ANALOG AND DIGITAL COMMUNINCATIONS LABORATORY MANUAL (R22A0484)

B.TECH (II YEAR – II SEM) (2023-24)

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MALLA REDDY COLLEGE OF ENGINEERING & TECHNOLOGY

(Autonomous Institution – UGC, Govt. of India)

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VISION

To evolve into a center of excellence in Engineering Technology through creative and innovative practices in teaching-learning, promoting academic achievement & research excellence to produce internationally accepted competitive and world class professionals.

MISSION

To provide high quality academic programmes, training activities, research facilities and opportunities supported bv continuous industrv institute interaction aimed at entrepreneurship, employability, leadership and research aptitude among students.

QUALITY POLICY

- Impart up-to-date knowledge to the students in Electronics & Communication area to make them quality engineers.
- Make the students experience the applications on quality equipment and tools.
- Provide systems, resources and training opportunities to achieve continuous improvement.
- Maintain global standards in education, training and services.

PROGRAMME EDUCATIONAL OBJECTIVES

PEO1: PROFESSIONALISM & CITIZENSHIP

To create and sustain a community of learning in which students acquire knowledge and learn to apply it professionally with due consideration for ethical, ecological and economic issues.

PEO2: TECHNICAL ACCOMPLISHMENTS

To provide knowledge based services to satisfy the needs of society and the industry by providing hands on experience in various technologies in core field.

PEO3: INVENTION, INNOVATION AND CREATIVITY

To make the students to design, experiment, analyze, interpret in the core field with the help of other multi disciplinary concepts wherever applicable.

PEO4: PROFESSIONAL DEVELOPMENT

To educate the students to disseminate research findings with good soft skills and become a successful entrepreneur.

PEO5: HUMAN RESOURCE DEVELOPMENT

To graduate the students in building national capabilities in technology, education and research.

CODE OF CONDUCT FOR THE LABORATORIES

- 1. All students must observe the Dress Code while in the laboratory.
- 2. Sandals or open-toed shoes are NOT allowed.
- 3. Foods, drinks and smoking are NOT allowed.
- 4. All bags must be left at the indicated place.
- 5. The lab timetable must be strictly followed.
- 6. Be PUNCTUAL for your laboratory session.
- 7. Program must be executed within the given time.
- 8. Noise must be kept to a minimum.
- 9. Workspace must be kept clean and tidy at all time.
- 10. Handle the systems and interfacing kits with care.
- 11. All students are liable for any damage to the accessories due to theirown negligence.
- 12. All interfacing kits connecting cables must be RETURNED if you taken from the lab supervisor.
- 13. Students are strictly PROHIBITED from taking out any items from the laboratory.
- 14. Students are NOT allowed to work alone in the laboratory without the Lab Supervisor
- 15. USB Ports have been disabled if you want to use USB drive consult lab supervisor.
- Report immediately to the Lab Supervisor if any malfunction of the accessories, is there.

Before leaving the lab

- Place the chairs properly.
- Turn off the system properly
- Turn off the monitor.
- Please check the laboratory notice board regularly for updates.

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ANALOG COMMUNICATION EXPERIMENTS

EXPERIMENT NO-1 DATE:

AMPLITUDE MODULATION & DEMODULATION

<u>AIM</u>: (i) To study the function of Amplitude Modulation & Demodulation (under modulation, perfect modulation & over modulation) and also to calculate the modulation index.

(ii) To verify the spectrum of AM signals using spectrum analyzer.

APPARATUS:

- 1. Amplitude Modulation & De modulation trainer kit.
- 2. C.R.O (20MHz)
- 3. Function generator (1MHz).
- 4. Connecting cords & probes.
- 5. PC with windows(95/98/XP/NT/2000)
- 6. MATLAB Software with communication toolbox

THEORY:

Modulation is defined as the process of changing the characteristics (Amplitude, Frequency or Phase) of the carrier signal (high frequency signal) in accordance with the intensity of the message signal (modulating signal).

Amplitude modulation is defined as a system of modulation in which the amplitude of the carrier is varied in accordance with amplitude of the message signal (modulating signal).

The message signal is given by the expression.

 $Em(t) = Em \cos Wmt$

Where Wm is ----- > Angular frequency

Em -----→ Amplitude

Carrier voltage Ec(t) = Ec cosWct

E(t)=Ec + KaEm cosWmt

KaEm cosWmt----- \rightarrow change in carrier amplitude

Ka ---- \rightarrow constant

The amplitude modulated voltage is given by

 $E=E(t) \cos Wct$

From above two equations

 $E = (Ec + KaEm \cos Wmt) \cos Wct.$

 $E = (1 + KaEm/Ec \cos Wmt) Ec \cos Wct$

 $E = Ec(1+Ma \cos Wmt)\cos Wct$

Where Ma ---- \rightarrow depth of modulation/ modulation index/modulation factor

Ma=KaEm/Ec

100* Ma gives the percentage of modulation.

BLOCK DIAGRAM:

Modulation



Demodulation



(i) **<u>PROGRAM</u>**:

% program for AM modulation and demodulation

close all

clear all

clc

fs=8000;

```
fm=20;
fc=500;
Am=1;
Ac=1;
t = [0:0.1*fs]/fs;
m=Am*cos(2*pi*fm*t);
c=Ac*cos(2*pi*fc*t);
ka=0.5;
u=ka*Am;
s1=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,1:3);
plot(t,m);
title('Modulating or Message signal(fm=20Hz)');
subplot(4,3,4:6);
plot(t,c);
title('Carrier signal(fc=500Hz)');
subplot(4,3,7);
plot(t,s1);
title('Under Modulated signal(ka.Am=0.5)');
Am=2;
ka=0.5;
u=ka*Am;
s2=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,8);
plot(t,s2);
title('Exact Modulated signal(ka.Am=1)');
Am=5;
ka=0.5;
u=ka*Am;
s3=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,9);
plot(t,s3);
```

```
title('Over Modulated signal(ka.Am=2.5)');
r1 = s1.*c;
[b a] = butter(1,0.01);
mr1= filter(b,a,r1);
subplot(4,3,10);
plot(t,mr1);
title('deModulated signal for(ka.Am=0.5)');
r2 = s2.*c;
[b a] = butter(1,0.01);
mr2= filter(b,a,r2);
subplot(4,3,11);
plot(t,mr2);
title('deModulated signal for(ka.Am=1)');
r3 = s3.*c;
[b a] = butter(1,0.01);
mr3= filter(b,a,r3);
subplot(4,3,12);
plot(t,mr3);
title('deModulated signal for(ka.Am=2.5)');
```

(ii) <u>PROGRAM:</u>

% program of spectrum analyzer and analysis of am 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,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');

PROCEDURE:-

- Connect the AC Adapter to the mains and the other side to the Experimental Trainer. Switch 'ON' the power.
- 2. Observe the carrier and modulating waveforms and note their frequencies. (Carrier frequency is around 100 KHz and amplitude is variable from 0 -8Vp-p, modulating signal is 1KHz).
- 3. Connect the carrier and modulating signals to the modulator circuit.
- 4. Observe the amplitude modulated wave.
- Connect Carrier I/P to ground and apply a 2V peak to peak AF Signal input to (modulating I/P) and adjust P1 in anti-clock wise position to get minimum A.C output.

- 6. Connect modulating I/P to ground and apply a 3V peak to peak carrier signal to carrier I/P and adjust P2 in clock wise direction to get minimum A.C ouyput..
- 7. Connect modulating input & carrier input to ground and adjust P3 for zero D.C output.
- 8. Make modulating i/p 2 Vpp and carrier i/p 3 Vpp peak to peak and adjust potentiometer P4 for maximum output.
- 9. Calculate maximum and minimum points on the modulated envelope on a CRO and calculate the depth of modulation.
- 10. Observe that by varying the modulating voltage, the depth of modulation varies.
- 11. During demodulation connect this AM output to the input of the demodulator.
- 12. By adjusting the RC time constant (i.e., cut off frequency) of the filter circuit we get minimum distorted output.
- Observe that this demodulated output is amplified has some phase delay because of RC components.
- 14. Also observe the effects by changing the carrier amplitudes.
- 15. In all cases, calculate the modulation index.

EXPECTED WAVEFORMS:-







Demodulated signal



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OBSERVATIONS:

Modulation

	Vc (V)	Vm (V)	Vmax (V)	Vmin (V)	m=(Vmax-Vmin)/ (Vmax+Vmin)	m= Vm/Vc
Under modulation						
Perfect modulation						
Over modulation						

Demodulation

Modulating signal Frequency	Demodulated output signal frequency

RESULT:

OUESTIONS

- 1. Define AM and draw its spectrum?
- 2. Draw the phase's representation of an amplitude modulated wave?
- 3. Give the significance of modulation index?
- 4. What are the different degrees of modulation?
- 5. What are the limitations of square law modulator?
- 6. Compare linear and nonlinear modulators?
- 7. Compare base modulation and emitter modulation?
- 8. Explain how AM wave is detected?
- 9. Define detection process?

10. What are the different types of distortions that occur in an envelop detector? How can they be eliminated?

- 11. What is the condition of for over modulation?
- 12. Define modulation & demodulation?
- 13. What are the different types of linear modulation techniques?
- 14. Explain the working of carrier wave generator.
- 15. Explain the working of modulator circuit.

EXPERMENT NO-2 DATE:

FREQUENCY MODULATION AND DEMODULATION

<u>AIM</u>: (i) To study the process of frequency modulation and demodulation and calculate the depth of modulation by varying the modulating voltage.

(ii) To verify the spectrum of FM signals using spectrum analyzer.

<u>APPARATUS :</u>

- 1. FM modulation and demodulation kit
- 2. Dual trace CRO.
- 3. CRO probes
- 4. Patch cards.
- 5. PC with windows(95/98/XP/NT/2000)
- 6. MATLAB Software with communication toolbox

THEORY:

The modulation system in which the modulator output is of constant amplitude, in which the signal information is super imposed on the carrier through variations of the carrier frequency.

The frequency modulation is a non-linear modulation process. Each spectral component of the base band signal gives rise to one or two spectral components in the modulated signal. These components are separated from the carrier by a frequency difference equal to the frequency of base band component. Most importantly the nature of the modulators is such that the spectral components which produce decently on the carrier frquency and the base band frequencies. The spetral components in the modulated wave form depend on the amplitude.

The modulation index for FM is defined as

Mf= max frequency deviation/ modulating frequency.

BLOCK DIAGRAM:

Modulation



Demodulation



(i) PROGRAM:-

% program for fm modulation and demodulation

close all

clear all

clc

%fm=35HZ,fc=500HZ,Am=1V,Ac=1V,B=10

fs=10000;

Ac=1;

Am=1;

fm=35;

fc=500;

B=10;

t=(0:.1*fs)/fs;

wc=2*pi*fc;

wm=2*pi*fm;

m_t=Am*cos(wm*t);

subplot(4,1,1);

plot(t,m_t);

title('Modulating or Message signal(fm=35Hz)');

c_t=Ac*cos(wc*t);

subplot(4,1,2);

plot(t,c_t);

title('Carrier signal(fm=500Hz)');

s_t=Ac*cos((wc*t)+B*sin(wm*t));

subplot(4,1,3);

plot(t,s_t);

title('Modulated signal'); d=demod(s_t,fc,fs,'fm'); subplot(4,1,4); plot(t,d);

title('demodulated signal');

(ii) PROGRAM:

% program of spectrum analyzer and analysis of fm signals

close all clear all clc Fs = 100;%sampling frq t = [0:2*Fs+1]'/Fs;% Carrier frequency Fc = 10:x = sin(2*pi*2*t);% message signal Ac=1;% spectrum of fm xfm=fmmod(x,Fc,Fs,10); zfm = fft(xfm);zfm = abs(zfm(1:length(zfm)/2+1));frqfm = [0:length(zfm)-1]*Fs/length(zfm)/2; figure; plot(frqfm,zfm);

title('Spectrum of fm signal');

PROCEDURE:

- 1. Switch on the experimental board.
- 2. Observe the FM modulator output without any modulator input which is the carrier signal and note down its frequency and amplitude.
- 3. Connect modulating signal to FM modulator input and observe modulating signal and FM output on two channels of the CRO simultaneously.
- 4. Adjust the amplitude of the modulating signal until we get less distorted FM output.
- 5. Apply the FM output to FM demodulator and adjust the potentiometer in demodulation until we get demodulated output.

OBSERVATIONS:

Modulation

Vm	F1	F2	Frequency deviation Fd (f1-f2)	Modulating index (f1-f2)/Fm	Band width= 2(Fd+Fm)

Demodulation

Modulating signal	Demodulating signal		
frequency	frequency		

EXPECTED WAVEFORMS:-



TIME(T)

 \mathbf{V}

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RESULT:

OUESTIONS

- 1. Define FM & PM.
- 2. What are the advantages of Angle modulation over amplitude modulation?
- 3. What is the relationship between PM and FM?
- 4. With a neat block diagram explain how PM is generated using FM.

EXPERIMENT NO-3 DATE:

DSB-SC MODULATOR & DETECTOR

<u>AIM</u>: To study the working of the Balanced Modulator and demodulator.

APPARATUS:

- 1. Balanced modulator trainer kit
- 2. C.R.O (20MHz)
- 3. Connecting cords and probes
- 4. Function generator (1MHz)
- 5. PC with windows (95/98/XP/NT/2000)
- 6. MATLAB Software with communication toolbox

THEORY:

Balanced modulator circuit is used to generate only the two side bands DSB-SC. The balanced modulation system is a system is a system of adding message to carrier wave frequency there by only the side bands are produced. It consists of two AM modulators arranged in a balanced configuration. The AM modulator is assumed to be identical. The carrier input to the two modulators is same.

If we eliminate or suppress the carrier then the system becomes suppressed carrier DSB-SC. In this we need reinsert the carrier is complicated and costly. Hence the suppressed carrier DSB system may be used in point to point communication system.

Generation of suppressed carrier amplitude modulated volt balanced modulator may be of the following types.

- 1. Using transistors or FET.
- 2. Using Diodes

Modulation





PROGRAM:

% program for dsbsc modulation and demodulation

close all

clear all

clc

```
t =0:0.000001:.001;
```

Vm=1;

Vc= 1;

fm = 2000;

fc= 50000;

```
m_t = Vm^*sin(2^*pi^*fm^*t);
```

```
subplot(4,1,1);
```

plot(t,m_t);

```
c_t = Vc^*sin(2^pi^*fc^*t);
```

```
subplot(4,1,2);
```

```
plot(t,c_t);
```

```
subplot(4,1,3);
```

```
s_t = m_t.*c_t;
```

hold on;

```
plot(t,s_t);
```

```
plot(t,m_t,'r:');
```

```
plot(t,-m_t,'r:');
```

```
hold off;
```

 $r = s_t.*c_t;$

[b a] = butter(1,0.01);

mr= filter(b,a,r);

subplot(4,1,4);

plot(t,mr);

PROCEDURE:-

- 1. Connect the circuit as per the given circuit diagram.
- 2. Switch on the power to the trainer kit.
- 3. Apply a 100KHz, 0.1 peak sinusoidal to the carrier input and a 5KHz, 0.1 peak sinusoidal to the modulation input.
- 4. Measure the output signal frequency and amplitude by connecting the output to CRO.
- 5. And note down the output signals.

EXPECTED WAVEFORMS:-





OBSERVATIONS:

Carrier Signal		Message signal		Modu	lated signal	Demodulated Signal		
				output		output		
Fc(Hz)	Vc(volts)	Fm(Hz)	Vm(v)	Fo(Hz) Vo(v)		F(Hz)	V(v)	

RESULT:

OUESTIONS

- 1. What are the two ways of generating DSB_SC?
- 2. What are the applications of balanced modulator?
- 3. What are the advantages of suppressing the carrier?
- 4. What are the advantages of balanced modulator?
- 5. What are the advantages of Ring modulator?
- 6. Write the expression for the output voltage of a balanced modulator?
- 7. Explain the working of balanced modulator and Ring Modulator using diodes.

EXPERIMENT.NO-4 DATE:

SSB-SC MODULATOR & DETECTOR

(PHASE SHIFT METHOD)

<u>AIM</u>:- To generate SSB using phase method and detection of SSB signal using Synchronous detector.

APPARATUS:-

1. SSB trainer kit

- 2. C.R.O (20MHz)
- 3. Patch cards
- 4. CRO probes

THEORY:

AM and DSBSC modulation are wasteful of band width because they both require a transmission bandwidth which is equal to twice the message bandwidth In SSB only one side band and the carrier is used. The other side band is suppressed at the transmitter, but no information is lost. Thus the communication channel needs to provide the same band width, when only one side band is transmitted. So the modulation system is referred to as SSB system.

The base band signal may not be recovered from a SSB signal by the Use of a diode modulator. The bae band signal can be recovered if the spectral component of the output i.e either the LSB or USB is multiplied by the carrier signal.

Consider the modulating signal

 $M(t)=A_m \cos W_{mt}$

$$C(t)=A_c \cos W_{ct}$$

 $M(t)c(t) = A_c A_m \cos W_{mt} \cos W_{ct}$

The above signal when passed through a filter, only one of the above component is obtained which lays the SSB signal.

BLOCK DIAGRAM: -

SSB MODULATION



SSB DEMODULATION/SYNCHRONOUS DETECTOR



PROGRAM:-

% program for ssb modulation and demodulation

close all

clear all

clc

fs=8000;

fm=20;

fc=50;

Am=1;

Ac=1;

```
t=[0:0.1*fs]/fs;
```

subplot(5,1,1);

```
m1=Am*cos(2*pi*fm*t);
```

plot(t,m1);

```
title('Message Signal');
```

```
m2=Am*sin(2*pi*fm*t);
```

```
subplot(5,1,2)
c1=Ac*cos(2*pi*fc*t);
plot(t,c1)
title('Carrier Signal');
c2=Ac*sin(2*pi*fc*t);
subplot(5,1,3)
% Susb=0.5* Am*cos(2*pi*fm*t).* Ac*cos(2*pi*fc*t) -- 0.5* Am*sin(2*pi*fm*t).*
Ac*sin(2*pi*fc*t);
Susb=0.5*m1.*c1-0.5*m2.*c2;
plot(t,Susb);
title('SSB-SC Signal with USB');
subplot(5,1,4);
Slsb=0.5*m1.*c1+0.5*m2.*c2;
plot(t,Slsb);
title('SSB-SC Signal with LSB');
r = Susb.*c1;
subplot(5,1,5);
[b a] = butter(1,0.0001);
mr= filter(b,a,r);
plot(t,mr);
title('demodulated output');
```

PROCEDURE:-

SSB MODULATION

1. Connect the Adaptor to the mains and the other side to the Experimental Trainer Switch 'ON' the power.

2. (a) Connect carrier $f_c 90^0$ to A_{in} of Balanced Modulator –A and adjust its amplitude to 0.1Vpp.

(b). Connect modulating signal $f_m 0^0$ 5Vpp to B_{in} of the Balanced Modulator-A.

3. Observe the DSB-A output on CRO.

4. Connect fc 0^0 at 0.1 Vpp at C_{in} of Balanced Modulator B. Connect fm 90^0 at 5 Vpp at D_{in} of Balanced Modulator B.

5. Connect the DSB-A output and DSB-B output to the summing amplifier. Observe the output (SSB output) on the spectrum analyzer. This gives single side band (upper) only while the lower side band is cancelled in the summing Amplifier.

SSB DEMODULATION

- 1. Connect the carrier fc 0^0 and SSB output to the synchronous detector.
- 2. Connect the demodulator output on the oscilloscope which is the recovered modulating signal.

OBSERVATIONS:

Carrier signal		Modulating signal		Balar modula	nced ntor-A	d Balance -A modulate		Ade Subtr Out	der/ actor tput	Syncl de	hronous tector
Fc	Vc	Fm	Vm	Vmax	Vmin	Vmax Vmin		Vmax	Vmin	Fd	Vd

EXPECTED WAVE FORMS: -






-

RESULT:

OUESTIONS

- 1. What is the advantage of SSB-SC over DSB-SC?
- 2. What are the different methods to generate SSB-SC signal?
- 3. Explain Phase Shift method for SSB generation.
- 4. Why SSB is not used for broadcasting?

SSB DETECTION

- 5. Give the circuit for synchronous detector?
- 6. What are the uses of synchronous or coherent detector?
- 7. Give the block diagram of synchronous detector?
- 8. Why the name synchronous detector?

EXPERIMENT NO-5 DATE:

FREQUENCY DIVISION MULTIPLEXING

& DE MULTIPLEXING

AIM: To study the frequency division multiplexing and De multiplexing Techniques.

APPARATUS/SOFTWARE REQUIRED:

- 1. FREQUENCY DIVISION MULTIPLEXING & DEMULTIPLEXING Trainer Kit.
- 2. C.R.O (30 MHz)
- 3. Patch chords.
- 4. PC with windows (95/98/XP/NT/2000)
- 5. MATLAB Software

BLOCK DIAGRAM:



PROGRAM:

% program for frequency division multiplexing and demultiplexing

close all

clear all

clc

Fs = 100; % sampling freq

$$t = [0:2*Fs+1]'/Fs;$$

x1 = sin(2*pi*2*t); % signal 1 signal

$$z1 = fft(x1);$$

z1=abs(z1);

```
x^2 = sin(2*pi*10*t); \% signal 2 signal
```

 $z^{2} = fft(x^{2});$

z2=abs(z2);

figure;

subplot(4,1,1); plot(x1);

title('signal 1');xlabel('time');ylabel('amplitude');

subplot(4,1,2); plot(x2);

title('signal 2');xlabel('time');ylabel('amplitude');

subplot(4,1,3); plot(z1);

title('Spectrum of signal 1');xlabel('freqency');ylabel('magnitude');

subplot(4,1,4); plot(z2);

title('Spectrum of signal 2');xlabel('freqency');ylabel('magnitude');

% freqency multiplexing

z=z1+z2;

figure;

plot(z);

```
title('frequency multiplexed signals');
```

figure;

% freqency demultiplexing

```
f1=[ones(10,1); zeros(182,1);ones(10,1)];% applying filter for signal 1
```

dz1=z.*f1;

```
d1 = ifft(dz1);
```

```
subplot(2,1,1)
```

```
plot(t*100,d1);
```

```
f2=[zeros(10,1); ones(182,1);zeros(10,1)];% applying filter for signal 2
```

```
dz2=z.*f2;
```

d2 = ifft(dz2);

title('recovered signal 1');xlabel('time');ylabel('amplitude');

subplot(2,1,2)

plot(t*100,d2);

title('recovered signal 2');xlabel('time');ylabel('amplitude');

PROCEDURE:

FDM Multiplexing:

- 1. Connect the circuit as shown in the figure.
- 2. Switch ON the power supply.

3. Set the amplitude of each modulating signal as 5Vp-p and frequency of each AF signal to

1kHz and 2kHz respectively.

4. Monitor the outputs at Tp1(signal-1), Tp2(signal-2), Tp10(RF-16kHz), Tp12(RF-

32KHz),Tpq(modulation-1),Tp11(Modulator-2),Tp17(BPF & adder)

5. Set output frequency of RF oscillator to 455 kHz and amplitude to10Vp-p.

6. Monitor the output at Tp18 the FDM DSB-SC wave will be observed.

FDM DeMultiplexing & LPF:

- 1. Connect the Tp18 to Tp22 and observe the output of main demodulator at Tp23.
- 2. Connect the main demodulator output to the BPF1 (28-32 kHz) and BPF1 (12-16 kHz).
- 3. Connect the output of BPF,s to the respective demodulator and then to LPF,s.
- 4. Monitor the demodulated signal1 and at TP32 and demodulated signal2 at TP39.

EXPECTED WAVEFORMS:



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RESULT:

EXPERIMENT NO-6 DATE:

PULSE AMPLITUDE MODULATION

<u>AIM:</u> 1.To study the Pulse amplitude modulation & demodulation Techniques.

2. To study the effect of amplitude and frequency variation of modulating signal on the output.

APPARATUS:-

- 1. Pulse amplitude modulation & demodulation Trainer Kit.
- 2. Dual trace CRO.
- 3. Patch chords.
- 4. PC with windows(95/98/XP/NT/2000)
- 5. MATLAB Software with communication toolbox

THEORY:-

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with syncing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse amplitude modulation is the simplest form of the pulse modulation. PAM is a pulse modulation system is 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 cables are used to modulated carrier.

The two types of PAM are i) Double polarity PAM, and ii) the single polarity PAM, in which a fixed dc level is added to the signal to ensure that the pulses are always positive. Instantaneous PAM sampling occurs if the pulses used in the modulator are infinitely short.

Natural PAM sampling occurs when finite-width pulses are used in the modulator, but the tops of the pulses are forced to follow the modulating waveform.

Flat-topped sampling is a system quite often used because of the ease of generating the modulated wave.

PAM signals are very rarely used for transmission purposes directly. The reason for this lies in the fact that the modulating information is contained in the amplitude factor of the pulses, which can be easily distorted during transmission by noise, crosstalk, other forms of distortion. They are used frequently as an intermediate step in other pulse-modulating methods, especially where time-division multiplexing is used.

Circuit description:-

Pulse and Modulation Signal Generator:-

A 4.096 MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comprising a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U10), to produce 50% duty cycle, 1 KHz square wave on pin no.1 of U10, and 2KHz square wave on pin no.15. the frequency is selectable by means of SW1. this goes to input of fourth order low pass filter U11(TL072) is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

The square wave which is generated by the oscillator is buffered by inverter 74HC04(U9), to produce 32KHz square wave at pin no.4 of the 74HC4040(U10). This pulse is given to the monostable multi to obtain the 16 KHz and 32 KHz square wave at the output which are selected by the frequency pot.

Modulation:-

The ICDG211 (U3) is used as a pulse amplitude modulation in this circuit. The modulation signal & pulse signals are given to TL074 (U2) & 7400(U1) IC's respectively. These outputs are fed to the inputs the D4211 (U3).

The sampled output is available at the pin no 2 of DG211 and it is buffered by using TL074 (U2) and then output is available at TP5.

Similarly the sample & hold output and the flat top output are available at pin no.15 &10 of DG211 respectively. These are buffered by TL074 (U2) and then output is available at TP6&TP7 respectively.

Demodulation:-

The demodulation section comprises of fourth order low pass filter and an AC amplifier. The TL074 (U5) is used as a low pass filter and AC amplifier. The output of the modulator is given as the input to the low pass filter.

The low pass filter output is obviously less and it is fed to the AC amplifier which comprises of a single op amp and whose output is amplified.

CIRCUIT DIAGRAM:



PROGRAM:-

% pulse amplitude modulation close all clear all clc t = 0 : 1/1e3 : 10;% 1 kHz sample freq for 1 sec d = 0 : 1/5 : 10; $x = 5 + \sin(2*pi/4*2*t);$ % message signal figure; subplot(3,1,1)plot(x); title('message'); xlabel('time');ylabel('amplitude'); y = pulstran(t,d,'rectpuls',0.1); % generation of pulse input subplot(3,1,2)plot(y);

title('Pulse Input '); xlabel('time');ylabel('amplitude'); z=x.*y; % PAM output subplot(3,1,3) plot(z); title('PAM modulation '); xlabel('time');ylabel('amplitude');

PROCEDURE:

Double Polarity:-

Modulation:-

- 1. Connect the circuit as shown in diagram 1.
 - a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position
 - b. 16KHz pulse output to pulse input TP1.(Keep the frequency in minimum position in pulse generator block).
- 2. Switch ON the power supply.
- 3. Monitor the outputs at TP5, TP6& TP7. And observe the outputs also by varying amplitude pot (Which is in modulation signal generator block).
- Now vary the frequency selection which position in modulating signal generator block to 2 KHz, amplitude pot to max position.
- 5. Observe the output at TP5, TP6& TP7 and observe the outputs also by varying amplitude pot (Which is in modulation signal generator block).
- 6. Repeat all the above steps for the pulse frequency 32KHz (By varying the frequency pot in the pulse generator block).
- 7. Switch OFF the power supply.

Single Polarity PAM:-

- 8. Connect the circuit as shown in diagram 2.
- a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position
 - b. 16KHz pulse output to pulse input TP1.

9. Switch ON the power supply.

10. Repeat above step 3 to 6 and observe the outputs.

11. Vary DC output pot until you get single polarity PAM at TP5, TP6, TP7.

12. Switch OFF the power supply.

Demodulation:-

1. Connect the circuit as shown in diagram 3.

a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position

b. 16KHz pulse output to pulse input TP1.

c. Sample output, sample and hold output and flat top outputs Respectively to the input of low pass filter(TP9) and LPF output (TP10) to AC amplifier input(TP11).

2. Observe the output of LPF and AC amplifier at TP10,TP12 respectively, corresponding to inputs from TP5,TP6 &TP7. The outputs will be the true replica of the input.

3. Now, set the switch position in modulating signal generator to 2KHz and observe the outputs at TP10&TP12 respectively, corresponding to inputs from TP5,TP6& TP7.

4. Vary the frequency of pulse to 32KHz (By varying the frequency pot(Put in max position) in pulse generator block) and repeat the above steps 2&3.

5. Switch OFF the power supply.

EXPECTED WAVEFORMS



RESULT:

OUESTIONS

1. TDM is possible for sampled signals. What kind of multiplexing can be used in continuous modulation systems?

2. What is the minimum rate at which a speech signal can be sampled for the purpose of PAM?

3. What is cross talk in the context of time division multiplexing?

4. Which is better, natural sampling or flat topped sampling and why?

5. Why a dc offset has been added to the modulating signal in this board? Was it essential for the working of the modulator? Explain?

6. If the emitter follower in the modulator section saturates for some level of input signal, then what effect it will have on the output?

7. Derive the mathematical expression for frequency spectrum of PAM signal.

- 8. Explain the modulation circuit operation?
- 9. Explain the demodulation circuit operation?

10. Is PAM & Demodulation is sensitive to Noise?

EXPERIMENT NO-7

DATE:

PULSE WIDTH MODULATION & DEMODULATION

<u>AIM:</u>

1. To study the Pulse Width Modulation (PWM) and Demodulation

Techniques.

2. To study the effect of Amplitude and Frequency of Modulating Signal on PWM output.

APPARATUS:

- 1. PWM trainer kit
- 2. C.R.O(30MHz)
- 3. Patch Chords.
- 4. PC with windows(95/98/XP/NT/2000)
- 5. MATLAB Software with communication toolbox

THEORY:-

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse Width Modulation of the PTM is also called as the Pulse Duration Modulation (PDM) & less often Pulse length Modulation (PLM).

In pulse Width Modulation method, we have fixed and starting time of each pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.

This method converts amplitude varying message signal into a square wave with constant amplitude and frequency, but which changes duty cycle to correspond to the strength of the message signal.

Pulse-Width modulation has the disadvantage, that its pulses are of varying width and therefore of varying power content. This means that the transmitter must be powerful enough to handle the maximum-width pulses. But PWM still works if synchronization between transmitter and receiver fails, whereas pulse-position modulation does not. Pulse-Width modulation may be generated by applying trigger pulses to control the starting time of pulses from a mono stable multivibrator, and feeding in the signal to be sampled to control the duration of these pulses.

When the PWM signals arrive at its destination, the recovery circuit used to decode the original signal is a sample integrator (LPF).

CIRCUIT DESCRIPTION:-

Pulse & Modulating Signal Generator:-

A 4.096MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comprising a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U2), to produce 50% duty cycle, 1KHz square wave on pin no.1 of U4, and 2KHz square wave on pin no.15. the frequency is selectable by means of SW1. This goes to input of fourth order low pass filter U3 is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

The square wave which is generated by the oscillator is buffered by inverter 74HC04, to produce 32KHz square wave at pin no.4 of the 74HC4040(U2). This pulse is given to the monostable multi to obtain the 16KHz and 32KHz square wave at the output which are selected by the frequency pot.

Modulation:-

The PWM circuit uses the 555 IC (U1) in monostable mode. The Modulating signal input is applied to pin no.5 of 555IC, and there Pulse input is applied to pin no.2.

The output of PWM is taken at the pin no.3 of 555IC i.e., TP3.

Demodulation:-

The demodulation section comprises of a fourth order low pass filter and an AC amplifier. The TL074(U5) is used as a low pass filter and an AC amplifier. The output of the modulator is given as the input to the low pass filter.

The low pass filter output is obviously less and it is feed to the AC amplifier which comprises of a single op amp and whose output is amplified.

CIRCUIT DIAGRAM:



PROGRAM:-

% 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(311);

plot(x1);

axis([0 50 0 1]);

title('original signal taken mesage,f1=500,fs=10000')

subplot(312);

plot(y1);

axis([0 500 -0.2 1.2]);

title('PWM')

% demodulation

x1_recov=demod(y1,fc,fs,'pwm'); subplot(313); plot(x1_recov); title('time domain recovered, single tone,f1=200') axis([0 50 0 1]);

PROCEDURE:

Modulation:-

- 1. Connect the circuit as shown in the diagram 1.
 - a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position
 - b. 16KHz pulse output (by varying the frequency pot (put it min position) in pulse generator block) from pulse generator to pulse input(TP1).
- 2. Switch ON the power supply.
- 3. Observe the output of pulse width modulation block at TP3.(By varying the amplitude pot).
- 4. Vary the modulating signal generator frequency by switching the frequency selector switch to 2 KHz.
- 5. Now, again observe the PWM output at TP3.(By varying the amplitude pot).
- 6. Repeat the above steps (3 to 5) for the pulse frequency of 32KHz(by varying the frequency pot(put it in max position) in pulse generator block).
- 7. Switch OFF the power supply.

Demodulation:-

8. Connect the circuit as shown in diagram 2.

a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position.

b. 16KHz pulse output (put frequency pot minimum) from pulse generator block to pulse input TP1.

- c. PWM output to LPF input.
- d. LPF output to AC amplifier input.
- 9. Switch ON the power supply.

10. Observe the output of low pass filter and AC amplifier respectively at TP6 & TP8. The output will be the true replica of the input.

11. Now vary the position of the switch in modulating signal generator to 2 KHz and observe the outputs at TP6 & TP8.

12. Repeat the steps 10& 11 for pulse frequency 32 KHz (By varying the frequency pot (put in max). in pulse generator block). Observe the output waveforms.

13. Switch OFF the power supply.

EXPECTED WAVEFORMS





OUESTIONS

1. An audio signal consists of frequencies in the range of 100Hz to 5.5KHz.What is the minimum frequency at which it should be sampled in order to transmit it through pulse modulation?

2. Draw a TDM signal which is handling three different signals using PWM?

3. What do you infer from the frequency spectrum of a PWM signal?

4. Clock frequency in a PWM system is 2.5 kHz and modulating signal frequency is 500Hzhowmany pulses per cycle of signal occur in PWM output? Draw the PWM signal?

5. Why should the curve for pulse width Vs modulating voltage be linear?

6. What is the other name for PWM?

7. What is the disadvantage of PWM?

8. Will PWM work if the synchronization between Tx and Rx fails?

9. Why integrator is required in demodulation of PWM?

10. What kind of conversion is done in PWM generation?

EXPERIMENT NO-8 DATE:

PULSE POSITION MODULATION AND DEMODULATION

AIM:

1. To study the generation Pulse Position Modulation (PPM) and Demodulation.

2. To study the effect of Amplitude and the frequency of modulating signal on its output and observe the wave forms.

APPARATUS:

- 1. Pulse Position Modulation (PPM) and demodulation Trainer Kit.
- 2. C.R.O(30MHz)
- 3. Patch chords.
- 4. PC with windows(95/98/XP/NT/2000)
- 5. MATLAB Software with communication toolbox

THEORY:-

Pulse Modulation is used to transmit analog information in this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples. Pulse modulation may be subdivided in to two types analog and digital. In analog the indication of sample amplitude is the nearest variable. In digital the information is a code.

The pulse position modulation is one of the methods of the pulse time modulation.PPM is generated by changing the position of a fixed time slot.

The amplitude& width of the pulses is kept constant, while the position of each pulse, in relation to the position of the recurrent reference pulse is valid by each instances sampled value of the modulating wave. Pulse position modulation into the category of analog communication. Pulse-Position modulation has the advantage of requiring constant transmitter power output, but the disadvantage of depending on transmitter receiver synchronization.

Pulse-position modulation may be obtained very simply from PWM. However, in PWM the locations of the leading edges are fixed, whereas those of the trailing edges are not. Their position depends on pulse width, which is determined by the signal amplitude at that instant. Thus, it may be said that the trailing edges of PWM pulses are, in fact, positionmodulated. This has positive-going narrow pulses corresponding to leading edges and negative-going pulses corresponding to trailing edges. If the position corresponding to the trailing edge of an un modulated pulse is counted as zero displacement, then the other trailing edges will arrive earlier or later. They will therefore have a time displacement other than zero; this time displacement is proportional to the instantaneous value of the signal voltage. The differentiated pulses corresponding to the leading edges are removed with a diode clipper or rectifier, and the remaining pulses, is position-modulated.

Circuit Description:-

Modulating Signal Generator:-

A 4.096 MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comparing a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096 MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U4), to produce 50% duty cycle, 1 KHz square wave on pin no.1 of U4, and 2 KHz square wave on pin no.15. The frequency is selectable by means of SW1. This goes to input of fourth order low pass filter U3 (TL072) is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

Modulation:-

The circuit uses the IC 555(U1) a Mono stable Multivibrator to perform the pulse position Modulation action.

The Modulating signal is given to Pin No. 5 at Pin No.2 the pulse is 32 KHz which is connected internally.

The PWM is available at TP2; this PWM output is differentiated by using differentiated circuit. This differentiated output is available at TP8. This differentiated output is fed to the 555 IC (U2) (Mono stable Mode) Pin No.2. The PPM output is available at TP3.

CIRCUIT DIAGRAM:



PROGRAM:-

% 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('original signal taken mesage,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]);

PROCEDURE:

Modulation:

- 1. Connect the circuit as shown in diagram 1.
 - a. Connect the modulating signal generator output to modulating signal input (TP1) in PPM block.
 - b. Keep the switch in 1 KHz position and amplitude pot in max position.
- 2. Switch ON the power supply
- 3. Observe the PWM output at TP2, and the differentiated output signal at TP8.
- 4. Now, monitor the PPM output at TP3.
- 5. Try varying the amplitude and frequency of sine wave by varying amplitude pot.
- 6. Repeat Step 5 for frequency of 2 KHz and observe the PPM output.
- 7. Switch OFF the power supply.

Demodulation:-

- 8. Connect the circuit as shown in diagram2.
 - a. Connect the modulating signal generator output to modulating signal input (TP1) in PPM block.
 - b. Keep the switch in 1 KHz position and amplitude pot in max position.
 - c. Connect the PPM output (TP3) to input of LPF(TP4).
- 9. Switch ON the power supply
- 10. Observe the demodulated signal at the output of LPF at TP5.
- 11. Thus the recovered signal is true replica of the input signal

12. a. As the output of LPF has less amplitude, connect the output of LPF to the input of an AC amplifier (TP5 to TP6).

b. Observe the demodulated out put on the oscilloscope at TP7 and also

observe the amplitude of demodulated signal by varying gain pot. This is amplitude demodulated output.

13. Repeat the steps (7 to 9) for the modulating signal for frequency 2 KHz.

14. Switch OFF the power supply.

EXPECTED WAVEFORMS:



RESULT:

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OUESTIONS:

- 1. What is the advantage of PPM over PWM?
- 2. Is the synchronization is must between Tx and Rx
- 3. Shift in the position of each pulse of PPM depends on what?
- 4. Can we generate PWM from PPM?
- 5. Why do we need 555 timers?
- 6. Does PPM contain derivative of modulating signal compared to PWM?
- 7. For above scheme, do we have to use LPF and integrator in that order?

8. If we convert PPM to PWM & then detect the message signal, will the o/p has less distortion?

- 9. Is synchronization critical in PPM?
- 10. How robust is the PPM to noise?



EXPERIMENT NO-1

PULSE CODE MODULATION & DEMODULATION

Aim: To convert an analog signal into a pulse digital signal using PCM system and to convert the digital signal into analog signal using PCM demodulation system.

Apparatu: 1. PCM transmitter trainer.

- 2. PCM receiver trainer.
- 3. CRO and connecting wires.

Theory:

In the PCM communication system, the input analog signal is sampled and these samples are subjected to the operation of quantization. The quantized samples are applied to an encoder. The encoder responds to each such a sample by generation unique and identifiable binary pulse. The combination of quantize and encoder is called analog to digital converter. It accepts analog signal and replaces it with a successive code symbol, each symbol consists of a train of pulses in which the each pulse represents a digit in arithmetic system.

When this digitally encoded signal arrives at the receiver, the first operation to be performed is separation of noise which has been added during transmission along the channel. It is possible because of quantization of the signal for each pulse interval; it has to determine which of many possible values has been received.


<u>"</u>

Procedure:

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- 1. The two inputs of function generator are connected to channel -0 and channel -1 simultaneously that is DC₁ output to channel -0 and DC₂ to channel -1.
- 2. With the help of oscillator DC_1 output is adjusted to 0 volts.
- 3. Transmitter and receiver are connected by the synchronization of clock pulses and by connecting ground transmitter to ground receiver.
- 4. The transmitter is connected to the input of receiver to go the original signal at the receiver output.
- 5. After connection is made the inputs channel 1 and channel 0 are noted. The sampled output of bit channels are taken by connecting DC_1 output to channel 0 and DC_2 output to channel-1.
- 6. The phase shift of a channel can be obtained by comparing the input and output of channels at the transmitter block.
- 7. Thus the output of transmitter can be noted down and input of receiver is similar to that.
- 8. The receiver output signals are noted down at channel 0 and channel 1 of the receiver block.

Result:

Questions:

- 1. What is the expression for transmission bandwidth in a PCM system?
- 2. What is the expression for quantization noise /error in PCM system?
- 3. What are the applications of PCM?
- 4. What are the advantages of the PCM?
- 5. What are the disadvantages of PCM?

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EXPERIMENT NO-2

TIME DIVISION MULTIPLEXING & DEMULTIPLEXING

Aim:

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Study of 4 Channel Analog Multiplexing and De multiplexing Techniques.

Apparatus:

1. Time division multiplexing & demultiplexing trainer kit.

2. CRO (30 mhz)

3.Patch chords.

Theory:

The TDM is used for transmitting several analog message signals over a Communication channel by dividing the time frame into slots, one slot for each message signal. The four input signals, all band limited by the input filters are sequentially sampled, the output of which is a PAM waveform containing samples of the input signals periodically interlaced in time. The samples from adjacent input message channels are separated by Ts/M, where M is the number of input channels. A set of M pulses consisting of one sample from each of the input M-input channels is called a frame.

At the receiver the samples from individual channels are separated by carefully synchronizing and are critical part TDM. The samples from each channel are filtered to reproduce the original message signal. There are two levels of synchronization. Frame synchronization is necessary to establish when each group of samples begins and word synchronization is necessary to properly separate the samples within each frame.

Besides the space diversity & frequency diversity there is a method of sending multiple analog signals on a channel using "TIME DIVISION MULTIPLEXING & DEMULTIPLEXING" Technique.



Procedure:

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Multiplexing:

1. Connect the circuit as shown in diagram.

2. Switch ON the power supply.

3.Set the amplitude of each modulating signal as 5vpeak-peak.

4.Monitor the outputs at test points 5,6,7,8. these are natural sampling PAM outputs. 5.Observe the outputs varying the duty cycle pot(P5). The PAM outputs will varying with 10% to 50% duty cycle.

6. Try varying the amplitude of modulating signal corresponding each channel by using amplitude pots P1,P2,P3,P4. Observe the effect on all outputs.

7. Observe the TDM output at pin no.13 (at TP9) OF 4052. all the multiplexer channel are observed during the full period of the clock(1/32 KHz).

Demultiplexing & Low Pass Filter:

1. Connect the circuit as shown in diagram 2.

2. Observe the demultiplexed outputs at test points 13,14,15,16 respectively. 3. Observe by varying the duty cycle pot P5 and see the effect on the outputs. 4. Observe the low pass filter outputs for each channel at test points 17,18,19,20 and at sockets channels CH1,CH2,CH3,CH4. These signals are true replica of the inputs. These signals have lower amplitude.



Result:

Questions:

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- 1. Draw the TDM signal with 2 signals being multiplexed over the channel?
- 2. Define guard time & frame time?
- 3. Explain block schematic of TDM?
- 4. How TDM differ from FDM?
- 5. What type of filter is used at receiver end in TDM system?
- 6. What are the applications of TDM?
- 7. If 2 signal band limited to 3 kHz, 5KHz & are to be time division multiplexed.
- What is the maximum permissible interval between 2 successive samples.?
- 8. Is the bandwidth requirement for TDM & FDM will be same?
- 9. Is TDM system is relatively immune to interference with in channels (inter channel cross talk) as compared to FDM?
- 10. Is the FDM susceptible to harmonic distortion compared to TDM?
- 11. In what aspects, TDM is superior to FDM?

EXPERIMENT NO-3

DIFFERENTIAL PULSE CODE MODULATION

Aim:

To study the differential pulse code modulation and demodulation by sending variable frequency sine wave and variable DC signal outputs.

Apparatus:

- 1. DPCM Trainer kit
- 2. Patch cards
- 3. CRO- (0-20MHz)
- 4. AC Adapter (± 8V)
- 5. CRO Probes.

Theory:

In Differential Pulse Code Modulation (DPCM), instead of quantizing each sample, the difference between the two successive samples is quantized, encode, and transmitted as in the PCM. This particularly useful in the Voice communication, because in this case two successive samples do not differ much in amplitude.

Thus, the difference signal is much less in amplitude than the actual sample and, hence, less number of quantization levels is needed. Therefore, the number of bits per code is reduced, resulting in a reduced bit rate. Thus, the band width required in this case is less than the one required in PCM.

The disadvantage of DPCM is that the modulator and demodulator circuits are more complicated than those in PCM.







Procedure:

- 1. Switch on the Kit.
- 2. Apply the variable DC signal to the input terminals of DPCMmodulator.
- 3. Observe the sampling signal output on CRO
- 4. Observe the output of DPCM on the second channel of CRO.

5. By adjusting the DC voltage potentiometer we can get the DPCM output from 0000 0000 to 1111 1111.

6. Now, disconnect the DC voltage and apply AF oscillator output to the input of the DPCM modulator

7. Observe the output of conditioning amplifier (differential output) and DPCM outputs in synchronization with the sampling signal.

8. During demodulation, connect DPCM output to the Input of demodulation and observe the output of Demodulator

Observations:

1.	mplitude of AF signal =-		
 Frequency of Amplitude o 	⁷ AF signal f Synchronous clock signa	= ıl =	
 Frequency of Amplitude of 	Synchronous clock signa f DPCM Modulated signal	l =	=
6. Frequency of 7. Amplitude of	DPCMModulated signal demodulated output		= =
8. Frequency of	demodulated output		=

Result:



EXPERIMENT NO-4

DELTA MODULATION & DEMODULATION

Aim:

To study the Delta modulation process by comparing the present signal with the previous signal of the given modulating signal.

Apparatus:

- 1. Delta Modulation trainer
- 2. CRO
- 3. Connecting wires.

Theory:

DM uses a single bit PCM code to achieve to achieve digital transmission of analog signal. With conventional PCM each code is binary representation of both sign and magnitude of a particular sample. With DM, rather than transmitting a coded representation of a sample a single bit is transmitted, which indicates whether the sample is smaller or larger than the previous sample. The algorithm for a delta modulation system is a simple one. If the current sample is smaller than the previous sample then logic 0 is transmitted or logic 1 is transmitted if the current sample is larger than the previous sample. The input analog is sampled and converted to a PAM signal followed by comparing it with the output of the DAC. The output of the DAC is equal to the regenerated magnitude of the previous sample which was stored in the up/down counter as a binary number. The up/down counter is incremented or decremented whether the previous sample is larger or smaller than the current sample. The up/down counter is clocked at a rate equal to the sample rate. So, the up/down counter is updated after each comparison.



Procedure:

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- 1. Switch on the experimental board
- 2. Connect the clock signal of Bit clock generator to the bit clock input of Deltamodulator circuit.
- 3. Connect modulating signal of the modulating signal generator to the
- modulating signal input of the Delta modulator.
- 4. Observe the modulating signal on Channel 1 of CRO
- 5. Observe the Delta modulator output on channel 2 of CRO
- 6. Connect the DM o/p of modulator to the DM I/P of Demodulator circuit.
- 7. Connect the clock signal to the Bit clock I/P of Demodulator circuit.
- 8. Observe the demodulated o/p on channel 2 ofCRO.

Connect the demodulated o/p to the filter input of demodulator circuit.

9. Observe the demodulated o/p with filter on CRO.

Expected Waveforms:





Filtered Output

Result:

Questions:

1.What is Delta

Modulation?

2.Differentiate DM and

ADM.

3. What are the drawbacks of DM and what is

theremedy?4.How DM differ from PCM?

5. What is slope overload distortion?

EXPERIMENT NO-5

ASK MODULATION AND DEMODULATION

Aim:

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To study the process of ASK modulation& demodulation and study various data formatting modulation and demodulation techniques.

Apparatus:

1. ASK MODULATION & DEMODULATION Trainer kit.

2. CRO 30MHz Dual Channel.

3. Patch Chords.

Theory:

Modulation also allows different data streams to be transmitted over the same channel. This process is called as 'Multiplexing' & result in a considerable saving in bandwidth no of channels to be used. Also it increases the channel efficiency.

The variation of particular parameter variation of the carrier wave give rise to various modulation techniques. Some of the basic modulation techniques are described as under.

ASK:-

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In this modulation involves the variation of the amplitude of the carrier waves in accordance with the data stream. The simplest method of modulating a carrier with a data stream is to change the amplitude of the carrier wave every time the data changes. This modulation technique is known as amplitude shift keying.

The simplest way of achieving amplitude shift keying is 'ON' the carrier whenever the data bit is 'HIGH' & switching 'OFF' when the data bit is low i.e. the transmitter outputs the carrier for HIGH & totally suppresses the carrier for low. This technique is known as ON-OFF keying Fig. illustrates the amplitude shift keying for the given data stream.

Thus,

DATA = HIGH CARRIER TRANSMITTED DATA = LOW CARRIER SUPPRESSED

The ASK waveform is generated by a balanced modulator circuit, also known as a linear multiplier, As the name suggests, the device multiplies the instantaneous signal at its two inputs, the output voltage being product of the two input voltages at any instance of time. One of the input is a/c coupled 'carrier' wave of high frequency. Generally the carrier wave is a sinewave since any other waveform would increase the bandwidth imparting any advantages requirement without improving or to it. The other i/p which is the information signal to be transmitted, is D.C. coupled. It is known as modulating signal.

In order to generate ASK waveform it is necessary to apply a sine wave at carrier input & the digital stream at modulation input. The double balanced modulator is shown in fig.

Block Diagram:

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The data stream applied is uniploar i.e. 0Volt at logic LOW & +4.5Volts at logic HIGH. The output of balanced modulator is a sinewave, unchanged in phase when a data bit 'HIGH' is applied to it. In this case the carrier is multiplied with a positive constant voltage when the data bit LOW is applied, the carrier is multiplied by 0 Volts, giving rise to 0Volt signal at modulator's o/p.

The ASK modulation results in a great simplicity at the receiver. The method to demodulate the ASK waveform is to rectify it, pass it through the filter &'square up' the resulting waveform. The o/p is the original digital data stream. Fig. shows the functional blocks required in order to demodulate the ASK waveform at receiver.



The function of the carrier is to generate a stable sine wave signal at the rest frequency, when no modulation is applied. It must be able to linearly change frequency when fully modulated, with no measurable change in amplitude.

Sine wave is generated by using the colpitts Oscillator. 500KHz and 1MHz frequencies are selected.

Modulation Generation:-

The square wave generated by 555 and is given to 74121. the o/p of this multivibrator is used as a clock i/p to a decade counter 7490. Which generators the modulating data outputs D1,D2, D3,D4.

Modulator:-

The ASK215 Modulator is based on U2(LM 1496). It is configured as a linear multiplier. At any movement of time the o/p of this U2(PIN 12) is proportional to the instantaneous product of the CARRIER INPUT and MODULATION INPUT saignals which serves as two inputs to this U2. The CARRIER INPUT can be monitored at TP7 & The MODULATION INPUT can be monitored at TP8.

The o/p voltage from U2 can be adjusted in amplitude by potentiometer P3(5K). it is now fed to the OP-AMP U3, LF 356 at its non-inverting terminal(pin 3). The op-amp configured as a simple non inverting amplifier with the gain of 2.47. the o/p(pin 6 is a/c coupled by capacitor C18 to appear at the o/p of OUTPUT socket.to signal multiplication. The DC bias from both the signals can be removed by careful setting of the two potentiometers.

Demodulation:-

The ASK demodulator comprises of

- 1) op-amp ICU6A configured as a unity gain, non inverting buffer, and
- 2) A simple half wave rectifier circuit, consisting diode D1 and resistor R72. The incoming ASK signal can be monitored at TP12. the signal at TP12 is then buffered by ICU6A & then rectified by half wave rectified CKT comprising of Diode D5 & resistor R72. This removes the negative half cycle of the wave form. The output of rectifier is available at OUTPUT socket of the demodulator & can be monitored at TP13. example waveforms are as shown in the timing diagram in Fig.

Low Pass Filters:-

The low pass filter block consists of two fourth-order butter worth low pass filter circuit. The filter is identical & i.e. is described in the section to follow.

The input signal to this block is first buffered by the op-amp ICU6B. The opamp is simply configured as a noninvert in, unity-gain buffer. The buffer output(TP15) is then fed into data squaring circuit. The final o/p's of the filter can be monitored at TP15.

Data Squaring Circuits:-

The data squaring circuit 'square up' the input signal. It does this with the help of voltage comparator. The function of comparator circuit is identical & hence only one is described. The input is connected to the non-inverting(+ve) input(pin 5) of the voltage comparator ICU4A whose inverting (-ve) input(pin 4) is connected to a voltage divider network of resistors R61, R60 & variable Resistor P4 through resistor R59. the input impedance of the comparator circuit is set to 100k by resistor R58. A hystersis of 0.3V is set by resistor R59& R57. the slider voltage of can be adjusted from 2.2V to +2.2V.

The output of the comparator is 0V when the input at inverting terminal is more positive then the input at non inverting terminal.

Procedure:

Modulation:

1. Connect the sinewave 500KHz from the carrier generator TP1 to the carrier input of the modulator TP7.

2. And also connect data clock D1 i.e., modulation signal TP3 to the modulation input TP8.

3.Switch ON the power supply.

4. Observe the output at TP9.

5. By varying the gain pot P3 observe the ASK output at TP10.

6. Adjusting the carrier offset and modulation offset we can observe the ASKoutput.

7. By changing the carrier signal 1MHz and different data clocks D2,D3,D4 observe the output.

Demodulation:

1. Connect ASK output TP10 to the rectifier input TP12 and observe thewaveform.

2. Now connect rectifier output TP13 to the low pass filter input TP14 and observe he output at TP15.

3. CONNECT LPF output TP15 to the data squaring circuit input TP16 and observe the demodulation output waveform at TP17.

4.By changing the different data clocks and observe the demodulation output.

Expected Waveforms:



Result:

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Questions:

1. If the bit rate of an ASK signal is 1200bps, what is the baud rate?

2.Is ASK highly susceptible?

3. What are the characteristics of transmission medium which effect speed of transmission in ASK?

4. Find the minimum band width for an ASK signal transmitting at 2000bps. The transmission made is half duplex?

5. If B.W is 5000Hz for an ASK signal, what are the baud rate?

6. What is the advantage of ON-OFF keying () in ASK?

7. Given the bandwidth of 10KHz(1Hz to 1KHz), Find the band width for upper side & lower side band of carrier in full duplex ASK?

8. For the above problem, what are the carrier frequencies in upper & lower side bands?

EXPERIMENT NO-6

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FREQUENCY SHIFT KEYING

Aim:

- 1. To generate FSK Modulation
 - 2. To Demodulate the FSK signals
 - 3.To generate

NRZ(L),RZ,NRZ(M),BIPHASE(MARK),BIPHASE(MANCHESTER).

Apparatus:

- 1. Frequency Shift Keying kit 2.C.R.O (30MHz)
- 3.Patch cords

Circuit Diagram:



Theory:

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Binary FSK is a form of constant-amplitude angle modulation and the modulating signal is a binary pulse stream that varies between two discreate voltage levels but not continuous changing analog signal. In FSK, the carrier amplitude(Vc) remains constant with modulation and the carrier radian frequency(wc) shifts by an amount equal to +w/2. The frequency shift is proportional to the amplitude and polarity of the input binary signal. For example, a binary 1 could be +1 volt and a binary zero could be -1 volt producing frequency shifts of +delta(w)/2 and -delta(w)/2 respectively. The rate at which the carrier frequency shifts is equal to the rate of change of the binary input signal vm(t). thus the output carrier frequency deviates(shifts) between wc+delta(w)/2 and wc-delta(w)/2 at the rate equal to fm.

Data Formating:-

A modulation code is defined as a rule by which a serial train of binary data is converted to a signal suitable for transmission. Some of the commonly used codes are listed for study in this experiment. There are few others which are outside the scope of this experiment.

In serial data transmission, a 'symbol' is a signal level that is held for a length of time. The capacity of a channel is the symbol rate. This is the symbols per second or baud. Channel capacity has the units of symbols per second or baud. Some modulation codes require several symbols per bit of data. For example self clocking codes require two symbols per bit of data. The various codes are described below. Relative features of the codes are given in the table. The waveform diagram the patterns for the serial train 11001100.

Non-return to zero(NRZ):-

This is level type code and is one that is widel used in serial data transmission. A '0' is low level and a '1' is a high level.

Return to Zero(RZ):-

This is an impulse type code where a '1' is represented by a high level that returns to zero. Its advantage is power conservation as transmission takes place only for '1'.

NRZ(M):-

If the logic '0' is to be transmitted the new level is inverse of the previous level i.e., change in level occurs. If '1' is transmitted the level remainsunchanged.

Biphase(Mark):-

This is an edge type invertible self-clocking code in which each bit cell starts with an edge and for a '0' an additional edge occurs during the middle of the bit cell.

Biphase(Manchester):-

This is a level type of code in which a '1' bit cell is initially high and then has a high to low transition in the middle of the bit cell. A '0' bit cell is initially low and has a low to high transition in the middle of the bit cell.

Circuit Description:-

Data clock

Generator:-

The bit clock generator is design around the tim IC 555(U1) operated in astable mode. The 100Kohm preset P1 in conjunction with .0047microfarad capacitor in the timing circuit facilitates the frequency to be set and at any chosen value from 300Hz to 1KHz. This output is available at TP1.

Data Selection:-

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The 8 bit parallel load serial shift IC 74165(U2) is used to generate the required word pattern. A dip switch is used to set ONE&ZERO pattern. The bit pattern set by the switch is parallely loaded by controlling the logic level at pin 1. The last stage output Q7 is coupled to the first stage input D0 in the shift register. The serial shift clock is given at pin 2. The 8 bit data set by the switch and loaded with the register parallely is now shifted serially right and circulated respectively. Thus the 8 bit word pattern is generated cyclically which is used as modulating signal in the FSK modulator. It is available at TP12.

Fsk Modulation:-

The XR-2206 can be operated with two separate timing resistors, R24 and R25, connected to the timing pin 7 and 8, respectively. Depending on the polarity of the logic signal at pin 9, either one or the other of these timing resistors is activated. If pin 9 is open-circuited or connected to a bias voltage>2V, only R24 is activated. Similarly, if the voltage level at pin 9 is<1v, only R25 is activated. Thus, the output frequency can be keyed between two levels. F1 and F2.

F1=1/R24C9 and f2=1/R25C9. In our circuit R24=3.9Kohm, R25=6.8Kohm, C9= 100nf.For split-supply operation, the keying voltage at pin 9 is referenced to V. the FSK output can be monitored at TP8Demodulation:-

Square Wave Converter:-

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The incoming FSK modulated signal can be monitored at TP9. This signal is then attenuated by resistor network R43,R44 then AC coupled via capacitor C12 to remove any dc component in the signal. The signal is connected to SIGIN input of the U12. The signal is first squared up by an inbuilt comparator and is connected to one of the input of on chip2 input EX-OR gate. The other 5 input of the gate is connected to the COMPIN input of IC U12. The output is monitored at TP10.

PLL Detector:-

A very useful application of the 565 PLL is as a FSK demodulator. In the 565 PLL the frequency shift is usually accomplished by driving a VCO with the binary data signal so that the two resulting frequencies correspond to the logic 0 and logic 1 states of the binary data signal. The frequencies corresponding to logic 1 and logic 0 states are commonly called the mark and space frequencies. Capacitive coupling is used at the input to remove a dc level. As the signal appears at the input of the 565, the loop to the input frequency and tracks it between the two frequencies with a corresponding dc shift at the output. Preset p2 and capacitor C15 determine the freerunning frequency of the VCO. A three-stage RC ladder filter is used to remove the carrier component from the output. The high cutoff frequency of the ladder filter is chosen to be approximately halfway between the max keying rate and twice the input frequency.This output signal can be made logic compatible by connecting voltage comparator(u11) between the output of ladder filter and pin 6 of PLL.

Phase Adjustment Circuit:-

U17,U18 used as phase adjustment circuit. The output of voltage comparator is fed yto pin 2 of U17 which is connected as monostable mode. And the output of U17 is again fed to U18. The output is available at pin 3 of U18 can be monitored at TP11. This is serial date of output.

Procedure:

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Modulation:

1. Switch ON the power supply.

2. Set the data selection switch ('<u>DATA SELECTION'</u>) to the desired code(say 11001100).

3. Set the switch (DATA ON-OFF) ON position. Observe the 8 bitWord pattern at TP12.

4. Observe the data clock at TP1 and also observe the NRZ(L) at TP2,RZ at TP3,NRZ(M) at TP4, BIPHASE(MARK) at TP5,BIPHASE(MANCHESTER) at TP6.

5. Connect the patch cord as shown in diagram 1. Observe the corresponding FSK output at(when data is logic '1', the frequency is high and data is logic '0' the frequency is low)TP8.

6. Repeat the step 5 for other inputs.(like NRZ(M),RZ,BIPHASE) observe the corresponding FSK outputs.

7. Now change the data selection and repeat the above steps 3 to 6 and observe the corresponding FSK outputs.

Demodulation:

- 1. Connect the patch cords as shown in diagram.
- 2. The incoming FSK input is observed at TP9.

3. The output of 'square wave converter' is available at TP10. The serial data output is available at TP11.

4. Repeat the above steps 1,2,3 for other serial data inputs and observe the corresponding serial data outputs. These outputs are true replica of the orginal inputs.

Expected Waveforms:



Result:

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Questions:

1.Explain the concept of FSK?

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2.Compare ASK, FSK & PSK?

3.Draw the waveforms of FSK?

4. What is M-ray signaling? What is its advantages over 2-ary signaling?

5. What are the different data coding formats & draw the waveforms what is advantages of Manchaster coding over other formats?

6. Explain the demodulation scheme of FSK?

7. What is the formula for Band Width required in FSK?

8. What is the minimum B.W for an FSK signal transmitting at 2000bps(haif duplex), if carriers are separated by 3KHz?

9. Is the FSK spectrum, a combination of two ASK spectra centered around two frequencies?

10. Is the FSK band width is more than ASK band width for a given band rate?

11. Is it more susceptible to noise than ASK?

12. What are the limiting factors of FSK?

13. Is the band rate & bit rate are same for FSK?

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EXPERIMENT NO-7

BINARY PHASE SHIFT KEYING

Aim: Study of carrier Modulation techniques by phase shift keying method.

Apparatus:

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- 1. Psk Modulation And Demodulation Trainer.
- 2. 30MHz Dual Trace Oscilloscope.
- 3. Patch chords

Circuit Diagram:




Theory:

To transmit the digital data from one place to another, we have to choose the transmission medium. The simplest possible method to connect the transmitter to the receiver with a piece of wire. This works satisfactorily for short distances in some cases. But for long distance communication & in situations like communication with the aircraft, ship, vehicle this is not feasible. Here we have to opt for the radio transmission.

It is not possible to send the digital data directly over the antenna because the antenna of practiced size works on very high frequencies, much higher than our data transmission rate.

To able to transmit the data over antenna, we have to 'module' the signal i.e., phase, frequency or amplitude etc. is varied in accordance with the digital data. At receiver we separate the signal from digital information by the process of demodulation. After this process we are left with high frequency signal which we discard & the digital information, which we utilize.

Modulation also allows different data streams to be transmitted over the same channel.

This process is called as 'multiplexing' & result in a considerable saving in bandwidth no of channels to be used. Also it increases the channel efficiency.

The variation of particular parameter variation of the carrier wave give rise to various modulation techniques. Some of the basic modulation techniques ASK,FSK, PSK,DPSK,QPSK.

Phase Shift Keying(PSK):

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The PSK is a form of angle modulated, constant amplitude digital modulation. Digital communications because important with the expansion of the use of computers and data processing, and have continued to develop into a major industry providing the interconnection of computer peripherals and transmission of data between distant sites. Phase shift keying is a relatively new system, in which the carrier may be phase shifted by +90 degree for a mark, and by-90 degrees for a space. PSK has a number of similarities to FSK in may aspects, as in FSK, frequency of the carrier is shifted according to the modulating square wave.

Circuit Description:

In this IC 8038 is a basic wave form generator which generates sine, square, triangle waveforms. The sine wave generated by this 8038 IC is used as carrier signal to the system. This square wave is used as a clock input to a decade counter which generates the modulating data outputs.

The digital signal applied to the modulation input for PSK generation is bipolar i.e. have equal positive and negative voltage levels. When the modulating input is negative the output of modulator is a sine wave in phase with the carrier input. Where as for the positive voltage levels, the output of modulator is a sine wave which is shifted out of phase by 180 degree from the carrier input compared to the differential data stream. This happens because the carrier input is now multiplied by the negative constant level.

Thus the output changes in phase when a change in polarity of the modulating signal results. Fig shows the functional blocks of the PSK modulator & demodulator.

Modulation:-

IC CD 4051 is an analog multiplexer to which carrier is applied with and without 180 degree phase shift to the two multiplex inputs of the IC. Modulating data input is applied to its control input. Depending upon the level of the control signal

carrier signal applied with or without phase shift is steered to the output. the 180 degree phase shift to the carrier signal created by an operational amplifier using 741C.

Demodulation:-

During the demodulation the PSK signal is converted into a +5volts square wave signal using a transistor and is applied to one input of an EX-OR gate. To the second input of the gate carrier signal is applied after conversion into a +5volts signal. So the EX-OR gate output is equivalent to the modulating data signal.

Procedure:

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- 1. Now switch ON the trainer and see that the supply LED glows.
- 2. Observe the carrier output at TP1.
- 3. Observe the data outputs(D1,D2,D3,D4).
- 4. Now the connect the carrier output TP1 to the carrier input of PSK modulator TP2 using patch chord(as shown in dig 1).
- 5. Connect the d1 to data input of PSK modulator TP3(As shown.in dig 1).
- 6. Observe the phase shifted PSK output waveform on CRO on channel1 and corresponding data output on channel2.
- 7. Repeat the steps 4,5,6 for data outputs D2,D3,D4 and observe the PSK outputs.
- 8. connect the PSK modulation output TP6 to the PSK input ofdemodulation TP4(as shown in dig 2).
- 9. connect the carrier output TP1 to the carrier input of PSK demodulation TP5.(As shown in dig 2).
- 10. Now, observe the PSK demodulated output at TP7 on CRO at channel 1 and corresponding data output on channel 2.
- 11. the demodulated output is true replica of dataoutput.
- 12. Repeat the steps 8 to 10 for other data outputs D2,D3,D4.



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EXPERIMENT NO-8

DIFFERENTIAL PHASE SHIFT KEYING

Aim: To study operation Differential Phase shift Keying modulation & demodulation Techniques.

Apparatus:

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- 1. DPSK MODULATION & DEMODULATION Trainer.
- 2. Oscilloscope 30MHz, Dual Channel
- 3. Path chords.

Block Diagram:







Theory:

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To transmit the digital data from one place to another, we have to choose the transmission medium. The simplest possible method to connect the transmitter to the receiver with a piece of wire. This works satisfactorily for short distances in some cases. But for long distance communication & institutions like communication with the aircraft, ship, vehicle this is not feasible. Here we have to opt for the radio transmission.

It is not possible to send the digital data directly over the antenna because the antenna of practiced size works on very high frequencies, much higher than our data transmission rate.

To be able to transmit the data over antenna, we have to 'module' the signal i.e. phase, frequency or amplitude etc. is varied in accordance with the digital data. At receiver we separate the signal from digital information by the process of demodulation. After this process we are left with high frequency signal(called as carrier signal) which we discard & the digital information, which we utilize.

Modulation also allows different data streams to be transmitted over the same channel(transmission medium).

This process is called as 'Multeplexing' & result in a considerable saving in bandwidth no of channels to be used. Also it increases the channel efficiency.

The variation of particular parameter variation of the carrier wave give rise to various modulation techniques. Some of the basic modulation techniques are ASK,FSK,PSK,DPSK & QPSK.

Differential Phase-Shift Keying(DPSK):-

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The DPSK is a non-coherent version of PSK. In coherent detection, the carrier wave's phase reference should be known for obtaining optimum error performance.(However it is impractical to have knowledge of the carrier phase at the receiver).

The DPSK eliminates the need for a coherent reference signal at the receiver by combining two basic operations at the transmitter:

1. Differential Encoding of the input binary wave2. Phase-shift keying

And hence the name differential phase shift keying. Thus to send symbol 0,we phase advance the current signal waveform by 180 degrees and to send 1, we have the phase of the current signal waveform unchanged. The receiver is equipped with a storage capability so that it can measure the relative phase difference between the wave forms received during two successive bit intervals. Provided that the unknown phase θ contained in the received wave varies slowly (slow enough and considered essentially constant over two bit intervals), the phase difference between waveforms received in two successive bit intervals will be independent of θ .

Circuit Description:-

In this IC 8038 is a basic wave form generator which generates since, square, triangle waveforms. The sine wave generated by this 8038 IC is used as carrier signal to the system. This square wave is used as a clock input to a decade counterwhich generates the modulating data outputs.

The digital signal applied to the modulation input for DPSK generation is bipolar have equal positive and negative voltage levels. When the modulating input is negative the output of modulator is a sinewave in phase with the carrier input. Where as for the positive voltage levels, the output of modulator is a sinewave which is shifted out of phase by 180 degrees from the carrier input compared to the differential data stream. This happens because the carrier input is now multiplied by the negative constant level.

Thus the output changes in phase when a change in polarity of the modulating signal results. Fig shows the functional blocks of the DPSK modulator &demodulator.

Modulation:-

The differential signal to the modulating signal is generated using an X-OR gate and 1-bit delay circuit(it is shown in fig).CD 4051 is an analog multiplexer to which carrier is applied with and without 180 degrees phase shift(created by using an operational amplifier connected in inverting amplifier mode) to the two inputs of the ICTL084.Differential signal generated by X-OR gate is given to the multiplexer's control signal input. depending upon the level of the control signal,carrier signal applied with or without phase shift is steered to the output. 1-bit delay generation of differential signal to the input is created by using a D-flip-flop(IC7474).

Demodulation:-

During the demodulation, the DPSK signal is converted into a +5V square wave signal using a transistor and is applied to one input of an X-OR gate. to the second input of the gate, carrier signal is applied after conversion into a +5V signal. So the X-OR gate output is equivalent to the differential signal of the modulating data. This differential data is applied to one input of an X-OR gate and to the second input, after 1-bit delay the same signal is given. So the output of this X-OR gate is modulating signal.

Output Waveforms:-

To see the DPSK demodulation process, examine the input of DPSK demodulator with the demodulation output.

Check the various test points provided at the output of the functional blocks of the DPSK demodulator. This will help you fully grasp the DPSK demodulation technique.

Figure 1.4:

b'(t)		0	1	1	0	0
b(t)	1	0	0	0	1	0
Phase	00	180 ⁰	180 ⁰	180 ⁰	00	180 ⁰
B(t)	0	1	1	1	0	1
Phase	180^{0}	0^{0}	0^{0}	0^{0}	180^{0}	0^0

I	Message signal(to be transmitted)	0	1	1	0	0	
*	Encoded data(differential data)	0	1	1	1	0	1
*	Trasnmitted signal phase:	180 ⁰	0^{0}	00	0^0	180 ⁰	0^0
~ ~	Received signal phase :	180 ⁰	0^0	00	00	180 ⁰	00
~	Encoded data(differential data)		0	1	1	1	0
e ⁶	Message signal (Demodulation)		0	1	1	0	0

Procedure:

- 1. Now switch ON the trainer and see that the supply LED glows.
- 2. Connect data output from 4(D1,D2,D3,D4) data outputs to the data input of the DPSK modulator TP7.
- 3. Connect clock output TP1 to the clock input of the DPSK modulator TP8.
- 4. Now connect carrier output TP2 to the carrier input of the DPSKmodulator TP10.
- 5. Observe the differential data output on the CRO at TP9 test point as shown on the front panel.
- 6. Observe the phase shifted DPSK output waveform on the CRO corresponding to the differential data output.
- 7. Connect DPSK MODULATOR output TP11 to the DPSK input of the DEMODULATOR TP12.
- 8. Connect carrier output TP2 to the carrier input of the DPSKDemodulator TP13.
- 9. Also connect clock output TP1 to the clock input of the DPSK demodulator TP14.
- 10. Now observe the DPSK demodulated output waveform TP15 on the CRO.



RESULT:

Questions:

1. How does DPSK differ from PSK?

- 2. Explain theoretical modulation & demodulation of DPSK using arbitrary bit sequence and assuming initial bit 0 and 1?
- 3. What is the advantage of DPSK over PSK?
- 4. Why do we need 1 bit delay in DPSK modulator & demodulator?
- 5. What does a synchronous detector (multiplier) do in DPSK demodulator?
- 6. what is the relation between carrier frequency & the bit interval 'T'?
- 7. What is the disadvantages of DPSK.?
- 8. Is the error rate of DPSK is greater than PSK?
- 9. What is the expression for DPSK error?
- 10. What are the applications of DPSK?