f2*t);
m=B/A;
O=A*(1+m*M).*N;
O1=O;
for i=1:length(t)
if O1(i)<=0
O1(i)=0;
end

\[
\text{[den, num]} = \text{butter}(2, 2*\pi*f1/fs);
\text{M1} = \text{filter(den, num, O1)};
\text{M11} = \text{filter(den, num, M1)};
\text{M12} = \text{filter(den, num, M11)};
\text{M13} = \text{filter(den, num, M12)};
\text{subplot(5,1,1)}
\text{plot(t, M)}
\text{title('Baseband signal')}
\text{subplot(5,1,2)}
\text{plot(t, N)}
\text{title('Carrier signal')}
\text{subplot(5,1,3)}
\text{plot(t, O)}
\text{title('Modulated Carrier')}
\text{subplot(5,1,4)}
\text{plot(t, O1)}
\text{title('Rectified Modulated Signal')}
\text{subplot(5,1,5)}
\text{plot(t, M13)}
\text{title('Demodulated Signal')}
\]

enter the carrier signal peak \( A = 5 \)
enter the baseband signal peak \( B = 2 \)
enter the baseband signal frequency \( f1 = 10 \)
enter the carrier signal frequency \( f2 = 100 \)
enter the sampling frequency \( fs = 1000 \)
4.1.4: Hardware
- Apparatus

1. PHYSITECH’s Amplitude modulation and Demodulation trainer kit.
2. Function Generator
3. Oscilloscope (DSO)
4. Connecting wires

-circuit diagram:

In amplitude Modulation and demodulation trainer, the IC 8038 is used as a carrier generator. It provides 100KHz sine wave as carrier output at pin2. In modulator section series modulation is used. The first transistor works as RF amplifier and second transistor as the modulator. The potentiometer controls the percentage of modulation. In this circuit it is not possible to obtain 100% modulation. This is due to the RF amplifiers junction capacitance which allows RF to feed through to the output when the transistor is normally shut off. By varying the potentiometer, percentage modulation is changed. At certain point, the AM waveform is over modulated. This causes severe distortion of the output wave. Diode detector is used in demodulator section, to get demodulated output.
### Demodulation circuit diagram

- **Power supply**
- **AF generator**
- **RF generator**
- **Modulator**
- **Demodulator**
- **Oscilloscope**

**Block diagram for Experiment**

**-procedure**

1. Switch on the trainer and check the O/P of carrier generator on oscilloscope.
2. Connect around 1KHz with 2 Volts A.F signal at AF I/P to the modulator circuit.
3. Connect the carrier signal at carrier I/P of modulator circuit.
4. Observe the modulator output signal at AM O/P Spring by making necessary changes in A.F. signal.
5. Vary the modulating frequency and amplitude and observe the effects on the modulated waveform.
6. The depth of modulation can be varied using the variable knob (potentiometer) provided at A.F. input.
7. The percentage of modulation or modulation factor can be calculated using the following formulas.

   \[
   \% \text{ of Modulation} = \frac{v_{\text{max}} - v_{\text{min}}}{v_{\text{max}} + v_{\text{min}}} \times 100
   \]

   or \ Modulation factor \ = \frac{v_{\text{max}} - v_{\text{min}}}{v_{\text{max}} + v_{\text{min}}}

8. Connect the output of the modulator to the input of demodulator circuit and observe the output.
Expected waveforms:

UNDER MODULATION

OVER MODULATION

Demodulated signal

MODULATING WAVE
4.2 DSB-SC modulation and demodulation

4.2.1 AIM: To study Balanced modulator for DSB-SC modulation, demodulation.

4.2.2 Theory:

In Balanced modulator, two non-linear devices are connected in the balanced mode, so as to suppress the carrier wave.

The Balanced Modulator consists of summing devices (operational amplifiers) and two matched nonlinear elements. If x(t) is band limited to fx and if fc>2fx, then the band pass filter output will be the desired product signal.

Figure shows IC that has been specifically designed for use as balanced modulators. Figure is the 1496 balanced modulator which is manufactured by Motorola, National, and Signetics. This device uses a differential amplifier configuration. Its carrier suppression is rated at a minimum of -50dB with a typical value -65dB at 500 KHz.

PHYSITECHS trainer contains a balanced modulator using a 1496 integrated circuit. You will verify that it does suppress the carrier and also adjust it for optimum carrier suppression.

4.2.3 MATLAB program

Program:

f1=input('enter the baseband signal frequency')
f2=input('enter the carrier signal frequency')
T=input('enter the duration over which the signal is to be plotted')
fs=input('enter the sampling frequency')
t=0:T/fs:T;
M=cos(2*pi*f1*t);
N=cos(2*pi*f2*t);
O=M.*N;
P=O.*N;
C=input('enter the value of the capacitor of the filter')
R=1/(2*pi*f1*C);
H=(1/(R*C))*exp(-t/(R*C));
h=conv(H,conv(P,H));
t1=t;
for i=length(t)+1:length(h)
    t1(i)=0;
end
subplot(2,2,1)
plot(t,M)
title('Baseband Signal')
subplot(2,2,2)
plot(t,N)
title('Carrier Signal')
subplot(2,2,3)
plot(t,O)
title('Modulated Carrier')
subplot(2,2,4)
plot(t1,h)
title('Demodulated Signal')

Result:
enter the baseband signal frequency
f1 = 10
enter the carrier signal frequency
f2 = 100
enter the duration over which the signal is to be plotted
T = 0.4000
enter the sampling frequency
fs=1000
enter the value of the capacitor of the filter
C = 1.0000e-008

![Graphs showing baseband signal, carrier signal, modulated carrier, and demodulated signal.](image-url)
4.2.4 Hardware:
- Apparatus

1. PHYSITECH's Balanced Modulator trainer.
2. Function Generator (2)
3. CRO
4. BNC Probes

- Circuit diagram
Procedure:

1. Switch on the trainer.
2. Connect 200 Hz sine wave, and 100 KHz square wave from the function Generators. Adjust R1, (1k linear pot). Connect your oscilloscope to the output.
3. Vary R1 (1K) both clockwise and counter clockwise. Observe the output.
4. Disconnect the SINE input to R1(1K). The output should now be close to zero.
5. Increase the oscilloscope's vertical input sensitivity to measure the output voltage. E out carrier only.
6. Set the vertical input control to 1V/cm. Connect the SINE input to R1(1K) and adjust R1 for maximum output without producing clipping. Measure the peak side band output voltage.

Epk sidebands = ______________

7. Calculate the carrier suppression in dB.

\[
\text{dB} = -20 \log \left( \frac{E_{\text{pk sideband}}}{E_{\text{out carrier only}}} \right)
\]

- Expected waveforms:-

![Carrier Signal](image1)

![Modulating Wave](image2)

![DSB_SC Output](image3)
EXPERIMENT - 3

4.3 SSB-SC modulation and demodulation (Phase shift method)

4.3.1 Aim: To generate SSB using phase method and demodulation of SSB signal using Synchronous detector.

4.3.2 Theory

The phase shift method makes use of two balanced modulators and two phase shift networks as shown in fig. One of the modulators receives the carrier signal shifted by 90° and the modulating signal with 0° (sine)phase shift, whereas the other receives modulating signal shifted by 90° (co-sine) and the carrier (RF) signal with 0° phase shift voltage.

Both modulators produce an output consisting only of sidebands. It will be shown that both upper sidebands lead the input carrier voltage by 90°. One of the lower sidebands leads the reference voltage by 90°, and the other lags it by 90°. The two lower sidebands are thus out of phase, and when combined in the adder, they cancel each other. The upper sidebands are in phase at the adder and therefore they add together and gives SSB upper side band signal.

When they combined in the subtractor, the upper side bands are cancel because in phase and lower side bands add together and gives SSB lower side band signal

4.3.3 MATLAB Program

Program:

```matlab
f1=input('enter the base band signal frequency')
f2=input('enter the carrier signal frequency')
t=0:0.001:0.4;
fs=input('enter sampling frequency')
M=cos(2*pi*f1*t);
N=cos(2*pi*f2*t);
DSB1=M.*N;
M1=cos(2*pi*f1*t-(pi/2));
N1=cos(2*pi*f2*t-(pi/2));
DSB2=M1.*N1;
USB=DSB1-DSB2;
LSB=DSB1+DSB2;
subplot(5,1,1)
plot(t,M,'k',t,M1,'--b')
title('Base band signal and its Hilbert Transform')
subplot(5,1,2)
plot(t,N,'k',t,N1,'--b')
title('Carrier Signal and its Hilbert Transform')
subplot(5,1,3)
plot(t,USB)
title('Upper side band signal')
```

MIST, Hyderabad – ECE Department
subplot(5,1,4)
plot(t,LSB)
title('Lower Side band Signal')
USBMULT=USB.*N;
[den num]= butter(2,(2*pi*f1)/fs);
Filter1=filter(den,num,USBMULT);
Filter2=filter(den,num,Filter1);
Filter3=filter(den,num,Filter2);
Filter4=filter(den,num,Filter3);
subplot(5,1,5)
plot(t,Filter4)
title('Demodulated Signal from USB')

Results:
enter the base band signal frequency
f1 =25
enter the carrier signal frequency
f2 = 50
enter sampling frequency
fs = 1000

4.3.4 Hardware:
   - Apparatus
     1. PHYSITECH’S SSB trainer Board.
     2. Dual trace Oscilloscope
     3. Frequency Counter.
     4. Patch Chords
- Circuit Diagram and description

SSB MODULATION

Ssb demodulation

SSB IP – Synchronous detector – Demodulated O/P

Carrier Signal IP
Circuit diagram

DSB-SC A Modulator

DSB-SC B Modulator
Summer Circuit

Demodulator Circuit
Circuit description:

**RF generator:**
Colpitts oscillator using FET is used here to generate RF signal of approximately 100KHz frequency to use as carrier signal in this experiment. Phase shift networks are included in the same block to produce another carrier signal of same frequency with 90° out of phase. As individual controls are provided to vary the output voltage. Facility is provided to adjust phase of the output signal.

**AF generator:**
This is a sine co-sine generator using OP-AMP. IC TL 084 is used as an active component. TL 084 is a FET input general purpose quad OP-AMP integrated circuit. A three position switch is provided to select output frequency. An individual controls are provided to vary the output voltage. AGC control is provided to adjust the signal shape.

**Balanced modulator:**
This has been developed using MC 1496 IC. MC 1496 is a monolithic integrated circuit Balanced modulator/Demodulator, is versatile and can be used up to 200MHz. These modulators are used in this experiment to produce DSB-SC signals. Control is provided to balance the output.

**Synchronous detector:**
The base band signal m(t) can be uniquely recovered from a DSB-SC signal s(t) by first multiplying s(t) with a locally generated sine wave carrier and then low pass filtering the product. It is assumed that the local oscillator signal is exactly coherent or synchronous, in both frequency and phase with the carrier wave c(t) used in the balanced modulator to generate s(t). This method of demodulation is known as coherent detection or synchronous detection.

In this unit IC MC 1496 is used as synchronous demodulator. The MC 1496 is a monolithic balanced modulator/balanced demodulator, is versatile and can be used up to 200MHz. On board generated carrier (which is used in the modulator) is used as synchronous signal.

**Summer and subtractor:**
These Circuits are simple summing and subtracting amplifiers using OP-AMP.

**Procedure:**
1. Study the circuit operation of SSB system thoroughly.
2. Observe the output of the RF generator using CRO. There are two outputs from the RF generator, one is direct output and the another is 90° phase shift with the direct output. The output frequency is 100KHz and the amplitude is ≥ 0.2 Vpp (potentiometers are provided to vary the output amplitude).
3. Observe the output of the AF generator using CRO. There are two outputs from the AF generator, one is direct output and the another is 90° phase shift with the direct output. AGC potentiometer is
provided to adjust the gain of the oscillator (or to set the output to good shape). And the amplitude is 10 Vpp (potentiometers are provided to vary the output amplitude).

4. Measure and record the RF signal frequency using frequency counter.

5. Set the amplitudes of the RF signals to 0.1 Vpp and connect 0° phase shift signal to one balanced modulator and 90° phase shift to another balanced modulator as shown in figure.

6. Select the required frequency of the AF generator with the help of switch and adjust the AGC potentiometer until the output amplitude is \( \approx 10 \) Vpp (when amplitude controls are in maximum condition).

7. Measure and record the AF signals frequency using frequency counter.

8. Set the AF signal amplitudes to 8 Vpp using amplitude control and connect to the balanced modulators as shown in below figure.

9. Observe the outputs of both the balanced modulators simultaneously using Dual trace Oscilloscope, and adjust the balance control until you get the output wave forms (DSB-SC) as shown in figure.

10. To get SSB lower side band signal, connect balanced modulator outputs (DSB-SC signals) to subtractor.

11. Measure and record the SSB signal frequency using counter.

12. Calculate theoretical frequency of SSB (LSB) and compare it with the practical value.

\[
\text{LSB} = \text{RF frequency} - \text{AF frequency}
\]

Ex: If RF frequency is 100KHz and AF frequency is 2KHz Then LSBSB = 100KHz - 2KHz = 98 KHz

13. To get SSB upper side band signal, connect the output of the balanced modulator to the summer circuit.

14. Measure and record the SSB upper side band signal frequency using counter.

15. Calculate theoretical value of the SSB(USB) frequency and compare it with practical value.

\[
\text{USB} = \text{RF frequency} + \text{AF frequency}
\]

Ex: If RF frequency is 100KHz and AF frequency is 2KHz Then USB = 100KHz + 2KHz = 102KHz.

**Demodulation of ssb signal:**

16. Connect SSB signal from the summer (or) subtractor to the SSB signal input of the synchronous detector and RF signal (0°) to the RF input of the synchronous detector.

17. Observe the detector output using CRO and compare it with the modulating signal (AF signal).

18. Observe the SSB signal for the different frequencies of the modulating (AF) signal.
**Expected waveforms**

- **RF/CARRIER FREQUENCY** (0 PHASE)
- **RF/CARRIER FREQUENCY** (90 PHASE)
- **AF/MODULATING FREQUENCY** (0 PHASE)
- **AF/MODULATING FREQUENCY** (90 PHASE)
- **DSBSC – SC OUTPUT (A & B)**
- **SINGLE SIDE BAND** (SSB - SC)
EXPERIMENT – 4

4.4 Frequency modulation and demodulation.

4.4.1 Aim: To study the functioning of frequency modulation & demodulation and to calculate the modulation index.

4.4.2 Theory:

FM is a system in which the amplitude of the modulated carrier is kept constant, while its frequency and rate of change are varied by the modulating signal.

By the definition of FM, the amount by which the carrier frequency is varied from its unmodulated value, called the deviation, is made proportional to the instantaneous amplitude of the modulating voltage. The rate at which this frequency variation changes or takes place is equal to the modulating frequency.

FM is that form of angle modulation in which the instantaneous frequency \( f_i(t) \) is varied linearly with the message signal \( m(t) \), as

\[
f_i(t) = f_C + K_f m(t)
\]

The term \( f_C \) represents the frequency of the unmodulated carrier, and the constant \( K_f \) represents the frequency sensitivity of the modulator expressed in Hertz per volt.

Unlike AM, the spectrum of an FM signal is not related in a simple manner to that of modulating signal, rather its analysis is much more difficult than that of an AM signal.

Generation of FM signals:

There are essentially two basic methods of generating frequency modulated signals, namely, direct FM and indirect FM. In the direct method, the carrier frequency is directly varied in accordance with the input base band signal, which is readily accomplished using a voltage-controlled oscillator. In the indirect method, the modulating signal is first used to produce a narrow band FM signal, and frequency multiplication is next used to increase the frequency deviation to the desired level. The indirect method is the preferred choice for FM when the stability of carrier frequency is of major concern as in commercial radio broadcasting.

Indirect FM:

A simplified block diagram of an indirect FM system is shown in fig below.
Demodulation of FM signals:

Frequency demodulation is the process that enables us to recover the original modulating signal from a frequency-modulated signal. Here we describe a direct method of frequency demodulation involving the use of a popular device known as a frequency discriminator, whose instantaneous output amplitude is directly proportional to the instantaneous frequency of the input FM signal.

Basically, the frequency discriminator consists of a slope circuit followed by an envelope detector.

![Frequency Discriminator Diagram]

### 4.4.3 MATLAB Program

Program:

```matlab
clc;
clear all;
close all;
fs=10000;
ac=1;
am=1;
fm=35;
fc=500;
b=10;
t=(0:0.1*fs)/fs;
wc=2*pi*fc;
wm=2*pi*fm;
mt=am*cos(wm*t);
subplot(4,1,1);
plot(t,mt);
title('modulating signal');
ct=ac*cos(wc*t);
subplot(4,1,2);
plot(t,ct);
title('carrier signal');
st=ac*cos((wc*t)+b*sin(wm*t));
subplot(4,1,3);
plot(t,st);
title('modulated signal');
d=demod(st,fc,fs,'fm');
subplot(4,1,4);
plot(t,d);
title('demodulated signal');?>```
4.4.4 Hardware

- **Apparatus**

1. PHYSITECH’S frequency modulation and demodulation trainer.
2. CRO
3. Function generator
4. Connecting wires and probes

- **Circuit diagram and description**

Frequency modulation circuit diagram:

![Modulating signal sine wave input](image1)

![Demodulated signal](image2)
**Frequency demodulation circuit diagram:**

![Circuit Diagram]

**Procedure:**

1. Switch on the **PHYSITECH**'s experimental board.
2. Connect Oscilloscope to the FM O/P and observe the carrier frequency at that point without any A.F. input.
3. Connect around 7KHz sine wave (A.F. signal) to the input of the frequency modulator (At AF input).
4. Now observe the frequency modulation output on the 1st channel of on CRO and adjust the amplitude of the AF signal to get clear frequency modulated waveform.
5. Vary the modulating frequency (A.F.Signal), and amplitude and observe the effects on the modulated waveform.
6. Connect the FM o/p to the FM i/p of De-modulator.
7. Vary the potentiometer provided in the demodulator section.
8. Observe the output at demodulation o/p on second channel of CRO.
- **Expected waveforms**

RF/CARRIER SIGNAL

AF/MODULATING SIGNAL

FREQUENCY MODULATED OUTPUT

FREQUENCY DEVIATION FOR ONE CYCLE

FREQUENCY DEMODULATED OUTPUT
EXPERIMENT - 5

4.5 Study of Spectrum Analyzer

4.5.1 Aim: To study the Spectrum analyzer

4.5.2 Theory
To study the frequency response of the signals, we use spectrum analyzers. In theory, applying Fourier transform on time domain signals gives their frequency domain response. In hardware, the frequency response of the time domain signals is obtained by spectrum analyzer.

4.5.3 MATLAB Program

Program:
%program of spectrum analyzer and analysis of am and fm signals
close all
clear all
clc
Fs = 100;           %sampling frq
t = [0:2*Fs+1]'/Fs;
Fc = 10;            % Carrier frequency
x = sin(2*pi*2*t);      % message signal
Ac=1;
%compute spectra of am
xam=ammod(x,Fc,Fs,0,Ac);
zam = fft(xam);
zam = abs(zam(1:length(zam)/2+1));
frqam = [0:length(zam)-1]*Fs/length(zam)/2;
% compute spectra of dsbsc
ydouble = ammod(x,Fc,Fs, 3.14,0);
zdouble = fft(ydouble);
zdouble = abs(zdouble(1:length(zdouble)/2+1));
frqdouble = [0:length(zdouble)-1]*Fs/length(zdouble)/2;
% compute spectra of ssb
ysingle = ssbmod(x,Fc,Fs,Fc,0,'upper');
zsingle = fft(ysingle);
zsingle = abs(zsingle(1:length(zsingle)/2+1));
frqsingle = [0:length(zsingle)-1]*Fs/length(zsingle)/2;
% Plot spectrums of am dsbsc and ssb
figure;
subplot(3,1,1);
plot(frqam,zam);
title('Spectrum of am signal');
subplot(3,1,2);
plot(frqdouble,zdouble);
title('Spectrum of double-sideband signal');
subplot(3,1,3); plot(frqsingle,zsingle);
title('Spectrum of single-sideband signal');
% spectrum of fm
xfm=fmmod(x,Fc,Fs,10);
zf = fft(xfm);
zf = abs(zf(1:length(zf)/2+1));
frqfm = [0:length(zf)-1]*Fs/length(zf)/2;
figure;
plot(frqfm,zf);
title('Spectrum of fm signal');
**Expected waveforms:**

4.5.4 Hardware

- **Apparatus**
  1. RF generator
  2. Function generator
  3. Spectrum analyzer
- *Circuit diagram*

![Circuit diagram]

- **Procedure:**
  To check the frequency spectrum RF signal, the RF generator’s output is connected to the spectrum analyzer as shown in the picture above.

  The frequency spectrums Square waves, triangular waves can also be analyzed with the Function generator connected with the spectrum analyzer.

- **Expected waveforms:**

![RF Spectrum]
EXPERIMENT – 6

4.6 Pre-emphasis and De-emphasis

4.6.1 AIM: To study the functioning of Pre-Emphasis and De-Emphasis circuits

4.6.2 Theory

Frequency modulation is much more immune to noise than amplitude modulation and is significantly more immune than phase modulation. The threshold effect is more serious in FM as compared to AM, because in FM, the signal to noise ratio at the input of a detector, at which threshold effect starts, is higher. Lower the threshold level, better is the system because threshold can be avoided at a comparatively lower ratio, and a small signal is needed to avoid threshold for an equivalent noise power. Hence, it is desirable to lower the threshold level in the FM receivers. The process of lowering the threshold level is known as threshold improvement, or threshold reduction. Two methods are used for the improvement of the threshold.

Pre-Emphasis and De-Emphasis circuits.

FMFB (Frequency Modulation with Feed Back.)

PRE-EMPHASIS AND DE-EMPHASIS:

The noise triangle shows, noise has a greater effect on the higher modulating frequencies than on the lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the signal-to-noise ratio. This boosting of the higher modulating frequencies, in accordance with a prearranged curve, is termed pre-emphasis, and the compensation at the receiver is called de-emphasis.

4.6.3 MATLAB program and description

% program for Pre-Emphasis and De-Emphasis
close all
clear all
clc
num_samples = 2^13;
fs=5000;
Ts=1/fs;
fm1=20;
fm2=30;
fc=200;
t=(0:num_samples-1)*Ts;
f=(-num_samples/2:num_samples/2-1)*fs/num_samples;
mt=sin(2*pi*fm1*t);
Mf=fftshift(abs(fft(mt)));
f_cutoff_pe=15;
Wn_pe=f_cutoff_pe/(fs/2);
[b_pe,a_pe]=butter(1,Wn_pe);
[H_pe,W]=freqz(a_pe,b_pe);
a_de=b_pe;
b_de=a_pe;
[H_de,W]=freqz(a_de,b_de);
mt_pe=filter(a_pe,b_pe,mt);
Mf_pe=fftshift(abs(fft(mt_pe))); figure(1) subplot(211);plot(t,mt) axis([0 0.6 min(mt)-1 max(mt)+1]) grid on; title('Modulating Signal (Time Domain)') subplot(212);plot(f,Mf) grid on;axis([-50 50 0 max(Mf)+100]) title('Modulating Signal (Frequency Domain)') figure(2) subplot(211) semilogx(W*pi*(fs/2),abs(H_pe),'m','linewidth',2) axis([0 fs/2 0 50]) grid on;title('Pre-emphasis Filter Magnitude Response') subplot(212) semilogx(W*pi*(fs/2),abs(H_de),'m','linewidth',2) axis([0 fs/2 0 1]) grid on;title('De-emphasis Filter Magnitude Response') results:
4.6.4 Hardware

- Apparatus
  1. PHYSITECH'S Pre-Emphasis and De-Emphasis trainer
  2. Function generator
  3. CRO
  4. Connecting Wires

- Circuit diagram

PRE-EMPHASIS CIRCUIT

DE-EMPHASIS CIRCUIT
- **Procedure:**

1. Switch on **PHYSITECH**’S Pre-emphasis and De-Emphasis Trainer.
2. Give the input from signal generator to AF I/P of pre-emphasis circuit. By varying the amplitude knob set the input voltage to some milli volts say (4mV,6mV, etc.).
3. Observe the output waveform on CRO channel-1, by connecting either 75mH or 50mH.
4. The output of pre-emphasis circuit must be below the audio frequency range.
5. Connect the output of Pre-Emphasis to the I/P of De-emphasis circuit.
6. Observe the De-Emphasis output at AF O/P of De-Emphasis circuit.
7. Measure the output voltage in CRO for each frequency and note down the values.
8. Calculate the attenuation and log F Values.
9. Plot the graph between frequencies on X-axis and attenuation on Y-axis to show the emphasis curve.
10. Various values of R and C are available so that, the time constant is suitably selected depending upon the application.

- **Expected waveforms**

![Pre-emphasis](image1)

![De-emphasis](image2)

**Observations**

<table>
<thead>
<tr>
<th>S.No</th>
<th>Input Frequency (50Hz to 20KHz)</th>
<th>Output Voltage (volts)</th>
<th>GAIN 20 log (Vo/ V1) db</th>
</tr>
</thead>
<tbody>
<tr>
<td>V1 =2v</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
EXPERIMENT - 7

4.7 Time division multiplexing and de-multiplexing

4.7.1 Aim To study time division multiplexing and demultiplexing.

4.7.2 Theory

The Sampling Theorem provides the basis for transmitting the information contained in a band-limited message signal m(t) as a sequence of samples of m(t) taken uniformly at a rate that is usually slightly higher than the nyquist rate. An important feature of the sampling process is a conservation of time. That is, the transmission of the message samples engages the communication channel for only a fraction of the sampling interval on a periodic basis, and in this way some of the time interval between adjacent samples is cleared for use by other independent message sources on a time-shared basis. We thereby obtain a time-division multiplexing (TDM) system, which enables the joint utilization of a common communication channel by a plurality of independent message sources without mutual interference among them.

The TDM system is highly sensitive to dispersion in the common channel, that is, to variations of amplitude with frequency or lack of proportionality of phase with frequency. Accordingly, accurate equalization of both magnitude and phase response of the channel is necessary to ensure a satisfactory operation of the system. Unlike FDM, TDM is immune to nonlinearities in the channel as a source of cross-talk. The reason for this is, that different message signals are not simultaneously applied to the channel.

The primary advantage of TDM is that several channels of information can be transmitted simultaneously over a single cable.

4.7.3 MATLAB program and description

N=input('enter the number of signals to be multiplexed')
f=zeros(1,N);
for i=1:N
  f(i)=input('enter the frequency of the signal')
end
fs=2*max(f);
T=input('enter the duration over which the signal is to be plotted')
t=0:T/fs:T;
Q=zeros(1,N*length(t));
R=Q;
S=Q;
for i=1:N
  Q(:,((i-1)*length(t))+1:i*length(t))=cos(2*pi*f(i)*t);
end
j=1;
for i=0:N:length(Q)-N
  for n=1:N
    ...
\[ R(i+n) = Q(j+(n-1)\times \text{length}(t)) \]
\[ S(j+(n-1)\times \text{length}(t)) = R(i+n) \]
end
\[ j = j+1; \]
end
\[ t_1 = 0:T/\text{fs}:N*T; \]
for \( i = \text{length}(t_1)+1:\text{length}(Q) \)
\[ t_1(i) = t_1(\text{length}(t_1)) \]
end
subplot(N+2,1,1)
plot(t1,Q)
title('Signals to be Multiplexed')
subplot(N+2,1,2)
stem(t1,R)
title('Multiplexed Signal')
for \( i = 1:N \)
subplot(N+2,1,i+2)
plot(t,S(:,((i-1)*\text{length}(t))+1:i*\text{length}(t)))
title('De-multiplexed Signal')
end

\textit{results:}
enter the number of signals to be multiplexed
\( N = 2 \)
enter the frequency of the signal
\( f = 25 \quad 50 \)
enter the duration over which the signal is to be plotted.
\( T = 0.2000 \)
4.7.4 **Hardware**

- **Apparatus**
  1. PHYSITECH’S Time-Division Multiplexing and Demultiplexing trainer.
  2. CRO
  3. BNC Probes and Connecting wires.

- **Circuit diagram**

![Circuit Diagram]

- **procedure**
  1. Switch on PHYSITECH’S Time division multiplexing and demultiplexing trainer.
  2. Connect the sine wave to ch₁, square wave to ch₂ and Triangle wave form to ch₃ terminals of 8 to 1 multiplexer.
  3. Observe the Multiplexer output on channel 1 of a CRO.
  4. Connect Mux output to demux input.
  5. Observe corresponding signal outputs at channel 2 of CRO.
-Expected waveforms

Multiplexed Waveform
EXPERIMENT - 8

4.8 Frequency division multiplexing and de-multiplexing

4.8.1 Aim: To study Frequency division multiplexing and de-multiplexing

4.8.2 Theory

In any communication system, there will be a transmitter, a receiver and channel. In a multiple subscriber two way communication system, there will be more transmitters at the transmitting end and more receivers at the receiving end. In general channel is only one. For the maximum utilization of the channel, all the signals which are meant to be transmitted, which are occupying the same bandwidth, need to be multiplexed to avoid mixing with each other and to avoid noise interruption. This multiplexing can be done either in time domain or in frequency domain. If it is done in time domain, it is called Time Division Multiplexing (TDM), which is described in experiment-7. On the other hand, if the multiplexing is done in the frequency domain, as in the case with AM, DSB-SC, SSB-SC or FM, it is called Frequency Division Multiplexing (FDM). At the destination end, a reverse process of multiplexing need to be done. This is known as de-multiplexing. In this experiment we will study Frequency division multiplexing and de-multiplexing.

4.8.3 MATLAB program and description

Program:

```matlab
%program for FDM and Demultiplexing
close all
clear all
clc
Fs = 200;           %sampling frq
t = [0:2*Fs+1]'./Fs;
Fc1 = 10;            % Carrier frequency
F1 = 2;
x1 = sin(2*pi*F1*t);      % message signal
Fc2 = 30;            % Carrier frequency
F2 = 4;
x2 = sin(2*pi*F2*t);      % message signal
% compute spectra of message signal
z1 = fft(x1);
z1 = abs(z1(1:length(z1)/2+1));
frq1 = [0:length(z1)-1]*Fs/length(z1)/2;
z2 = fft(x2);
z2 = abs(z2(1:length(z2)/2+1));
frq2 = [0:length(z2)-1]*Fs/length(z2)/2;
% compute spectra of dsbsc
ydouble1 = ammod(x1,Fc1,Fs);
zdouble1 = abs(ydouble1(1:length(zdouble1)/2+1));
frqdouble1 = [0:length(zdouble1)-1]*Fs/length(zdouble1)/2;
ydouble2 = ammod(x2,Fc2,Fs);
zdouble2 = abs(ydouble2(1:length(zdouble2)/2+1));
frqdouble2 = [0:length(zdouble2)-1]*Fs/length(zdouble2)/2;
% Plot spectrums of message signal and dsbsc
figure;
subplot(6,1,1); plot(t,x1);
```

MIST, Hyderabad – ECE Department  Page 41
Analog Communication Lab Manual, Prepared by Nakka Ravi Kumar Asst. Prof. & Roopalakshmi Asst. Prof MIST

```matlab
% FDM signal
m = ydouble1 + ydouble2;
zfdm = abs(zfdm(1:length(zfdm)/2+1));
frqfdm = [0:length(zfdm)-1]*Fs/length(zfdm)/2;

% Separating AM-DSB-SC-1 from FDM signal
[den1 num1] = butter(1, (Fc1-F1)/50, 'high');
M11 = filter(den1, num1, m);
[den2 num2] = butter(1, (Fc1+F1)/50, 'low');
M12 = filter(den2, num2, M11);
y1 = amdemod(M12, Fc1, Fs);

% Separating AM-DSB-SC-2 from FDM signal
[den3 num3] = butter(1, (Fc2-F2)/50, 'high');
M21 = filter(den3, num3, m);
[den4 num4] = butter(1, (Fc2+F2)/50, 'low');
M22 = filter(den4, num4, M21);
y2 = amdemod(M22, Fc2, Fs);

% Plot FDM and spectrum of FDM and demultiplexed and demodulated signals
figure;
subplot(4,1,1); plot(t,m);
title('FDM signal');
subplot(4,1,2); plot(frqfdm, zfdm);
title('spectrum of FDM signal');
subplot(4,1,3); plot(t,y1);
title('Demodulated signal 1');
subplot(4,1,4); plot(t,ba22);
title('Demodulated signal 2');
```

MIST, Hyderabad – ECE Department
results
4.8.4 Hardware

- Apparatus
  1. PHYSITECH’S Frequency Division Multiplexing and Demultiplexing trainer kit.
  2. C.R.O.
  3. Patch Cords.
  4. BNC Cables.

- Circuit diagram

- Procedure
  1. Observe the Message signals (AF1 & AF2) and note down the waveforms.
  2. Connect the AF1 signals to i/p of FM(X)
  3. Connect the AF2 signals to i/p of FM(Y)
  4. Observe the FM (X) and FM(Y) O/P’s and note down the FM waveforms
  5. Connect the FM(X) Signal to i/p(X) of Mux
  6. Connect the FM(Y) Signal to i/p(Y) of Mux
  7. Connect clock o/p to clock i/p of Mux
  8. Observe the FDM O/P and note down the wave form
  9. Connect FDM O/P to i/p of Demux and clock O/P to clock i/p of Demux
 10. Observe the O/P(X) & O/P(Y) and note down the wave forms.
 11. Connect O/P(X) & O/P(Y) to Corresponding i/p of FM demodulator CKT’s
 12. Adjust PLL’s to get demodulating signals
 13. Observe original demodulating Message Signals and note down
 14. Compare the demodulated Signal with original message signals
- Expected waveform

Modulating signal with 1 KHz frequency

Modulating signal with 2 KHz frequency
FM signal with 10 KHz Carrier (FM1)

FM signal with 20 KHz Carrier (FM2)
Demodulated FM(X) with 10KHz Carrier signal

Demodulated FM(Y) with 20KHz Carrier signal

1KHz Demodulated message signal

2KHz Demodulated message signal
EXPERIMENT – 9

4.9 Verification of sampling theorem

4.9.1 Aim: To Verify Sampling theorem.

4.9.2 Theory

Sampling theorem states that if the sampling rate in any pulse modulation system exceeds twice the maximum signal frequency the original signal can be reconstructed in the receiver with minimum distortion.

Let \( m(t) \) be a signal whose highest frequency component is \( f_m \). Let the value of \( m(t) \) be obtained at regular intervals separated by time \( T \) far less than \((1/2 f_m)\) The sampling is thus periodically done at each \( T_s \) seconds. Now the samples \( m(nT_s) \) where \( n \) is an integer which determines the signals uniquely. The signal can be reconstructed from these samples without distortion.

Time \( T_s \) is called the SAMPLING TIME.
The minimum sampling rate is called NYQUIST RATE.
The validity of sampling theorem requires rapid sampling rate such that at least two samples are obtained during the course of the interval corresponding to the highest frequency of the signal under analysis.

Let us consider an example of a pulse modulated signal, containing speech information, as is used in telephony. Over standard telephone channels the frequency range of A.F. is from 300 Hz to 3400 Hz. For this application the sampling rate taken is 8000 samples per second. This is an Inter-national standard. We can observe that the pulse rate is more than twice the highest audio frequency used in this system. Hence the sampling theorem is satisfied and the resulting signal is free from sampling error.

4.9.3 MATLAB program and description

Program:

```matlab
t=0:0.001:0.1;
t1=zeros(1,length(t));
f=input('enter the baseband signal frequency')
x=sin(2*pi*f*t);
n=input('enter the integer which decides the sampling frequency')
for i=1:length(t)
    if n*i<=length(t)
        t1(n*i)=1;
    end
end
s1=x.*t1;
[den,num]=butter(1,2*pi*f/1000);
s11=filter(den,num,s1);
subplot(2,1,1)
stem(t,s1);
title('n=8, Sampling rate less than Nyquists Rate')
subplot(2,1,2)
plot(t,s11)
title('Reconstructed signal')
```

Results:

enter the baseband signal frequency
\[
f = 10 \\
\text{enter the integer which decides the sampling frequency} \\
n = 8 \\
\]

\[
\text{n=8, Sampling rate less than Nyquist's Rate} \\
\]

\[
\text{Reconstructed signal} \\
\]

\[
f = 10 \\
\text{enter the baseband signal frequency} \\
n = 2 \\
\]

\[
\text{n=2, Sampling rate more than Nyquist's Rate} \\
\]

\[
\text{Reconstructed signal} \\
\]
4.9.4 **Hardware**

**Apparatus**
- PHYSITECH’s Sampling Theorem Trainer Kit
- Function Generator
- CRO
- Connecting wires.
- BNC Probes.

**Circuit diagram**

![Circuit Diagram](image)
Procedure

1. Connections are made as per the Circuit diagram.
2. Apply the input signal with a frequency of 500Hz (VP-P) using a function generator.
3. Sampling clock frequency which is variable of 3KHz to 50KHz should be connected across the terminals which is indicated.
4. Now observe the sampling output of the circuit at the o/p.
5. By using the capacitors provided on the trainer, reconstruct the signal and verify it with the given input.
6. Reconstructed signal voltage will be depends on capacitor value.
7. Vary the sampling frequency and study the change in reconstructed signal.
8. If the sampling clock frequency is below 20KHz you will observe the distorted demodulated output.

-Expected waveform
EXPERIMENT - 10

4.10 Pulse Amplitude modulation and demodulation

4.10.1 AIM: To study the Pulse Amplitude modulation and de-modulation and their waveforms.

4.10.2 Theory

Pulse Amplitude Modulation (PAM) is the simplest and most basic form of analog pulse modulation. In PAM, the amplitudes of regularly spaced pulses are varied in proportional to the corresponding sample values of a continuous message signal; the pulses can be of a rectangular form or some other appropriate shape.

PAM is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cable, or else are used to modulate a carrier. The two types of PAM are shown in fig. above. The two types are Double-polarity PAM, and single-polarity PAM. The largest pulse represents the greatest positive signal amplitude sampled, while the smallest pulse represents the largest negative sample. The time duration of each pulse may be quite short, and the time interval between pulses may be relatively long. In single-polarity PAM, in which a fixed dc level is added to the signal, to ensure that the pulses are always positive. The ability to use constant-amplitude pulses is a major advantage of pulse modulation and since PAM does not utilize constant amplitude pulses, it is infrequently used. When it is used, the pulses frequency modulate the carrier.

If a radio frequency is pulse-amplitude modulated instead of simply being amplitude modulated, much less power is required for the transmission of information because the transmitter is actually switched off between pulses. This is one advantage of pulse modulation. It is very easy to generate and demodulate PAM. In a generator the signal to be converted to PAM is fed to one input of an AND gate. Pulses at the sampling frequency are applied to the
other input of the AND gate to open it during the wanted time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant. The pulses are then passed through a pulse-shaping network, which gives them flat tops. Frequency modulation is then employed, so that the system becomes PAM-FM.

In the receiver, the pulses are first recovered with a standard FM de-modulator. They are then fed to an ordinary diode detector, which is followed by a low-pass filter. If the cutoff frequency of this filter is high enough to pass the highest signal frequency, but low enough to remove the Sampling frequency ripple, an undistorted replica of the original signal is reproduced.

### 4.10.3 MATLAB program and description

**Program:**

```matlab
% pulse amplitude modulation
close all
clear all
clc
t = 0 : 1/1e3 : 3;         % 1 kHz sample freq for 1 sec
d = 0 : 1/5 : 3;
x = sin(2*pi/4*2*t);        %message signal
den, num]=butter(1,2*pi*0.5/1000);
s11=filter(den,num,z);
s12=filter(den,num,s11);
```

```matlab
subplot(4,1,1)
plot(x);
title('message');
xlabel('time');
ylabel('amplitude');
y = pulstran(t,d,'rectpuls',0.1);    %generation of pulse input
subplot(4,1,2)
plot(y);
title('Pulse Input ');
xlabel('time');
ylabel('amplitude');
z=x.*y;                  % PAM output
subplot(4,1,3)
plot(z);
title('PAM modulation ');
xlabel('time');
ylabel('amplitude');
[s11]=butter(1,2*pi*0.5/1000);
s11=filter(den,num,z);
s12=filter(den,num,s11);
subplot(4,1,4)
plot(t,s12)
axis([0 3.5 -1 1]);
title('filtered signal')
```
4.10.4 Hardware

- Apparatus
1. Physitech’s Pulse Amplitude Modulation trainer (PHY-60)
2. Signal generator
3. CRO
4. BNC Probes, Connecting wire

- Circuit diagram

PAM MODULATOR
**PAM DEMODULATOR**

- **Procedure**

1. Switch on the Physitech’s pulse amplitude modulation and demodulation trainer
2. In clock generator section, connect pin-6 of 555 IC to the 33 pf capacitor terminal
3. Check the clock generator RF output signal.
4. Connect RF output of clock generator to the RF input of modulator section
5. Connect a 1 KHz, 2 V p-p of sine wave from a function generator to the AF input of modulator section.
6. Short the 10 F terminal and 10K terminal of modulator
7. Connect 10k Terminal to pin-1 of IC 4016
8. Connect the CRO to modulated output of modulator section.
9. Adjust the 1k potentiometer to vary the amplitude of the modulated signal
10. Adjust the AF signal frequency from 1KHz – 10KHz to get stable output waveform while increasing the AF signal frequency decreases the output signal pulses.
11. During demodulation, connect the modulated output to the PAM input of demodulator section.
12. Connect channel 1 of CRO to modulating signal and channel -2 to demodulated output. Observe the two waveforms that they are 180 degrees out of phase., since the transistor detector operates in CE configuration.
Expected waveform
EXPERIMENT – 11

4.11 Pulse width modulation and demodulation

4.11.1 AIM: To study the Pulse Width Modulation (PWM) and Demodulation process and record the corresponding waveforms

4.11.2 Theory

The Pulse-width modulation of PTM is also called as Pulse-duration modulation (PDM), or pulse length modulation (PLM). In this modulation, the pulses have a constant amplitude and a variable time duration. The time duration (or width) of each pulse is proportional to the instantaneous amplitude of the modulating signal. In this system, as shown in fig. below, we have a fixed amplitude and starting time of watch pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.

In this case, the narrowest pulse represents the most negative sample of the original signal and the widest pulse represents the most positive sample.

When PDM is applied to radio transmission, the carrier frequency has constant amplitude, and the transmitter on time is carefully controlled in some circumstances, PDM can be more accurate than PAM. One example of this is in magnetic tape recording, where pulse widths can be recorded and reproduced with less error than pulse amplitudes.

PWM or PPM are not used in telephony. To use PWM or PPM in such an application, we have to ensure that full scale modulation will not cause a pulse from one message signal to enter a time slot belonging to another message signal. This restriction results in a wasteful use of time space in telephone systems that are characterized by high peak factors.

4.11.3 MATLAB program and description

Program:

```matlab
% pulse width modulation & demodulation
close all
clear all
clc
fc=1000;
fs=10000;
f1=200;
t=0:1/fs:(2/f1)-(1/fs));
x1=0.4*cos(2*pi*f1*t)+0.5;
% modulation
y1=modulate(x1,fc,fs,'pwm');
subplot(411);
plot(x1);
axis([0 100 0 1]);
title('modulating signal,f1=200,fs=10000')
subplot(412);
plot(y1);
axis([0 1000 -0.2 1.2]);
title('PWM')

% demodulation
x1_recov=demod(y1,fc,fs,'pwm');
[den, num]=butter(1,2*pi*f1/fs);
```

```
s11=filter(den,num,x1_recov);
s12=filter(den,num,s11);
    subplot(413);
    plot(x1_recov);
    title('time domain recovered, single tone,f1=200')
    axis([0 100 0 1]);
s12=filter(den,num,s11);
    subplot(414);
    plot(s12);
    title('filtered output')
    axis([0 100 0 1]);

result:

4.11.4 Hardware

- Apparatus

1. Physitech’s Pulse width Modulation and demodulation trainer
2. CRO
3. BNC probes and connecting wires.
- Circuit diagram

MODULATOR CIRCUIT

Demodulation Circuit
- **Procedure**
  1. Switch on Physitech’s pulse width modulation and demodulation trainer
  2. Connect the Clk O/P to the Clk I/P terminal PWM modulation.
  3. Connect the AF O/P to AF I/P terminal of PWM modulator
  4. Observe the PWM O/P at pin-3 of 555 IC on CRO
  5. By varying frequency and amplitude of the modulating signal, observe the corresponding change in the width of the output pulses.
  6. During demodulation, connect the PWM O/P of PWM modulation to the PWM I/P of PWM demodulation.
  7. Observe the demodulated output at AF O/P of PWM demodulation on CRO.

- **Expected waveform**

![Expected waveform diagram](image)

**Fig (2) PULSE WIDTH MODULATION**
(a) Signal
(b) Unmodulated pulses
(c) PWM
EXPERIMENT - 12

4.12 Pulse position modulation and demodulation

4.12.1 AIM: To study the Pulse Position Modulation (PPM) and demodulation process and record corresponding waveforms.

4.12.2 Theory

Pulse position modulation (PPM) is more efficient than PAM or PDM for radio transmission. In PPM all pulses have the same constant amplitude and narrow pulse width. The position in time of the pulses is made to vary in proportion to the amplitude of the modulating signal.

The simplest modulation process for pulse position modulation is a PDM system with the addition of a monostable multivibrator. The monostable is arranged so that it is triggered by the trailing edges of the PDM pulses. Thus, the monostable output is a series of constant-width, constant amplitude pulses which vary in position according to the original signal amplitude.

4.12.3 MATLAB program and description

Program:

```matlab
% pulse position modulation
close all
clear all
clc
fc=100;
fs=1000;
f1=80;
t=0:1/fs:((2/f1)-(1/fs));
x1=0.4*cos(2*pi*f1*t)+0.5;
%modulation
y1=modulate(x1,fc,fs,'ppm');
subplot(311);
plot(x1);
axis([0 15 0 1]);
title('modulating signal,f1=80,fs=1000')
subplot(312);
plot(y1);
axis([0 250 -0.2 1.2]);
title('PPM')
%demodulation
x1_recov=demod(y1,fc,fs,'ppm');
subplot(313);
plot(x1_recov);
title('time domain recovered, single tone,f1=80')
axis([0 15 0 1]);
```

MATLAB program and description.
4.12.4 Hardware

- **Apparatus**
  1. PHYSITECH’s Pulse position modulation and demodulation trainer.
  2. CRO
  3. BNC probes and Connecting Wires
- Circuit diagram

**PWM MODULATOR**

**PWM - PPM (MONOSTABLE)**

**PPM - PWM (JK-FLIP-FLOP)**
PWM DEMODULATOR

- **Procedure**
  1. Switch on PHYSITECH’s PPM modulator and demodulator trainer.
  2. Connect the Clk O/P to the Pin 2 of 555 IC.
  3. Connect the AF O/P to the pin 5 of 555 IC.
  4. Observe the PPM O/P at pin 3 of second IC 555 on CRO.
  5. Connect the PPM O/P to the PPM I/P of PPM demodulation.
  6. Observe the demodulated O/P on CRO

- **Expected waveform**
EXPERIMENT-13

4.13 Frequency synthesizer

4.13.1 AIM: To study the operation of frequency synthesizer using PLL

4.13.2 Theory

Synthesizer is an equipment capable of generating a very large number of extremely stable frequencies with in same range of design ,while employing only one single stable source .The required frequency range in most synthesizers now a days is obtained from a variable voltage controller oscillator(vco),whose output is corrected by comparison with that of a reference source. This in built source is virtually a direct synthesizer.

There are two methods by which frequency multiplication can be achieved by using LM565 IC

1.locking to the harmonic of the input signal.
2.inclusion of a digital frequency divider or counter in loop between the VCO and phase comparator.

The first method is simplest, and can be achieved by setting the free running frequency of the VCO to a multiple of the input frequency. A limitation of this method is that the lock range decreases as successively higher and weaker harmonics are used for locking .If the input frequency is to be constant with little tracking required the loop can generally be locked to any one of the first five harmonics. For higher orders of multiplication, a large lock range is desired for which the second scheme is more desirable.

4.13.3 MATLAB program and description

```matlab
% program for frequency synthesizer
close all;
clear all;
clc
fs = 10000;
t = 0:1/fs:1.5; f=50;
x1 = square(2*pi*f*t);
subplot(3,1,1)
plot(t,x1);
axis([0 0.2 -1.2 1.2])
xlabel('Time (sec)');
ylabel('Amplitude');
title('Square wave input with freq=50HZ');
t = 0:1/fs:1.5;
x2 = square(2*pi*2*f*t);
subplot(3,1,2)
plot(t,x2);
axis([0 0.2 -1.2 1.2])
xlabel('Time (sec)');
ylabel('Amplitude');
title('frequency multiplication by a factor of 2');
x3 = square(2*pi*f/2*t);
subplot(3,1,3)
plot(t,x3);
axis([0 0.2 -1.2 1.2])
```

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xlabel('Time (sec)');
ylabel('Amplitude');
title('frequency division by a factor of 2');

4.13.4 Hardware

- **Apparatus**
  1. Physitech frequency synthesizer trainer (PHY-704).
  2. Oscilloscope (OSC-5030, 30MHz) or Equivalent.
  3. BNC Probes (1:1/10:1, switch selectable).
  4. Connecting wires

- **Circuit diagram**
**CIRCUIT DIAGRAM:**

![Circuit Diagram](image)

- **Procedure**
  1. Pin 2 is connected to the 0.1 micro farad capacitor and the other end of the capacitor is connected to the 10k resistor. To the other end of this resistor input 1khz signal is given from function generator
  2. For the same pin 2 connect one end of the 680ohms resistor and the other end to GND
  3. Connect one end of 680ohms resistor to pin3 and the other end to ground
  4. The output of vco (i.e 4th & 5th pin shorted)
  5. 7th pin is connected to 0.1 micro farad capacitor and the other end to +5v.
  6. 8th pin of the IC is given to 10k variable resistor and the other end to +5v.
  7. 9th pin of the IC is connected to capacitor C1 which is variable and the other end is connected to -5v.
  8. Connect the input signal (1 KHz) i.e from function generator is given to the pin 2 at the input of 10 Kohm that signal is observed at channel 1 of CRO
  9. Connect VCO output to the second channel of the CRO
  10. By varying the frequency (1KHz to 7KHz ) in different steps observe that at one frequency the wave form will be phase locked
  11. By varying the frequency knob of function generator in anti clock wise direction we get capture range
  12. Change RC component to shift VCO center frequency and see how lock range of the input varies
  13. Now compare the theoretical value and practical values using the given formula

\[
F_0 = \text{in Hz} \ 4R1C1
\]

Where
\[
F_0 = \text{free running frequency.}
\]
\[
R_1 = \text{external resistor.}
\]
\[
C_1 = \text{external capacitor}
\]
\[ F_c = \pm \left\{ \frac{f}{2} \right\} \frac{3.6 \times 10^3 C}{2p} \frac{1}{2} \]

Where

- \( F_c \) = Capture Range.
- \( C_2 \) = the filter capacitor in farad.

\[ F_L = \pm \frac{8f_0}{V_c} \text{ Hz} \]

Where, \( F_L \) = Lock Range

**Expected waveforms:**

\[ f_{IN} \]

\[ f_{OUT} \]

\[ V_c \]

\[ t \]

**TABULAR COLUMN:**

<table>
<thead>
<tr>
<th>Value of C'</th>
<th>( f_{IN} ) KHz</th>
<th>N</th>
<th>( f_{OUT} = N f_{IN} ) KHz</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
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</tbody>
</table>

MIST, Hyderabad – ECE Department  Page 68
EXPERIMENT - 14

4.14 AGC characteristics

4.14.1 AIM : To Study the AGC characteristics of a Radio receiver

4.14.2 Theory

The main purpose of the receiver is to recreate the original message signal from the degraded version of the transmitted signal after propagation through the free space.

The Super Heterodyne Receiver :

The Basic receiver is shown in fig. (1) The first stage is a tuned RF amplifier, using two variable tuned circuits that track each other and the local oscillator. The two tuned RF circuits form a band pass filter to pass the desired RF signal frequency while blocking others. This stage acts to boost the weak signal level from the antenna above the noise level to provide same signal selectivity and to prevent other channel signal to pass through.

The output signal from the amplifier is fed to one input of the mixer circuit and the local oscillator signal to the other. The local oscillator is variable tuned so as to track the incoming signal frequencies.

The mixer output (the difference frequency for down-conversion) is fed to multistage tuned. IF amplifiers, which are fixed-tune and provided with sufficient selectivity to reject adjacent channel signals. The output from the IF amplifier chain is fed to the detector circuit. Where the audio signal is extracted from the IF signal or demodulated. The detector also provides signals for automatic gain control (AGC). The AGC signal is used as a bias signal to reduce the gain of the RF and the IF amplifiers to prevent detector overload. AGC is a system means of which the overall gain of a radio receiver is varied automatically with the changing strength of the received signal to to keep the output substantially constant.

The audio signal from the detector is passed through a low pass filter to remove unwanted high frequency components and then through a volume control to an audio amplifier. The audio amplifier is usually one low-level audio stage followed by a power amplifier and a speaker.

The gain required in the RF and IF amplifier chain of the receiver depends on the required input and output. The input is the minimum variable signal level to be presented at the antenna terminals. The output is the minimum signal level at the input of the detector required to make the detector perform satisfactorily . In this trainer we used Radio receiver IC 1610S which is having in-built RF amplifier , Local oscillator, a mixer, IF amplifiers and AGC detector, audio amplifier to drive a speaker.
4.14.3 **MATLAB program and description**

**Program:**

```matlab
% program for AGC
close all;
clear all;
clc;
A=input('enter the carrier signal peak')       % 1
B=input('enter the baseband signal peak')       % 0.25
f1=input('enter the baseband signal frequency')  % 10
f2=input('enter the carrier signal frequency')  % 100
fs=input('enter the sampling frequency')        % 1000
t=0:0.001:1;
M=cos(2*pi*f1*t);
N=cos(2*pi*f2*t);
m=B/A;
O=A*(1+m*M).*N;
O1=O;
for i=1:length(t)
    if O1(i)<=0
        O1(i)=0;
    end
end
[den,num]=butter(2,2*pi*f1/fs);
M1=filter(den,num,O1);
M12=filter(den,num,M1);
M13=filter(den,num,M12);
Av=mean(M13);
disp('AGC Bias is')
disp(Av)
M131=M13/Av;
disp('Max of demodulated signal before Bias Correction is')
disp(max(M13))
disp('Max of demodulated signal after Bias Correction is')
disp(max(M131))
subplot(3,1,1)
plot(t,O);
title('Modulated Carrier')
subplot(3,1,2)
plot(t,M13)
axis([0 1 0 4])
title('demodulated Signal')
subplot(3,1,3)
plot(t,M131)
title('Demodulated signal after Bias correction')
axis([0 1 0 4])
```

```
result

[Graphs showing Modulated Carrier, demodulated Signal, and demodulated signal after Bias correction]

Hardware

Apparatus

1. Physitech's AGC characteristics trainer.
2. 20MHz Dual trace Oscilloscope
3. Patch Chords
- **Circuit diagram**

![Circuit Diagram](image)

- **Procedure**
1. Select carrier frequency of 1000 KHz. AF frequency 1 KHz and apply AM signal to the input of receiver. Set amplitude to around 1mV.
2. Connect CRO at the output of the Audio amplifier.
3. Tune the mixer- Local oscillator for maximum AF signal output at detector output and measure the audio signal.
4. Increase the RF level in appropriate steps and note down corresponding output AF signal amplitude.
5. Plot the AF output vs. RF input on graph which will be as shown in the figure.
- *Expected waveform*

![Graph showing AGC characteristics curve]

Receiver voltage

Input signal (AM) in mV

Fig. AGC characteristics curve
EXPERIMENT – 15

4.15 PLL

4.15.1 AIM: To study phase lock loop and its capture range, lock range and free running VCO Frequency.

4.15.2 Theory

PLL has emerged as one of the fundamental building block in electronic technology. It is used for the frequency multiplication, FM stereo detector, FM demodulator, frequency shift keying decoders, local oscillator in TV and FM tuner.

The block diagram of PLL is shown below.

The PLL consists of Phase detector, a LPF and a voltage controlled oscillator (VCO) connected together in the form a feedback system. The VCO is a sinusoidal generator whose frequency is determined by a voltage applied to it from an external source. In effect, any frequency modulator may serve as a VCO.

The phase detector or comparator compares the input frequency, fin, with feedback frequency, fout. The output of the phase detector is proportional to the phase difference between fin and fout. The output voltage of the phase detector is a DC voltage and therefore is often referred to as error voltage. The output of the phase detector is then applied to the LPF, which removes the high frequency noise and produces a DC level. The DC level, intern is the input to the VCO.

The output frequency of the VCO is directly proportional to the input DC level. The VCO frequency is compared with the input frequencies and adjusted until it is equal to the input frequency. In short, PLL keeps its output frequency constant at the input frequency.
4.15.3 MATLAB program and description

```matlab
close all;
clear all;
reg1 =0;
reg2 =0;
reg3 = 0;
eta =sqrt(2)/2;
theta =2*pi*1/100;
Kp = [(4*eta*theta)/(1+2*eta*theta+theta^2)];
Ki = [(4*theta^2)/(1+2*eta*theta+theta^2)];
d_phi_1 = 1/20;
n_data = 100;

for nn =1:n_data
    phi1= reg1 +d_phi_1;
    phi1_reg(nn) = phi1;
    s1 =exp(j*2*pi*reg1);
    s2 =exp(j*2*pi*reg2);
    s1_reg(nn) =s1;
    s2_reg(nn) =s2;
    t =s1*conj(s2);
    phi_error =atan(imag(t)/real(t))/(2*pi);
    phi_error_reg(nn) = phi_error;
    sum1 =Kp*phi_error + phi_error*Ki+reg3;
    reg1_reg(nn) =reg1;
    reg2_reg(nn) = reg2;
    reg1 =phi1;
    reg2=reg2+sum1;
    reg3 =reg3+phi_error*Ki;
    phi2_reg(nn) =reg2;
end

figure(1)
plot(phi1_reg);
hold on
plot(phi2_reg,'r');
hold off;
grid on;
title('phase plot');
xlabel('Samples');
ylabel('Phase');
figure(2)
plot(phi_error_reg);
title('phase Error of phase detector');
grid on;
xlabel('samples(n)');
ylabel('Phase error(degrees)');
figure(3)
plot(real(s1_reg));
hold on;
```

plot(real(s2_reg), 'r');
hold off;
grid on;
title('Input signal & Output signal of VCO');
xlabel('Samples');
ylabel('Amplitude');
axis([0 n_data -1.1 1.1]);
Hardware

- Apparatus
  1. PHYSITECH’s Phase Lock Loop Using LM 565 trainer
  2. Function generator
  3. CRO
  4. Connecting Wires

- Circuit diagram
- **Procedure**

Free running frequency:
1. Switch on the trainer and measure the output of the regulated power supplies i.e., +12V and ±5V
2. Observe the output of the square wave generator using oscilloscope and measure the frequency range. The frequency range should be around 1KHz to 10KHz.
3. Calculate the free running frequency range of the circuit for different values of timing capacitor and \( R_t \).
4. Connect 0.1\( \mu \)F capacitor (\( C_C \)) to the circuit and open the loop by removing short between pin 4 and 5. Measure the minimum and maximum free running frequencies obtainable at the output of the PLL (Pin4) by varying the pot. Compare your results with your calculation from step 3 (theoretical value). Simultaneously you can observe the output signal using CRO.

<table>
<thead>
<tr>
<th>( R_t ) value (pot resistance in Ohms)</th>
<th>Theoretical value (frequency in KHz)</th>
<th>Practical value (frequency in KHz)</th>
</tr>
</thead>
<tbody>
<tr>
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</tbody>
</table>

Lock range:
5. Calculate the lock range of the circuit for a 5KHz free running frequency and record in table 1.2.
6. Connect pins 4,5 with the help of springs and adjust potentiometer to get a free running frequency of 5KHz. Connect square wave generator output to the input of PLL circuit. Provide a 5KHz square signal of 1 V<sub>pp</sub> approximately (make this input frequency as close to the V<sub>cc</sub> frequency as possible).
7. Observe the input & Output of the PLL.
8. Observe the input and output frequencies while slowly increasing the frequency of the square wave at the input. For some range output and input are equal (This is known as lock Range and PLL is said to be in lock with the input signal). Record the frequency at which the PLL breaks lock. (Output frequency of the PLL will be around V<sub>CO</sub> frequency and in oscilloscope you will see a jittery waveform when it breaks lock instead of clean square wave). This frequency is called as upper end of the lock range and records this as \( F_2 \).
9. Beginning at 5KHz, slowly decrease the frequency of the input and determine the frequency at which the PLL breaks lock on the low end record it as \( F_1 \).
10. Find the lock range from \( F_2 - F_1 \) and compare with the theoretical values from step 5.
Capture range:
11. Calculate the capture range of the circuit for a 5KHz free-running frequency (consider filter capacitor (C_C) is 0.1µF).
12. With the oscilloscope and counter still on pin 4, slowly increase the input frequency from minimum (say 1KHz), Record frequency (as F_3) at which the input and output frequencies of the PLL are equal, this is known as lower end of the capture range.
13. Now keep input frequency at maximum possible (say 10KHz) and slowly reduce and record the frequency (as F_4) at which the input and output frequencies of PLL are equal. This is known as upper end of the capture range.
14. Find capture range from F_4 - F_3 and compare it with the theoretical value (from step 11)
15. Repeat the steps from 11 to 14 with C_C value 0.2µF

<table>
<thead>
<tr>
<th>Filter Capacitance(Cc)</th>
<th>Theoretical value (frequency in KHz)</th>
<th>Practical value (frequency in KHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0.1µF</td>
<td></td>
<td></td>
</tr>
<tr>
<td>0.2µF</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>