Govt. of N.C.T of Delhi

G.B PANT ENGINEERING COLLEGE

Okhla Industrial Estate Phase-III, New Delhi – 110020

Department of Electronics and Communication Engineering

B. Tech (CSE) – 5th Semester

ETEC-357 Digital Communication Lab Practical File

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AIM

To study Sampling Theorem

APPARATUS REQUIRED

System Running with MATLAB software

THEORY

Sampling:

By sampling a continuous time signal at isolated equally spaced points in time, we obtain sequence of numbers

$n \in \{ \dots, -2, -1, 0, 1, 2, 3, \dots \}$

T_s is the Sampling Period

Many Signal originate as continuous-time signal, eg-> Conventional music or voice

Frequency Domain:

- Replicates Spectrum of Continuous-Time signal at offsets that are integer multiples of sampling frequency.
- Fourier Series of impulse train where $\omega_s = 2\pi f_s$
- As sampling rate increases, sampled waveform looks more and more like the original.
- Many applications (eg. communication systems) care more about frequency content in the waveform and not its shape
- zerocrossing: frequency content of sinusoid when it meets the axis.

Teacher's Signature :

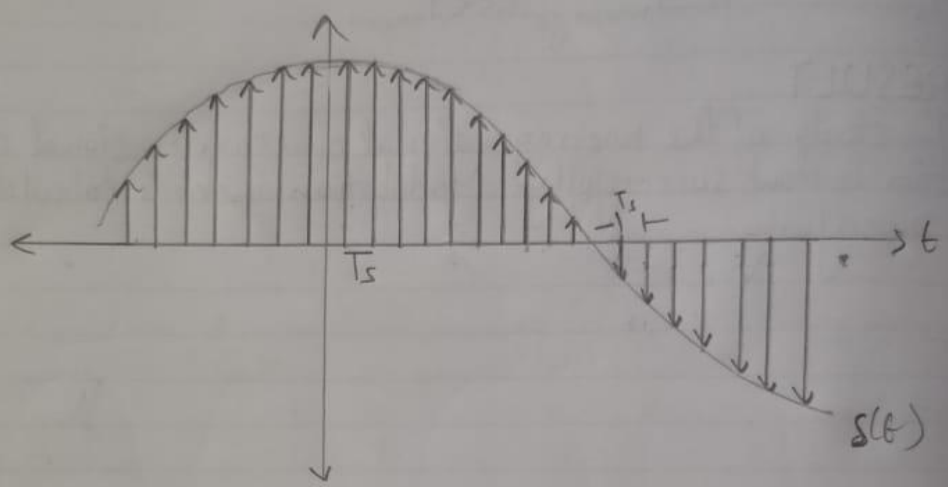

Fig: Sampled Analog Waveform

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Shanon Sampling Theorem

A continuous time signal $x(t)$ with frequency no higher than f_{max} can be reconstructed from its samples $x[n] = x(nT_s)$ by taking samples at a rate f_s which is greater than $2f_{max}$.

$$\text{Nquist rate} = 2f_{max}$$
$$\text{Nqust frequency} = f_s/2$$

Consider a sinusoid $\sin(2\pi f_{max} t)$
use a sampling period of $T_s = 1/f_s = 1/2f_{max}$

Assumptions

1. Continuous Time signal has no frequency content above f_{max}
2. Sampling time is exactly the same b/w any two samples
3. Sequence of no. obtained by sampling is represented in exact precision.
4. Conversion of sequence to continuous time is ideal.

Aliasing

→ Analog sinusoid.
$x(t) = A\cos(2\pi f_0 t + \phi)$

→ Sampled at $T_s = 1/f_s$
$x[n] = x(T_s n) = A\cos(2\pi f_0 T_s n + \phi)$

→ Keeping the sampling period same, sample
$y(t) = A\cos(2\pi(f_0 + l f_s)t + \phi)$
where l is an integer
$y[n] = y(T_s n)$
$= A\cos(2\pi(f_0 + l f_s)T_s n + \phi)$
$= A\cos(2\pi f_0 T_s n + 2\pi l f_s T_s n + \phi)$

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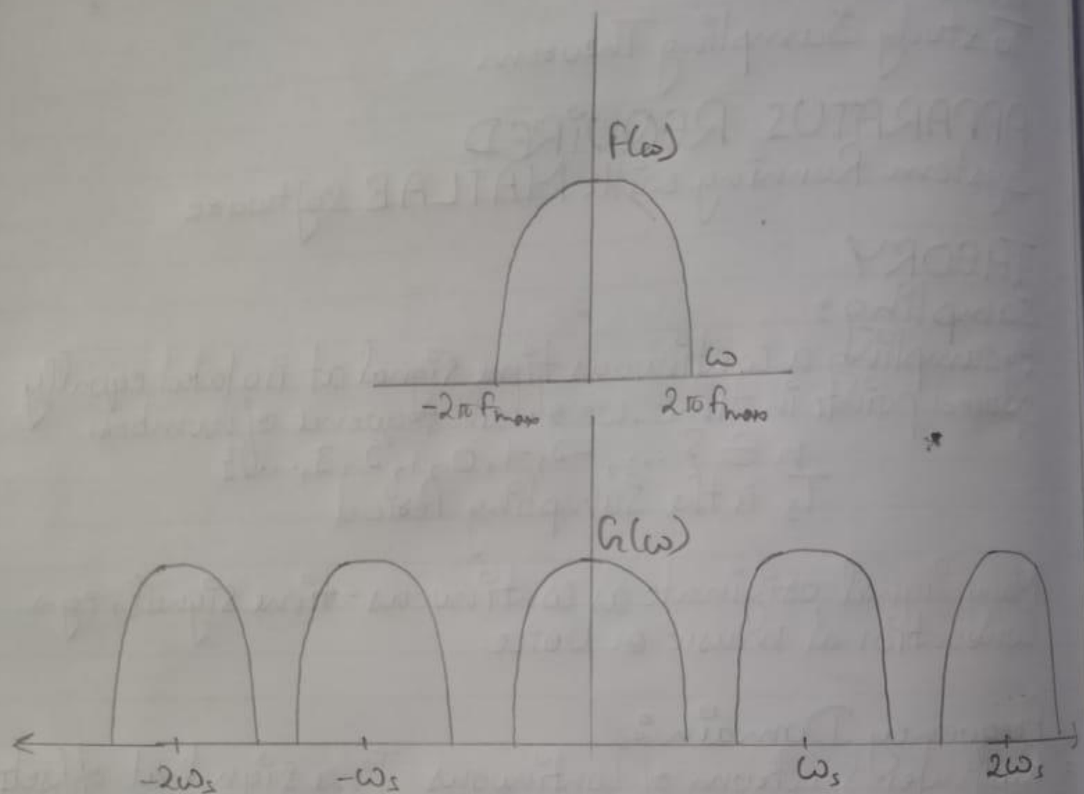

Fig:- Frequency Domains.

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$$= A \cos(2\pi f_0 T_s n + 2\pi l n + \phi)$$
$$= A \cos(2\pi f_0 T_s n + \phi)$$
$$= x[n]$$

Here, $F_s T_s = 1$

Since l is an integer

$\cos(n \pm 2\pi l) = \cos(n)$

$\Rightarrow$ $y[n]$ indistinguishable from $x[n]$.

CONCLUSION

Hence, we successfully studied about Sampling Theorem.

Teacher's Signature : ______

Program (Sampling Theorem)

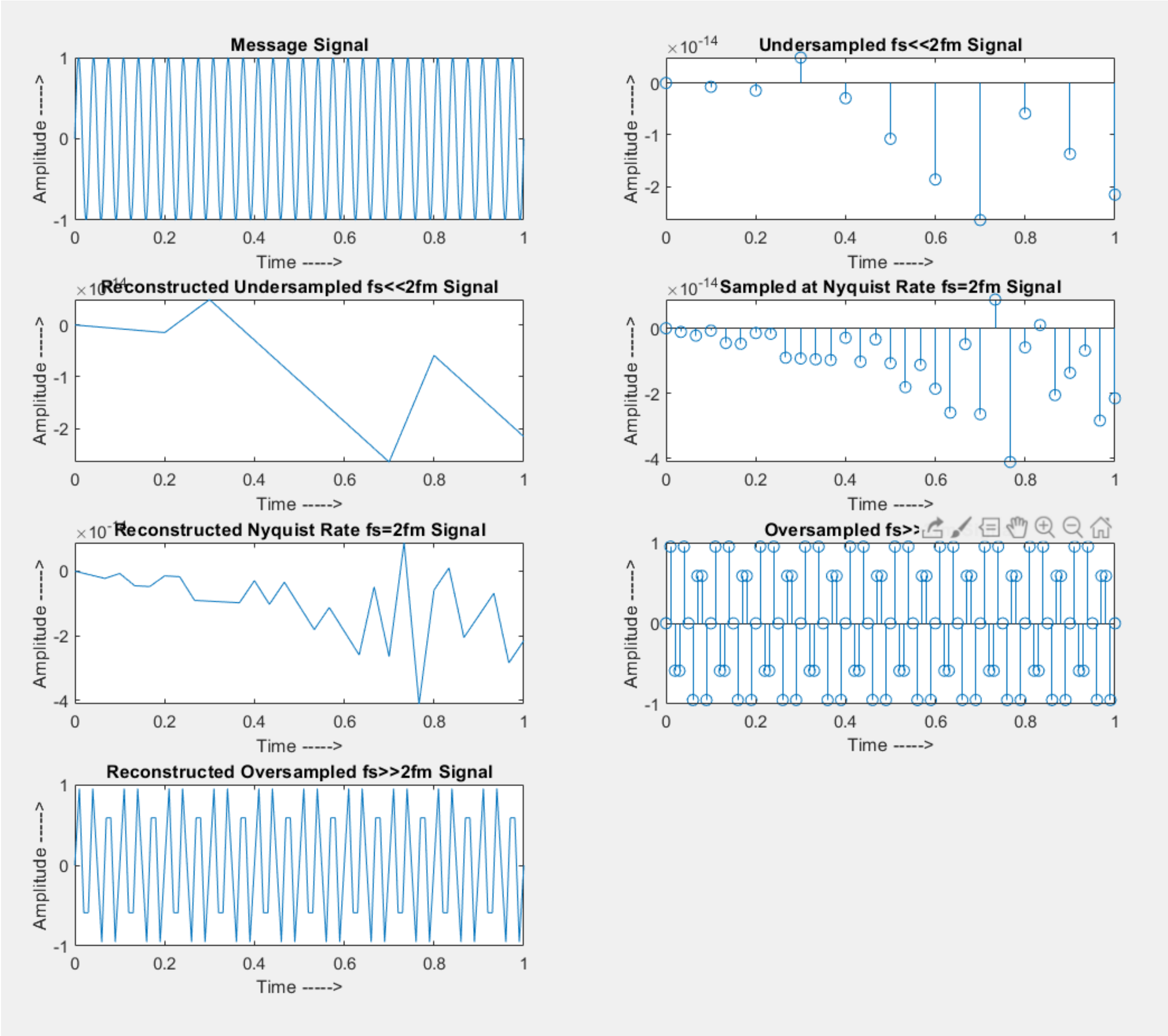
MATLAB PROGRAM TO IMPLEMENT SAMPLING THEOREM

```
t = 0:0.001:1;
fm = input ('Enter the modulating signal frequency = ');
x = sin(2*pi*fm*t);
subplot (4,2,1);
plot(t,x);
xlabel('Time ----->');
ylabel('Amplitude ----->');
title('Message Signal');
fs1 = input('Enter Sampling Frequency < Modulating Signal Frequency = ');
fs2 = input('Enter Sampling Frequency = Modulating Signal Frequency = ');
fs3 = input('Enter Sampling Frequency > Modulating Signal Frequency = ');

%Sampling at fs<<2fm
n = 0:1/fs1:1;
x1 = sin(2*pi*fm*n);
subplot(4,2,2);
stem(n,x1);
xlabel('Time ----->');
ylabel('Amplitude ----->');
title('Undersampled fs<<2fm Signal');
subplot(4,2,3);
plot(n,x1);
xlabel('Time ----->');
ylabel('Amplitude ----->');
title('Reconstructed Undersampled fs<<2fm Signal');

%Sampling at fs=2fm
n = 0:1/fs2:1;
x2 = sin(2*pi*fm*n);
subplot(4,2,4);
stem(n,x2);
xlabel('Time ----->');
ylabel('Amplitude ----->');
title('Sampled at Nyquist Rate fs=2fm Signal');
subplot(4,2,5);
plot(n,x2);
xlabel('Time ----->');
ylabel('Amplitude ----->');
title('Reconstructed Nyquist Rate fs=2fm Signal');

%Sampling at fs>>2fm
n = 0:1/fs3:1;
x3 = sin(2*pi*fm*n);
subplot(4,2,6);
stem(n,x3);
xlabel('Time ----->');
ylabel('Amplitude ----->');
title('Oversampled fs>>2fm Signal');
subplot(4,2,7);
plot(n,x3);
xlabel('Time ----->');
ylabel('Amplitude ----->');
title('Reconstructed Oversampled fs>>2fm Signal');
```


Message Signal
Undersampled fs<<2fm Signal
Reconstructed Undersampled fs<<2fm Signal
Sampled at Nyquist Rate fs=2fm Signal
Reconstructed Nyquist Rate fs=2fm Signal
Reconstructed Oversampled fs>>2fm Signal
Amplitude ----->
Time ----->

Practical 2

Date

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Page No.

AIM

To study Pulse Code Modulation (PCM)

THEORY

Pulse Code Modulation (PCM) is a method used to digitally represent sampled analog signals. It is the standard form for digital audio in computers and various Blu-ray, Compact Disc and DVD formats, as well as other uses such as digital telephone systems. A PCM stream is a digital representation of an analog signal, in which the magnitude of the analog signal is sampled regularly at uniform intervals, with each sample being quantized to the nearest value within a range of digital steps.

Basis of Pulse Code Modulation.

The three steps for developing an equivalent PCM digital signal from an analog signal are -

1. Sampling
2. Quantization
3. Coding

1. Sampling

The foundation of PCM is based on Nyquist Sampling Theorem. If a unlimited signal is sampled at regular intervals of time and at a rate equal to or higher than twice the highest significant signal frequency, then the sample contains all the information of the original signal. The original signal may then be reconstructed by use of a Low pass filter.

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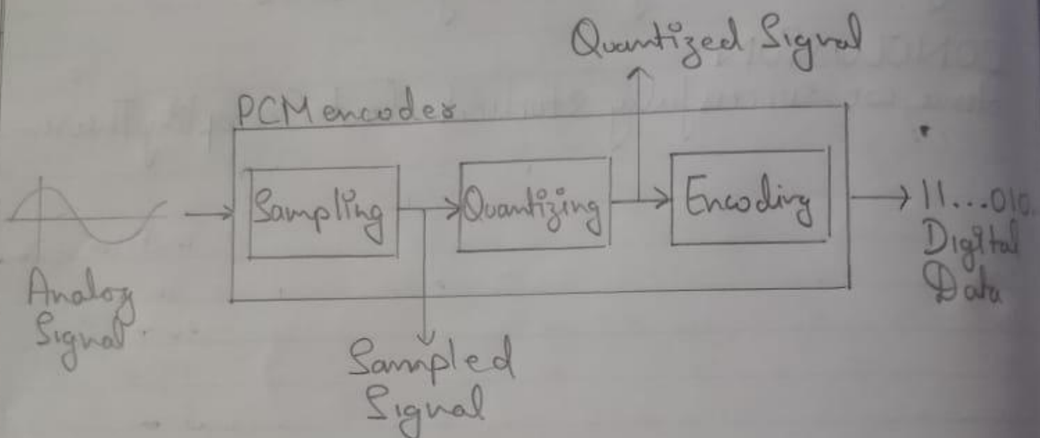

Fig: Block Diagram Pulse Code Modulation

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② Quantization.
The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value of a near stabilized value. Such process is called Quantization.

③ Coding
Older PCM system uses 7 bit code, and modern systems use an 8 bit code with its improved quantizing distortion performance. The companding and ~~expanding~~ coding is done together simultaneously. The compression and later expansion functions are logarithmic. A pseudologarithmic curve made up of linear segments imparts finer granularity to low level signals and less granularity to the higher-level signals.

Advantages of Pulse Code Modulation
1. Immune to channel induced noise and distortion.
2. Repeaters can be employed along the transmitting channel
3. Encoders allow secured data transmission.
4. It ensures uniform transmission quality

Disadvantages of Pulse Code Modulation.
1. Pulse code Modulation increases the transmission Bandwidth
2. A PCM system is somewhat more complex than other system.

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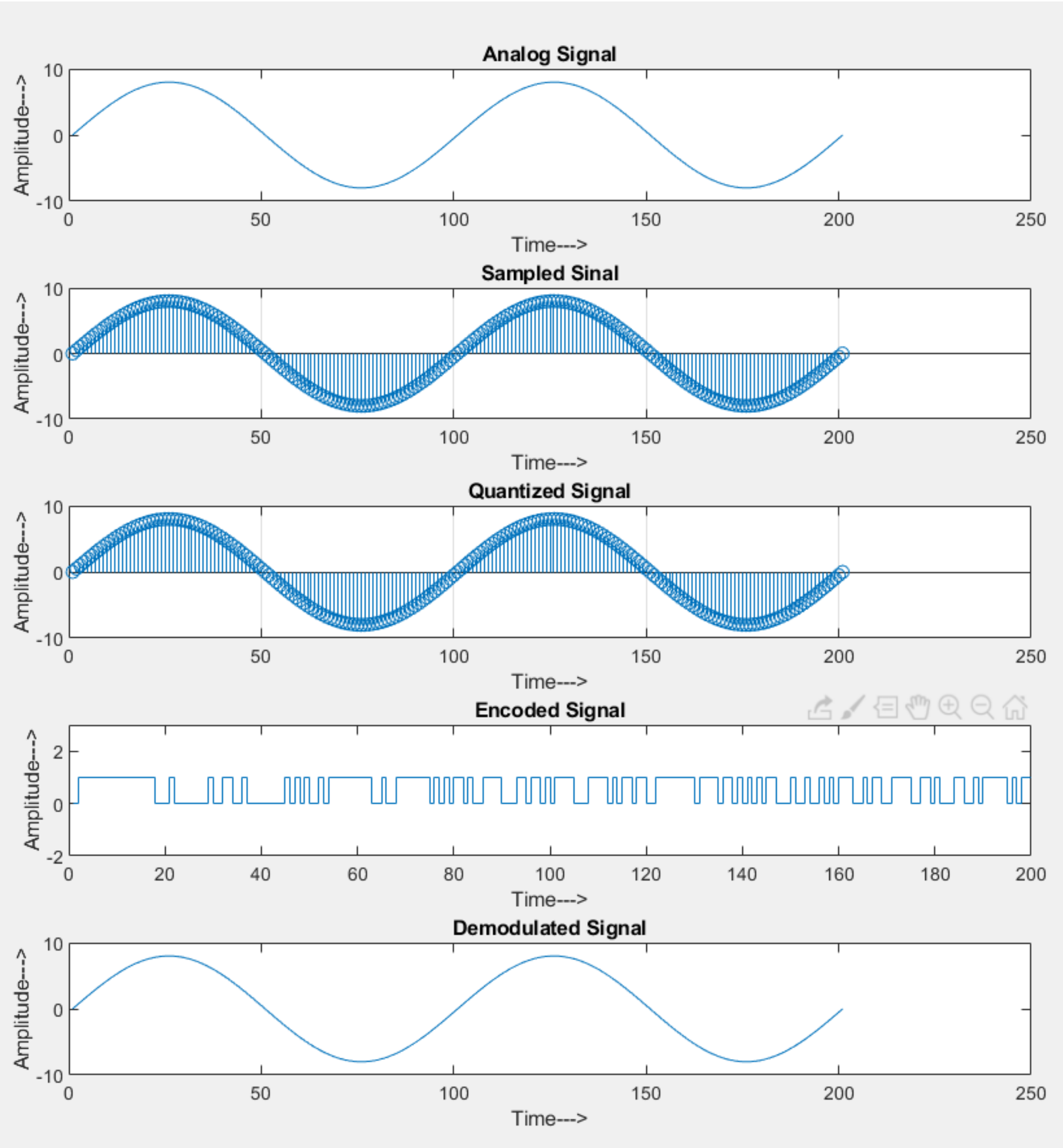
MATLAB PROGRAM TO IMPLEMENT PULSE CODE MODULATION AND DEMODULATION

```
n=input('Enter n value for n-bit PCM system : ');
n1=input('Enter number of samples in a period : ');
L=2^n;

% Sampling Operation
x=0:2*pi/n1:4*pi;
s=8*sin(x);
subplot(5,1,1);
plot(s);
title('Analog Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(5,1,2);
stem(s);grid on; title('Sampled Sinal'); ylabel('Amplitude--->'); xlabel('Time--->');

% Quantization Process
vmax=8;
vmin=-vmax;
del=(vmax-vmin)/L;
part=vmin:del:vmax;
code=vmin-(del/2):del:vmax+(del/2);
[ind,q]=quantiz(s,part,code);
l1=length(ind);
l2=length(q);
for i=1:l1
if(ind(i)~=0)
ind(i)=ind(i)-1;
end
i=i+1;
end
for i=1:l2
if(q(i)==vmin-(del/2))
q(i)=vmin+(del/2);
end
end
subplot(5,1,3);
stem(q);grid on;
title('Quantized Signal');
ylabel('Amplitude--->');
xlabel('Time--->');

% Encoding Process
code=de2bi(ind,'left-msb');
k=1;
for i=1:l1
for j=1:n
coded(k)=code(i,j);
j=j+1;
k=k+1;
end
i=i+1;
end
subplot(5,1,4); grid on;
```

```
stairs(coded);
axis([0 2*n1 -2 3]); title('Encoded Signal');
ylabel('Amplitude--->');
xlabel('Time--->');

% Demodulation Of PCM signal
qunt=reshape(coded,n,length(coded)/n);
index=bi2de(qunt','left-msb');
q=del*index+vmin+(del/2);
subplot(5,1,5); grid on;
plot(q);
title('Demodulated Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
```



Practical 3

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AIM

To study Delta Modulation (DM).

APPARATUS

System running MATLAB software

THEORY

Delta Modulation is an Analog to Digital and Digital to Analog Signal Conversion Technique used for Transmission of voice information where quality is not of primary importance. DM is the simplest form of Differential Pulse-Code Modulation (DPCM) where the difference b/w successive samples are encoded into n-bit data streams. In delta modulation, the transmitted data are reduced to 1-bit stream. It's main features are—

- The analog signal is approximated with a series of segment
- Each segment of the approximated signal is compared to the preceding bits and the successive bits are determined by this comparison
- Only the change of information is sent, that is, only an increase or decrease of the signal amplitude from the previous sample is sent whereas a no-change condition causes the modulated signal to remain at the same 0 or 1 state of the previous sample.

To achieve high SNR ratio, delta modulation must use over sampling techniques, that is, the analog signal is

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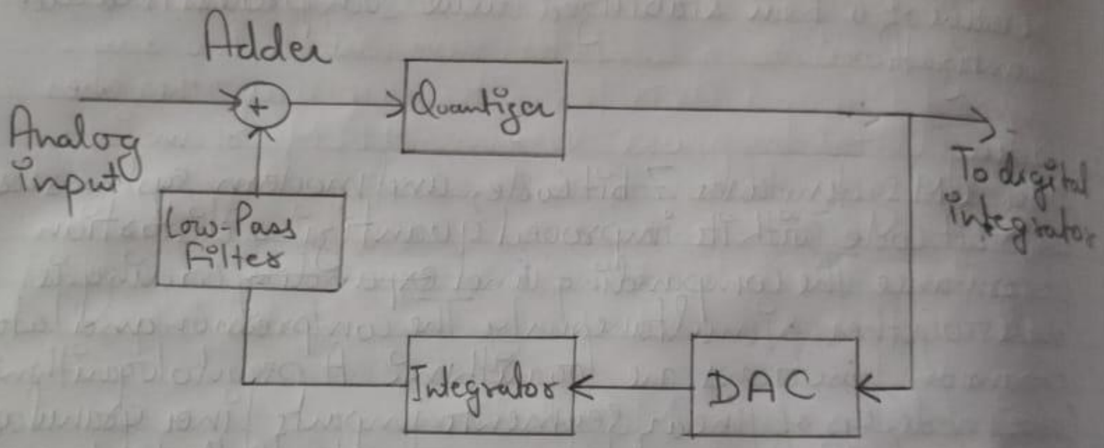


Fig: Block Diagram Delta Modulation

Sampled at a rate several times higher than Nyquist rate

Delta Modulation

Delta Modulation comprises of 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of delta modulator. The predictor circuit in DPCM is replaced by a simple delay circuit in DM. A staircase approximated waveform will be output of the delta modulator with the step-size as delta (Δ). The output quality of waveform is moderate.

Delta Demodulator

The delta demodulator comprises a low pass filter, a summer and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

A binary sequence will be given as a input to the demodulator. The staircase approximated output is given to the LPF. Low pass filters are used for many reasons, but the prominent reason is noise elimination for out of bands signals. The step-size error that may occur at the transmitter is called granular noise, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

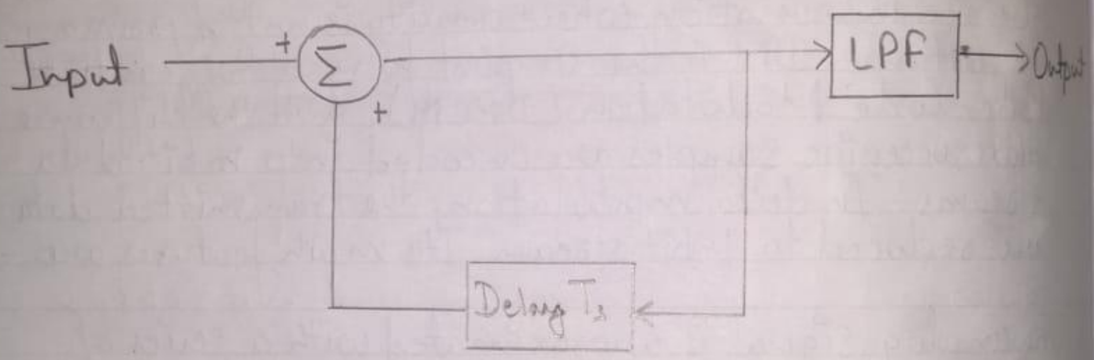
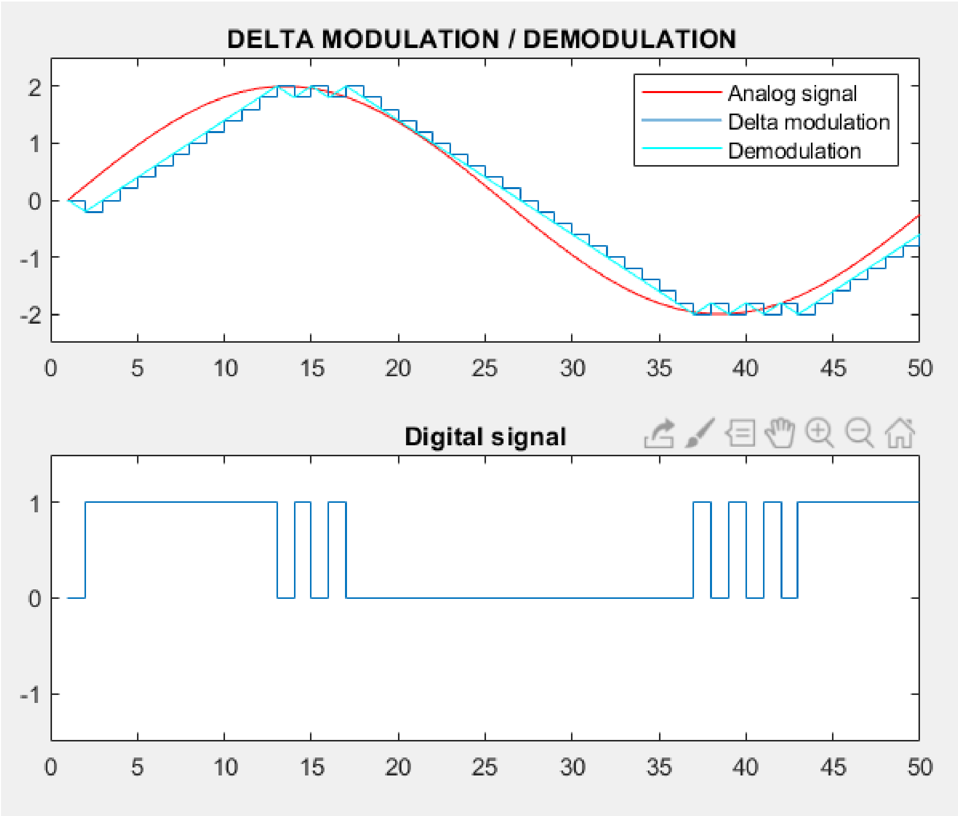


fig → Delta Demodulator Block Diagram

MATLAB PROGRAM TO IMPLEMENT DELTA MODULATION AND DEMODULATION

```
a=2;
t=0:2*pi/50:2*pi;
x=a*sin(t);
l=length(x);
subplot(2,1,1);
plot(x,'r');
delta=0.2;
hold on
xn=0;
for i=1:l;
if x(i)>xn(i)
d(i)=1;
xn(i+1)=xn(i)+delta;
else
d(i)=0; xn(i+1)=xn(i)-delta;
end
end
stairs(xn)
if d(i)==0
xn(i+1)=xn(i)-delta;
else
xn(i+1)=xn(i)+delta;
end
plot(xn,'c');
ylim([-2.5,2.5]);
xlim([0,50]);
legend('Analog signal','Delta modulation','Demodulation')
title('DELTA MODULATION / DEMODULATION ');
subplot(2,1,2);
stairs(d);
ylim([-1.5,1.5]);
xlim([0,50]);
title('Digital signal');
```


DELTA MODULATION / DEMODULATION
Analog signal
Delta modulation
Demodulation
Digital signal

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AIM

To generate and demodulate amplitude shift keyed (ASK) signal using MATLAB

APPARATUS

System Running MATLAB

THEORY

Generation of ASK

Amplitude shift keying - ASK - is a modulation process, which imparts to a sinusoid two or more discrete amplitude levels. These are related to the number of levels adopted by the digital message. For a binary message sequence there are two levels, one of which is typically zero. The data rate is a sub multiple of the carrier frequency. Thus the modulated waveform consists of bursts of a sinusoid. One of the disadvantages of ASK, compared with FSK and PSK, for example, is that it has not got a constant envelope. This makes its processing (eg, power amplification) more difficult, since linearity becomes an important factor. However, it does make for ease of demodulation with an envelope detector.

Demodulation.

ASK signal has a well defined envelope. Thus it enable to demodulation by an envelope detector. Some sort of decision-making circuitry is necessary for detecting the message. The signal is recovered by using a correlator

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and decision making circuitry is used to recover the binary sequence.

ASK modulation

1. Generate Carrier Signal
2. Start FOR loop
3. Generate Binary Data, Message Signal (on-off form)
4. Generate ASK modulated Signal
5. Plot message signal and ASK modulated signal
6. End FOR loop.
7. Plot the binary data and carrier

ASK demodulation.

1. Start FOR loop
2. Perform Correlation of ASK signal with carrier to get decision variable
3. Make decision to get demodulated Binary data. If $x>0$ choose 1 else choose 0
4. Plot the demodulated Binary data.

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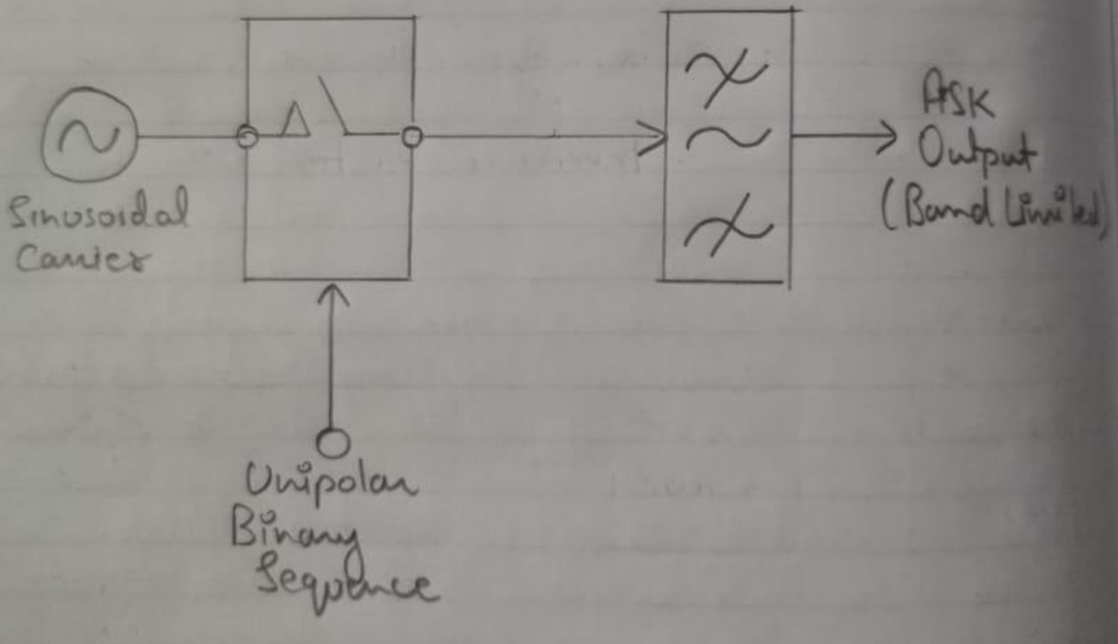
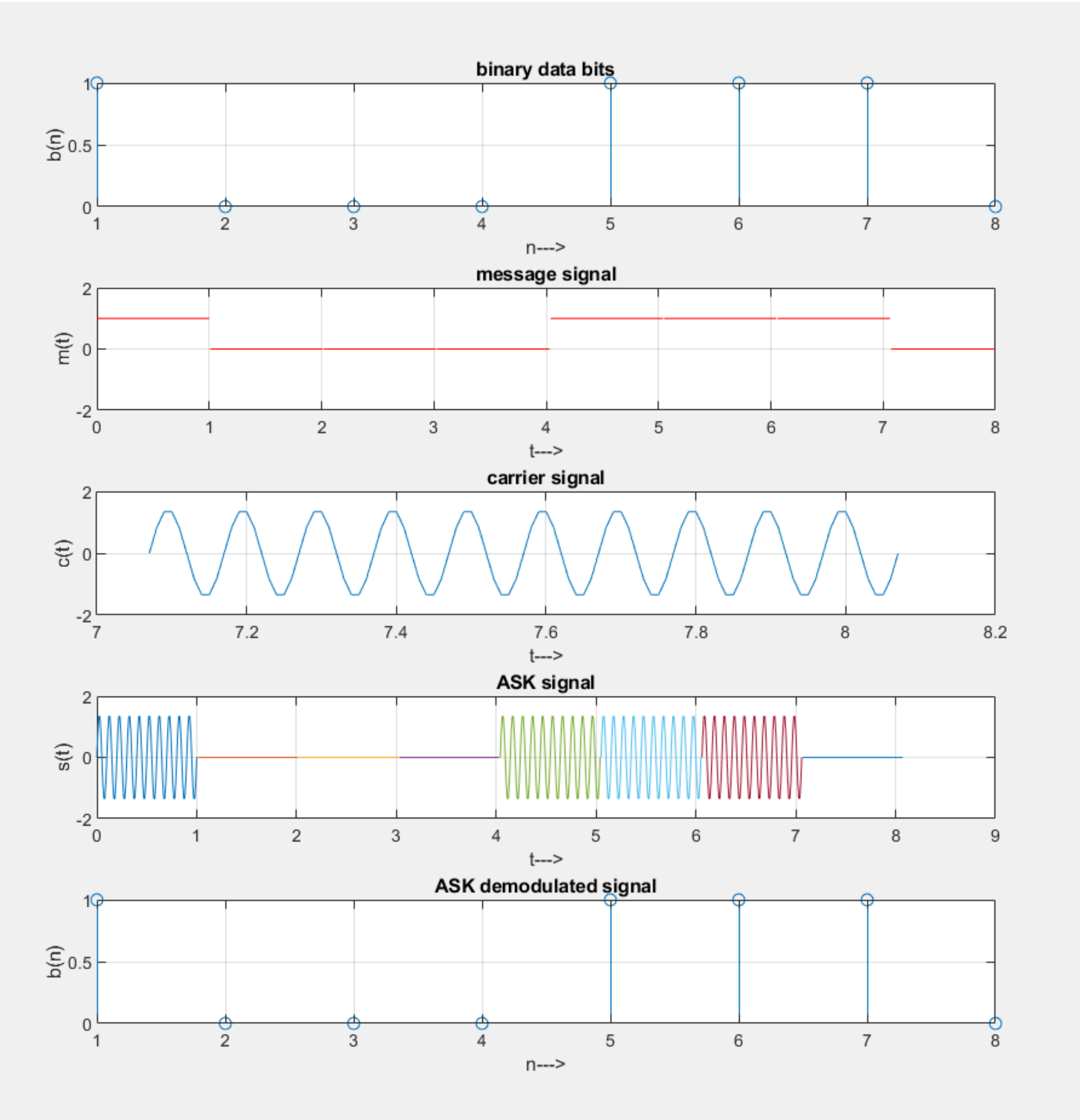


Fig:- ASK generation method

MATLAB PROGRAM TO IMPLEMENT ASK MODULATION AND DEMODULATION

```
%ASK Modulation

%GENERATE CARRIER SIGNAL
Tb=1; fc=10;
t=0:Tb/100:1;
c=sqrt(2/Tb)*sin(2*pi*fc*t);

%generate message signal
N=8;
m=rand(1,N);
t1=0;
t2=Tb;
for i=1:N
t=[t1:.01:t2];
if m(i)>0.5
m(i)=1;
m_s=ones(1,length(t));
else
m(i)=0;
m_s=zeros(1,length(t));
end
message(i,:)=m_s;

%product of carrier and message
ask_sig(i,:)=c.*m_s;
t1=t1+(Tb+.01);
t2=t2+(Tb+.01);

%plot the message and ASK signal
subplot(5,1,2);axis([0 N -2 2]);plot(t,message(i,:),'r');
title('message signal');xlabel('t--->');ylabel('m(t)');grid on
hold on
subplot(5,1,4);plot(t,ask_sig(i,:));
title('ASK signal');xlabel('t--->');ylabel('s(t)');grid on
hold on
end
hold off

%Plot the carrier signal and input binary data
subplot(5,1,3);plot(t,c);
title('carrier signal');xlabel('t--->');ylabel('c(t)');
grid on
subplot(5,1,1);stem(m);
title('binary data bits');xlabel('n--->');ylabel('b(n)');
grid on


% ASK Demodulation
t1=0;t2=Tb;
for i=1:N
t=[t1:Tb/100:t2]

%correlator
x=sum(c.*ask_sig(i,:));

%decision device
if x>0
```

```
demod(i)=1;
else
demod(i)=0;
end
t1=t1+(Tb+.01);
t2=t2+(Tb+.01);
end

%plot demodulated binary data bits
subplot(5,1,5);stem(demod);
title('ASK demodulated signal'); xlabel('n--->');ylabel('b(n)');
grid on;
```



Practical 5

Date

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AIM

To generate and demodulate phase shift keyed (PSK) signal using MATLAB

APPARATUS

System Running MATLAB

THEORY

Generation of PSK signal

PSK is a digital modulation scheme that conveys data by changing, or modulating, the phase of a reference signal (the carrier wave). PSK uses a finite number of waves phases, each assigned a unique pattern of binary digits. Usually, each phase encodes an equal number of bits. Each pattern of bits forms the symbol that is represented by the particular phase. The demodulator, which is designed specifically for the symbol-set used by the modulator, determines the phase of the received signal and maps it back to the symbol it represents, thus recovering the original data.

In a coherent binary PSK system, the pair of signal $S_1(t)$ and $S_2(t)$ used to represent binary symbols 1 & 0 are defined by

$$S_1(t) = \sqrt{2E_b/T_b}\ \cos 2\pi f_c t$$

$$S_2(t) = \sqrt{2E_b/T_b}\ (2\pi f_c t + \pi)$$

$$= -\sqrt{2E_b/T_b}\ \cos 2\pi f_c t$$

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where $0 \leq t < T_b$

E_b = Transmitted Signal Energy for bit

The Carrier Frequency $f_c = n/T_b$ for some fixed integer n

PSK Modulation

1. Generate Carrier Signal
2. Start FOR loop
3. Generate Binary data, message signal in polar form
4. Generate PSK modulated signal.
5. Plot message signal and PSK modulated signal.
6. End FOR loop.
7. Plot the binary data and Carrier.

PSK Demodulation.

1. Start FOR loop,
   Perform Correlation of PSK signal with carrier to get decision variable
2. Make decision to get demodulated binary data. If $n > 0$, choose '1' else choose '0'
3. Plot the demodulated Binary data.

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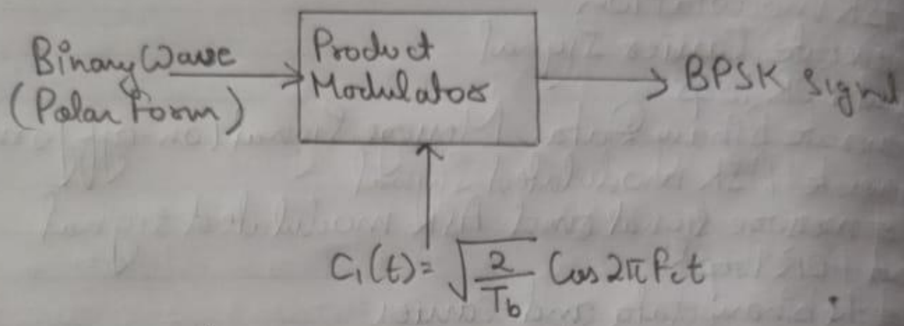


fig: BPSK Transmitter

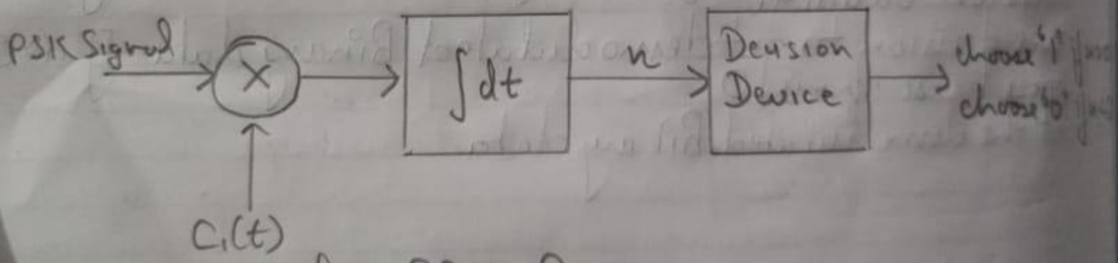
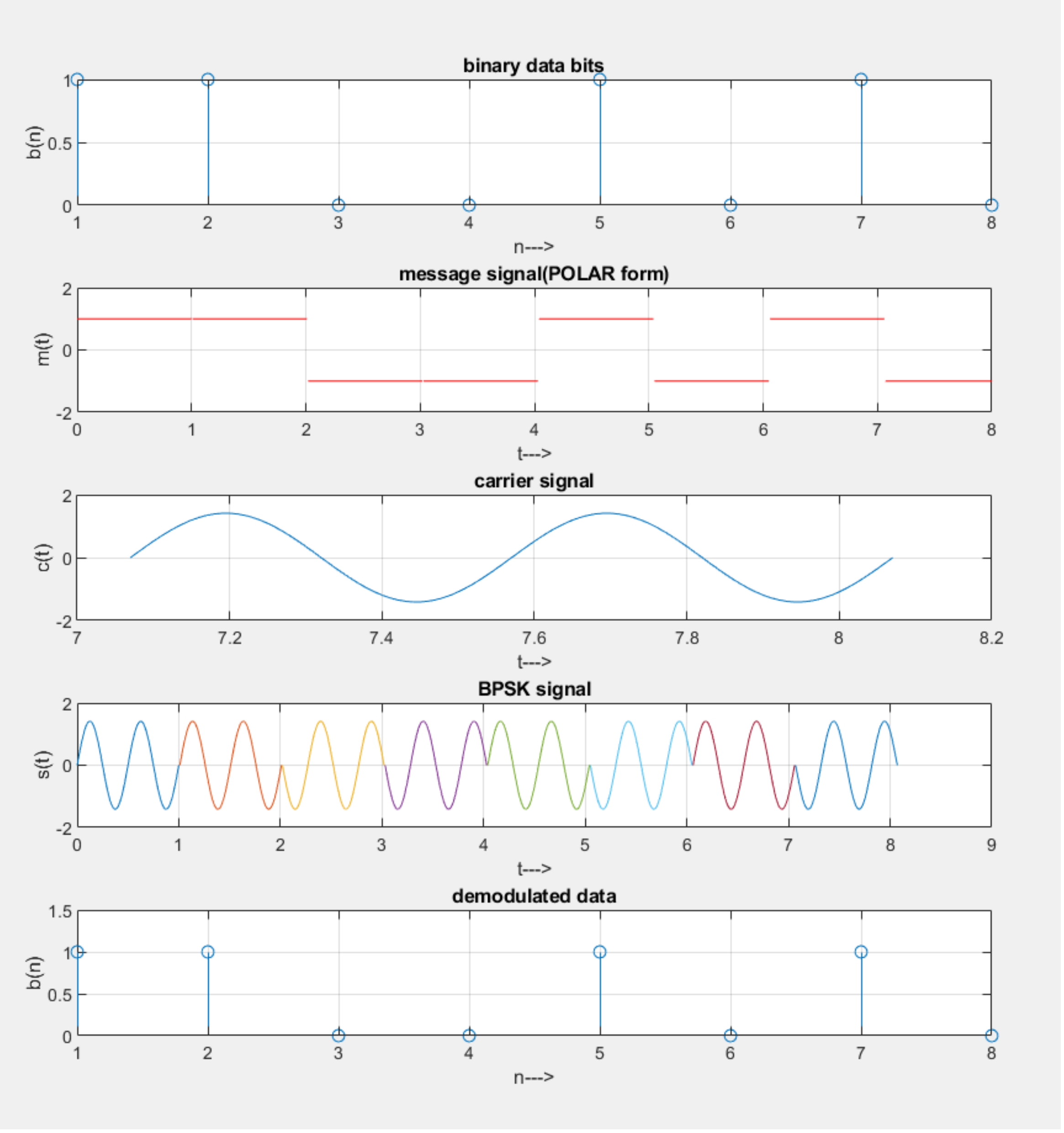


fig: BPSK Receiver

MATLAB PROGRAM TO IMPLEMENT PSK MODULATION AND DEMODULATION

```
% PSK modulation

%GENERATE CARRIER SIGNAL
Tb=1;
t=0:Tb/100:Tb;
fc=2;
c=sqrt(2/Tb)*sin(2*pi*fc*t);

%generate message signal
N=8;
m=rand(1,N);
t1=0;t2=Tb;
for i=1:N
t=[t1:.01:t2];
if m(i)>0.5
m(i)=1;
m_s=ones(1,length(t));
else
m(i)=0;
m_s=-1*ones(1,length(t));
end
message(i,:)=m_s;

%product of carrier and message signal
bpsk_sig(i,:)=c.*m_s;

%Plot the message and BPSK modulated signal
subplot(5,1,2);axis([0 N -2 2]);plot(t,message(i,:),'r');
title('message signal(POLAR form)');xlabel('t--->');ylabel('m(t)');
grid on; hold on;
subplot(5,1,4);plot(t,bpsk_sig(i,:));
title('BPSK signal');xlabel('t--->');ylabel('s(t)');
grid on; hold on;
t1=t1+1.01; t2=t2+1.01;
end
hold off

%plot the input binary data and carrier signal
subplot(5,1,1);stem(m);
title('binary data bits');xlabel('n--->');ylabel('b(n)');
grid on;
subplot(5,1,3);plot(t,c);
title('carrier signal');xlabel('t--->');ylabel('c(t)');
grid on;
10

% PSK Demodulation
t1=0;t2=Tb;
for i=1:N
t=[t1:.01:t2];

%correlator
x=sum(c.*bpsk_sig(i,:));

%decision device
if x>0
    demod(i)=1;
```

```
else
demod(i)=0;
end
t1=t1+1.01;
t2=t2+1.01;
end

%plot the demodulated data bits
subplot(5,1,5);
stem(demod);
ylim([0,1.5])
title('demodulated data');xlabel('n--->');ylabel('b(n)');
grid on
```



Practical 6

Date

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AIM

To generate and demodulate frequency Shift Keyed (FSK) signal using MATLAB

APPARATUS

System running MATLAB software

THEORY

Generation of FSK

Frequency Shift Keying (FSK) is a frequency modulation scheme in which digital information is transmitted through discrete frequency changes of a carrier wave. The simplest FSK is binary FSK (BFSK). BFSK uses a pair of discrete frequencies to Transmit Binary (0s and 1s) information. With this scheme, the "1" is called the mark frequency and "0" is called the space frequency.

In binary FSK system, symbols 1 & 0 are distinguished from each other by transmitting one of the two sinusoidal waves that differ in frequency by a fixed amount.

$$S_i(t) = \sqrt{\frac{2E_b}{T_b}} \cos 2\pi f_i t$$

$$0 \le t \le T_b$$

E_b = Transmitted Energy / bit

Transmitted frequency $= f_i = (n_c + i)/T_b$

n = constant (integer)

T_b = bit interval.

Teacher's Signature :

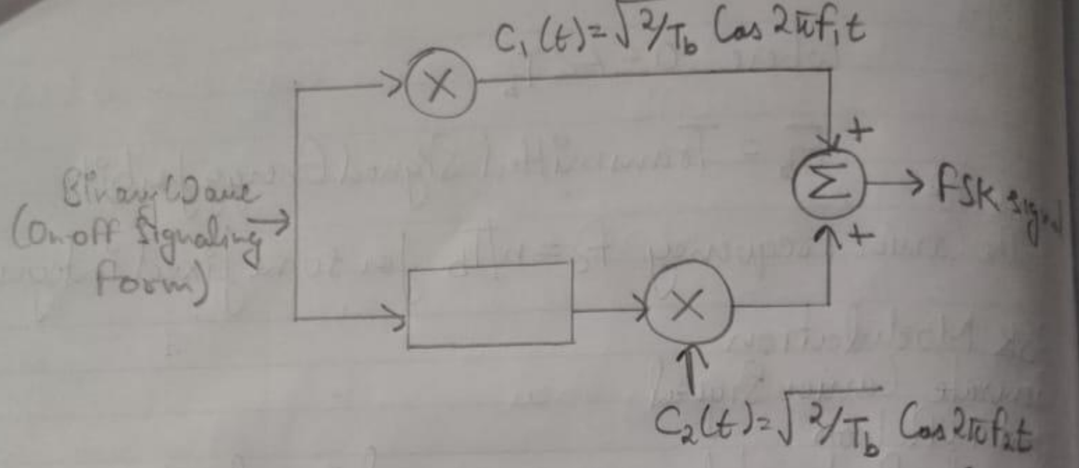

Fig → BFSK Transmitter

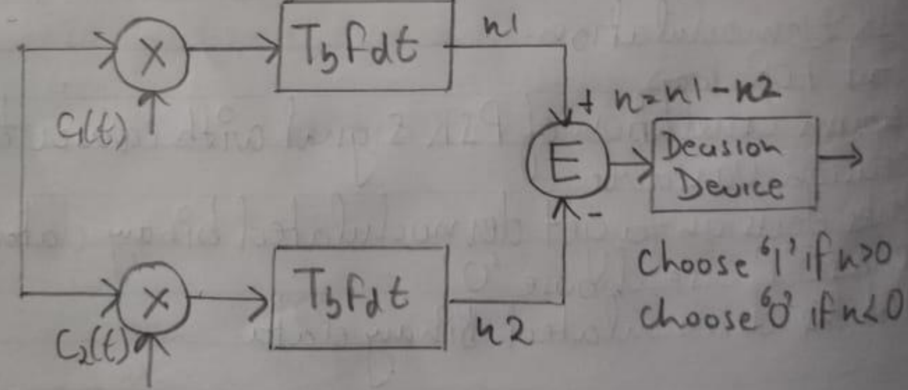

Fig : BFSK Receiver

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FSK modulation.
1. Generate two Carrier signals
2. Start FOR loop
3. Generate Binary data, message signal and inverted message signal
4. Multiply carrier1 with message signal and carrier 2 with inverted message signal
5. Perform addition to get FSK modulated signal.
6. Plot message signal and FSK modulated signal
7. End FOR loop.
8. Plot the Binary data and Carriers.

FSK demodulation
1. Start FOR loop
2. Perform Correlation of FSK modulated signal with carrier 1 and carrier 2 to get two decisions variables x1 and x2
3. Make decision x = x1 - x2 to get demodulated binary data. If x>0 choose '1' else choose '0'
4. Plot the demodulated Binary data.

Teacher's Signature :

MATLAB PROGRAM TO IMPLEMENT FSK MODULATION AND DEMODULATION

```
% FSK Modulation

%GENERATE CARRIER SIGNAL
Tb=1; fc1=2;fc2=5;
t=0:(Tb/100):Tb;
c1=sqrt(2/Tb)*sin(2*pi*fc1*t);
c2=sqrt(2/Tb)*sin(2*pi*fc2*t);

%generate message signal
N=8;
m=rand(1,N);
t1=0;t2=Tb;
for i=1:N
t=[t1:(Tb/100):t2];
if m(i)>0.5
m(i)=1;
m_s=ones(1,length(t));
invm_s=zeros(1,length(t));
else
m(i)=0;
m_s=zeros(1,length(t));
invm_s=ones(1,length(t));
end
message(i,:)=m_s;

%Multiplier
fsk_sig1(i,:)=c1.*m_s;
fsk_sig2(i,:)=c2.*invm_s;
fsk=fsk_sig1+fsk_sig2;

%plotting the message signal and the modulated signal
subplot(3,2,2);axis([0 N -2 2]);
plot(t,message(i,:),'r');
title('message signal');
xlabel('t---->');
ylabel('m(t)');
grid on;
hold on;
subplot(3,2,5);
plot(t,fsk(i,:));
title('FSK signal');
xlabel('t---->');
ylabel('s(t)');
grid on;hold on;
t1=t1+(Tb+.01);
t2=t2+(Tb+.01);
end
hold off

%Plotting binary data bits and carrier signal
subplot(3,2,1);stem(m);
title('binary data');
xlabel('n---->');
ylabel('b(n)');
grid on;
subplot(3,2,3);
plot(t,c1);
title('carrier signal-1');
```

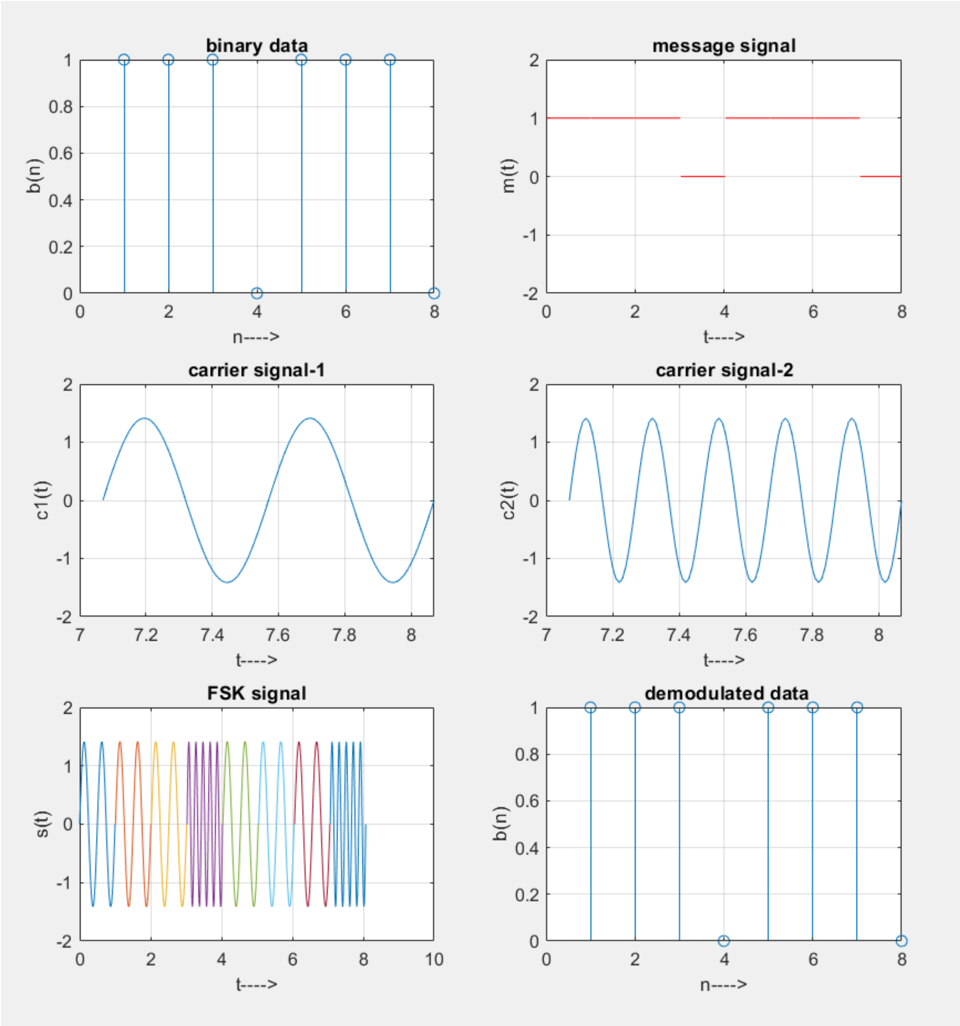
```
xlabel('t---->');
ylabel('c1(t)');
grid on;
subplot(3,2,4);
plot(t,c2);
title('carrier signal-2');
xlabel('t---->');
ylabel('c2(t)');
grid on;


% FSK Demodulation
t1=0;
t2=Tb;
for i=1:N
t=[t1:(Tb/100):t2];

%correlator
x1=sum(c1.*fsk_sig1(i,:));
x2=sum(c2.*fsk_sig2(i,:));
x=x1-x2;

%decision device
if x>0
demod(i)=1;
else
demod(i)=0;
end
t1=t1+(Tb+.01);
t2=t2+(Tb+.01);
end

%Plotting the demodulated data bits
subplot(3,2,6);
stem(demod);
title(' demodulated data');
xlabel('n---->');
ylabel('b(n)');
 grid on;
```


binary data
b(n)
n---->
message signal
m(t)
t---->
carrier signal-1
c1(t)
t---->
carrier signal-2
c2(t)
t---->
FSK signal
s(t)
t---->
demodulated data
b(n)
n---->

Practical 7

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AIM

To generate and demodulate quadrature phase shifted (QPSK) signal using MATLAB

THEORY

Generation of Quadrature PSK signal

QPSK is also known as quaternary PSK, quadriphase PSK, 4-PSK, or 4-QAM. It is a phase modulation technique that transmits two bits in four modulation states.

Phase of the carrier takes on one of the four equally spaced value such as $\frac{\pi}{4}$, $\frac{3\pi}{4}$, $\frac{5\pi}{4}$ and $\frac{7\pi}{4}$

$$S_i(t) = \sqrt{\frac{2E}{T}} \cos(2\pi f_c t + (2i-1)\pi/4), 0 \le t \le T$$

$$= 0$$

where $i = 1, 2, 3, 4,$

E = Tx signal energy per symbol

T = symbol duration.

Each of the possible value of phase corresponds to a pair of bits called dibits. Thus, the gray encoded sets of digits 10, 00, 01, 11

$$S_i(t) = \sqrt{\frac{2E}{T}} \cos[(2i-1)\pi/4] \cos(2\pi f_c t) -$$

$$\sqrt{\frac{2E}{T}} \sin\left[(2i-1)\frac{\pi}{4}\right] \sin(2\pi f_c t)$$

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for $0 \le t \le T_b$

$= 0$, elsewhere

There are two orthonormal basis functions

$$c_1(t) = \sqrt{\frac{2}{T}} \cos 2\pi f_c t \qquad 0 \le t \le T_b$$

$$c_2(t) = \sqrt{\frac{2}{T}} \sin 2\pi f_c t \qquad 0 \le t \le T_b$$

The I/p binary sequence b(t) is represented in polar form with symbols 1 & 0 represented as $+\sqrt{\frac{E}{2}}$ and $-\sqrt{\frac{E}{2}}$. This binary wave is demultiplexed into 2 two separate binary waves consisting of odd & even numbered I/P bits denoted by $b_1(t)$ & $b_2(t)$. $b_1(t)$ & $b_2(t)$ are used to modulate a pair of quadrature carrier. The result is two PSK waves. These two binary PSK waves are added to produce the desired QPSK signal.

QPSK receiver consists of a pair of correlators with common I/P and supplied with locally generated signal $c_1(t)$ & $c_2(t)$. The correlator output, x_1 & x_2 are each compared with a threshold of 0 volts. If $x_1 > 0$, decision is made in favour of symbol '1' for upper channel. If $x_1 < 0$, decision is made in favour of symbol 1 for lower channel. Parallelly If $x_2 > 0$, decision is made in favour of symbol 1 for lower channel & if $x_2 < 0$, decision is made in favour of symbol 0. These two channels are combined in a multiplexer to get the

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Original binary output.

QPSK modulation

1. Generate quadrature carriers
2. Start FOR loop
3. Generate binary data, message signal (bipolar form)
4. Multiply carrier 1 with odd bits of message signal and carrier 2 with even bits of message signal
5. Perform addition of odd and even modulated signals to get the QPSK modulated signal.
6. Plot QPSK modulated Signal.
7. End FOR loop.
8. Plot the binary data and carriers.

QPSK demodulation.

1. Start FOR loop.
2. Perform Correlation of QPSK modulated signal with quadrature carriers to get two decision variable n1 and n2.
3. Make decision on n1 & n2 and multiplexed to get demodulated Binary data.
   If n1>0 and n2>0, choose '11'
   If n1>0 and n2<0, choose '10'
   If n1<0 and n2>0, choose '01'
   If n1<0 and n2<0, choose '00'
4. End FOR loop.
5. Plot demodulated data.

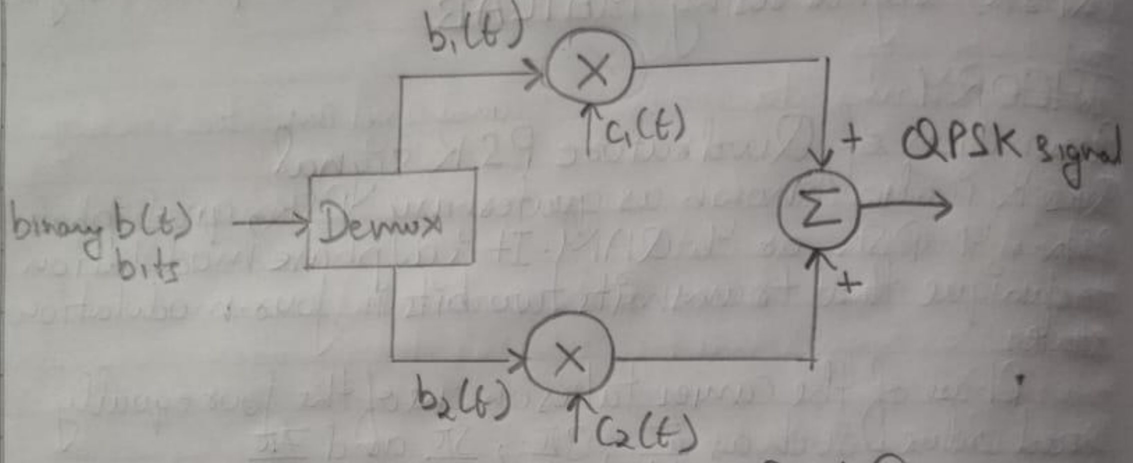
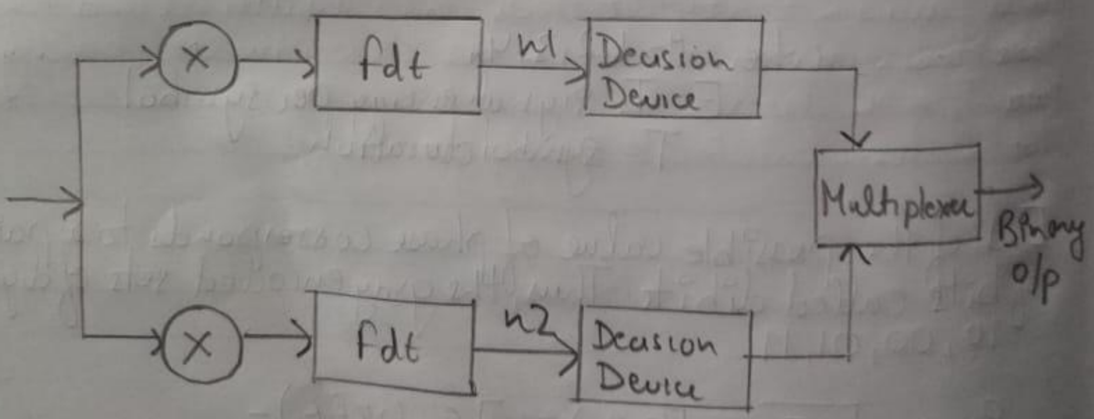


fig: QPSK Transmitter Block Diagram



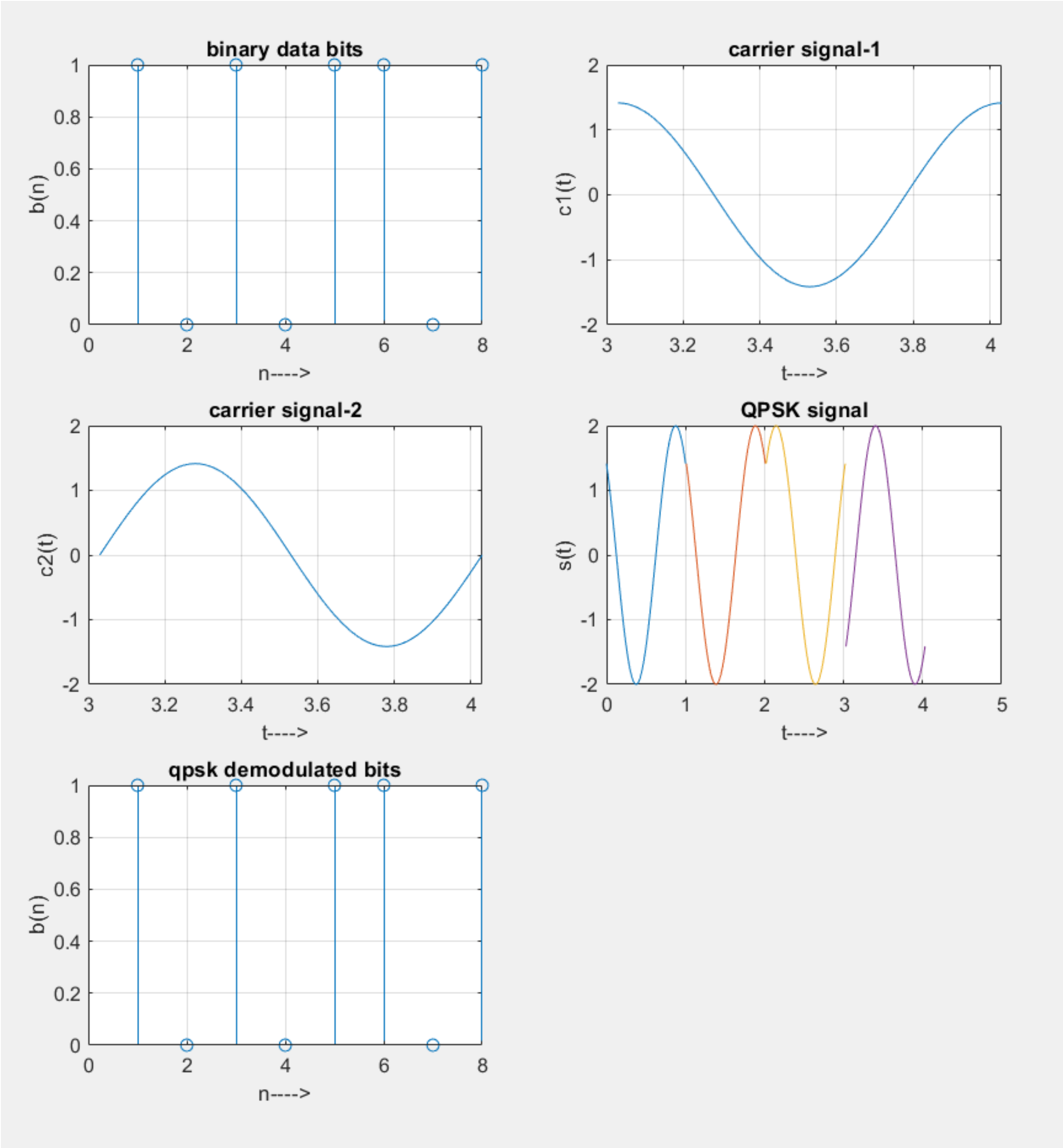
Input Debits	Phase of QPSK Signal	Co-ordinates of msg Signals	
		S1	S2
10	$\pi/4$	$\sqrt{E/2}$	$-\sqrt{E/2}$
00	$3\pi/4$	$-\sqrt{E/2}$	$-\sqrt{E/2}$
01	$5\pi/4$	$-\sqrt{E/2}$	$+\sqrt{E/2}$
11	$7\pi/4$	$+\sqrt{E/2}$	$+\sqrt{E/2}$

Four message Points of QPSK

MATLAB PROGRAM TO IMPLEMENT QPSK MODULATION AND DEMODULATION

```
% QPSK Modulation

%GENERATE QUADRATURE CARRIER SIGNAL
Tb=1;t=0:(Tb/100):Tb;fc=1;
c1=sqrt(2/Tb)*cos(2*pi*fc*t);
c2=sqrt(2/Tb)*sin(2*pi*fc*t);

%generate message signal
N=8;m=rand(1,N);
t1=0;t2=Tb;
for i=1:2:(N-1)
t=[t1:(Tb/100):t2];
if m(i)>0.5
m(i)=1;
m_s=ones(1,length(t));
else
m(i)=0;
m_s=-1*ones(1,length(t));
end

%odd bits modulated signal
odd_sig(i,:)=c1.*m_s;
if m(i+1)>0.5
%21
m(i+1)=1;
m_s=ones(1,length(t));
else
m(i+1)=0;
m_s=-1*ones(1,length(t));
end

%even bits modulated signal
even_sig(i,:)=c2.*m_s;

%qpsk signal
qpsk=odd_sig+even_sig;

%Plot the QPSK modulated signal
subplot(3,2,4);
plot(t,qpsk(i,:));
title('QPSK signal');
xlabel('t---->');
ylabel('s(t)');
grid on;
hold on;
t1=t1+(Tb+.01);
t2=t2+(Tb+.01);
end
hold off

%Plot the binary data bits and carrier signal
subplot(3,2,1);
stem(m);
title('binary data bits');
xlabel('n---->');
ylabel('b(n)');
grid on;
```

```matlab
subplot(3,2,2);
plot(t,c1);
title('carrier signal-1');
xlabel('t---->');
ylabel('c1(t)');
grid on;
subplot(3,2,3);
plot(t,c2);
title('carrier signal-2');
xlabel('t---->');
ylabel('c2(t)');
grid on;

% QPSK Demodulation
t1=0;t2=Tb;
for i=1:N-1
t=[t1:(Tb/100):t2]

%correlator
x1=sum(c1.*qpsk(i,:));
x2=sum(c2.*qpsk(i,:));

%decision device
if (x1>0&&x2>0)
demod(i)=1;
demod(i+1)=1;
elseif (x1>0&&x2<0)
demod(i)=1;
demod(i+1)=0;
elseif (x1<0&&x2<0)
demod(i)=0;
demod(i+1)=0;
elseif (x1<0&&x2>0)
demod(i)=0;
demod(i+1)=1;
end
t1=t1+(Tb+.01);
t2=t2+(Tb+.01);
end
subplot(3,2,5);
stem(demod);
title('qpsk demodulated bits');
xlabel('n---->');
ylabel('b(n)');
grid on;
```



binary data bits
b(n)
n---->
carrier signal-1
c1(t)
t---->
carrier signal-2
c2(t)
t---->
QPSK signal
s(t)
t---->
qpsk demodulated bits
b(n)
n---->

Practical 8

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AIM

To study about Delta Pulse Code Modulation and Demodulation of a signal and to Calculate the S/N Ratio.

APPARATUS REQUIRED

System running MATLAB software

THEORY

Delta Pulse Code Modulation (DPCM)

DPCM is a procedure of converting an analog into a digital signal in which an analog signal is sampled and the difference b/w the actual sample and its predicted value (predicted value is based on previous sample or samples) is quantized and then encoded forming a digital value.

DPCM code words present differences b/w samples unlike PCM where code words represent a sample value. Basis Concept of DPCM is Coding a difference b/w samples unlike based on the fact that most sources signals show significant Correlations b/w successive samples so encoding uses redundancy in sample values which implies lower bitrate

Realization of basic concepts (described above) is based on a technique in which we have to predict current sample value based upon previous samples (or sample) and we have to encode the difference b/w samples actual value and predicted value

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Delta Pulse Code Modulator

The delta modulator comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a modulator. The predicted circuit in DPCM is replaced by a simple delay circuit in DM. A staircase approximated waveform will be the output of the delta modulator with the step size as delta (Δ). The output quality of the waveform is moderate.

The signals at each point are named as

$x(nT_s)$ → sampled input

$\hat{x}(nT_s)$ → predicted sample

$e(nT_s)$ → difference b/w sampled i/p and predicted o/p known as prediction error

$v(nT_s)$ → quantized output

$u(nT_s)$ → predictor input, summation of predictor output and the quantizer output

The predictor produces the assumed samples from the previous outputs of the transmitter circuit. The input to this predictor is the quantized version of input signals $x(nT_s)$

Quantizer Output is represented as

$$v(nT_s) = Q[e(nT_s)] = e(nT_s) + q(nT_s)$$

where $q(nT_s)$ is quantization error

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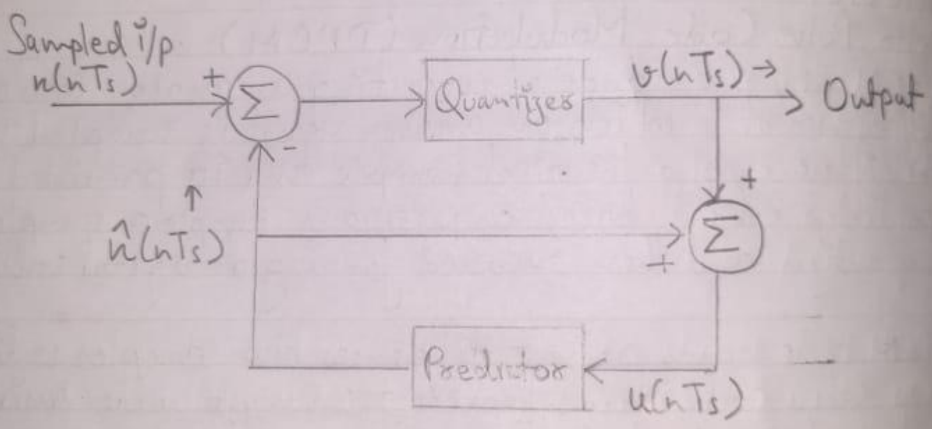

fig → DPCM Modulator

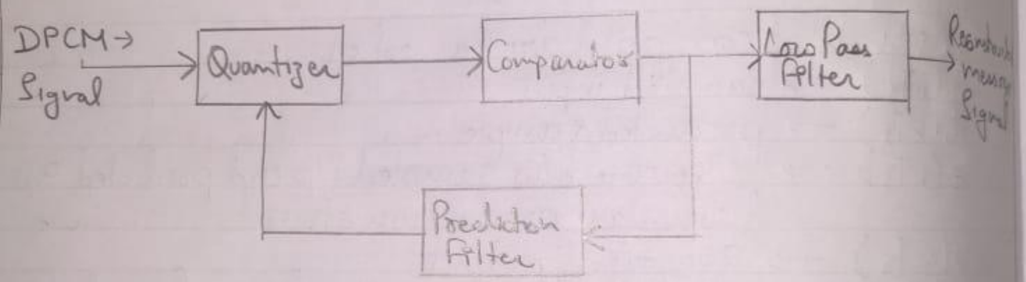

fig → DPCM Demodulator

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Predictor input is the sum of quantizer output and predictor output

$$u(nT_s) = \hat{x}(nT_s) + v(nT_s)$$
$$u(nT_s) = \hat{x}(nT_s) + e(nT_s) + q(nT_s)$$
$$u(nT_s) = x(nT_s) + q(nT_s)$$

The same predictor circuit is used in the decoder to re-construct the original i/p.

Delta Pulse Code Demodulator.

The delta demodulator comprises a low pass filter, a summer and a delay circuit. The predictor circuit is eliminated here and hence the assumed i/p is given to the demodulator.

A Binary Sequence will be given as i/p to the demodulator. The stair case approximated output is given to LPF. Low pass filters are used for many reasons, but the prominent reason is noise elimination for out of bound signals. The step size error that may occurs at the transmitter is called granular noise which is eliminated here. If there is no noise present then the modulator output equals to demodulator i/p.

CONCLUSION

The study for Modulation and Demodulation for DPCM is successfully done.

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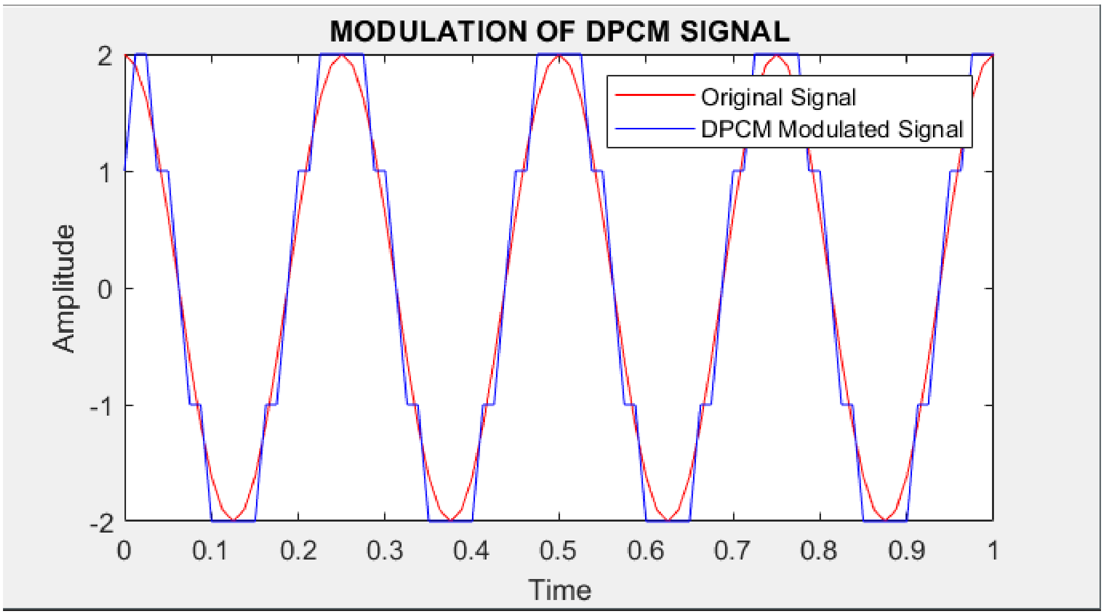
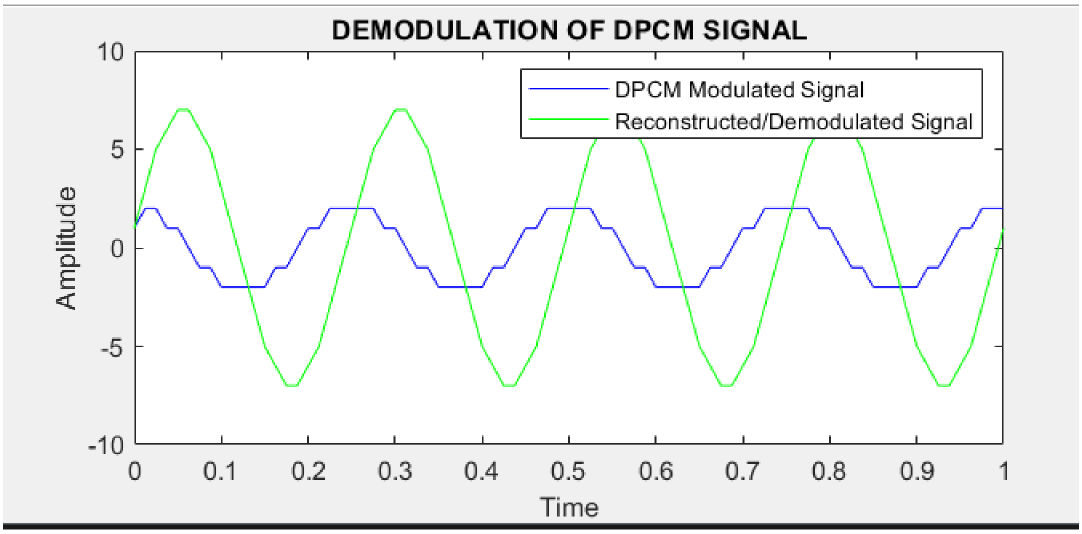
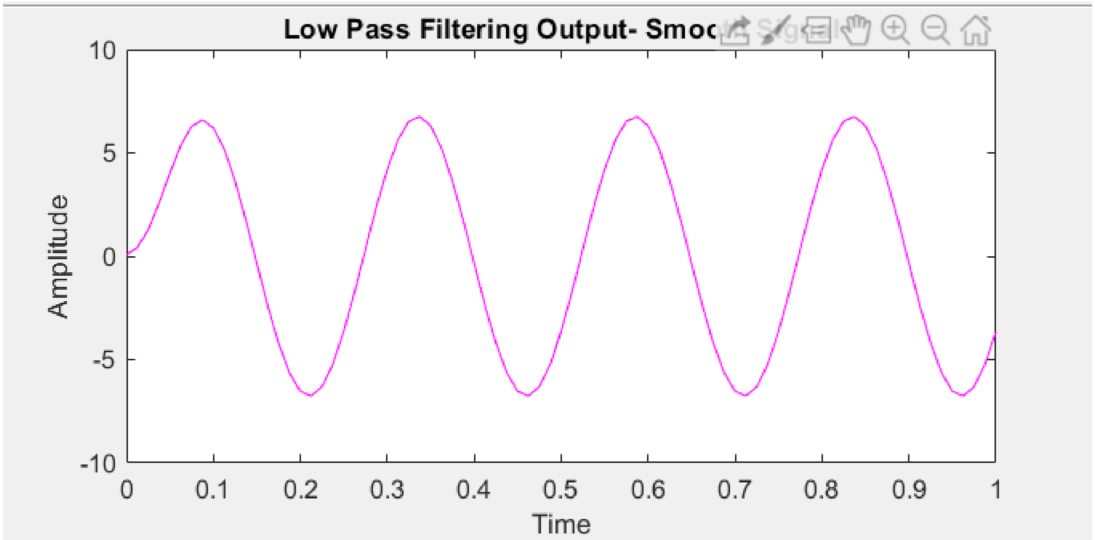
MATLAB PROGRAM TO IMPLEMENT DPCM MODULATION AND DEMODULATION

```
%DPCM Modulator implemented in Transmitter

%Input Sinusoidal Signal:

%frequency
fm = 4;
%Sampling Frequency
fs = 20*fm;
%Amplitude
am = 2;
%time
t = 0:1/fs:1;
%Sinusoidal Signal
x = am*cos(2*pi*fm*t);
figure(1);
plot(t,x,'r-');
hold on;
xlabel('Time');
ylabel('Amplitude');

for n=1:length(x)
    if n==1
        e(n) = x(n);
        eq(n) = round(n);
        xq(n) = eq(n);
    else
        e(n) = x(n) - xq(n-1);
        eq(n) = round(e(n));
        xq(n) = eq(n) + xq(n-1);
    end
end

plot(t,xq,'b-');
hold on;
xlabel('Time');
ylabel('Amplitude');
legend('Original Signal','DPCM Modulated Signal');
title('MODULATION OF DPCM SIGNAL');

%DPCM Demodulator implemented in Receiver

for n=1:length(xq)
    if n==1
        xqr(n) = xq(n);
    else
        xqr(n) = xq(n) + xqr(n-1);
    end
end
figure(2)
plot(t,xq,'b-');
hold on;
xlabel('Time');
ylabel('Amplitude');

plot(t,xqr,'g-');
xlabel('Time');
ylabel('Amplitude');
```

```
legend('DPCM Modulated Signal ','Reconstructed/Demodulated Signal');
title('DEMODULATION OF DPCM SIGNAL');

%Low Pass Filtering

%Butterworth filter
[num den] = butter(2, 4*fm/fs);
%Low Pass Filtering
rec_op = filter(num,den,xqr);
figure(3);
plot(t,rec_op,'m-');
xlabel('Time');
ylabel('Amplitude');
title('Low Pass Filtering Output- Smooth Signal');
```







Low Pass Filtering Output- Smoo
10
5
0
-5
-10
Amplitude
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Time

Practical 9

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AIM

To study Adaptive Delta Modulation and Demodulation for a Signal.

APPARATUS

System Running MATLAB Software.

THEORY

Adaptive Delta Modulation (ADM)

In digital communication, a problem of determining the step-size, which influences the quality of the output wave. A larger step size is needed at steep slope of the modulating signal and a smaller step size is needed where the message has a small slope. The minute details get missed in the process. So it would be better to control the adjustments of step size, according to our requirements in order to obtain the sampling in a desired fashion. This is the concept of ADM.

In ADM, the step size of the staircase signal is not fixed and changes depending upon the input signal. Here, first the difference b/w the present sample value and previous approximation is calculated. This error is quantized i.e if the present sample is smaller than the previous approximation, quantized value is high or else it is low. The o/p of the one-bit quantizer is given to the Logic step size control circuit where the step size is decided. At the Logic step size control circuit, the o/p is decided based on the quantizer output. If the quantizer o/p is high,

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then the step size is doubled for the next sample. If the quantizer output is low, the step size is reduced by one step for the next sample.

Adoptive Delta Modulator

The gain of the voltage controlled amplifier is adjusted by the output signal from the sampler. The amplifier gain determines the step size and both are proportional.

ADM quantizes the differences b/w the value of the current sample and the predicted value of the next sample. It uses a variable step height to predict the next value, for the faithful reproduction of the fast varying values

Adoptive Delta Demodulator

The receiver has two portions. The first portion produces the step size from each incoming bit. Follows exactly same process as that of Transmitter.

The previous i/p and the present i/p decide the step size. It is then applied to the second portion i.e, an accumulator which builds up a staircase waveform.

The low pass filter then smoothes out the staircase waveform to reconstruct the original symbol.

Difference b/w Delta Modulation and Adoptive Delta Modulation.

→ In Delta Modulation step size is fixed for the whole signal. Whereas in Adoptive Delta Modulation, the step size

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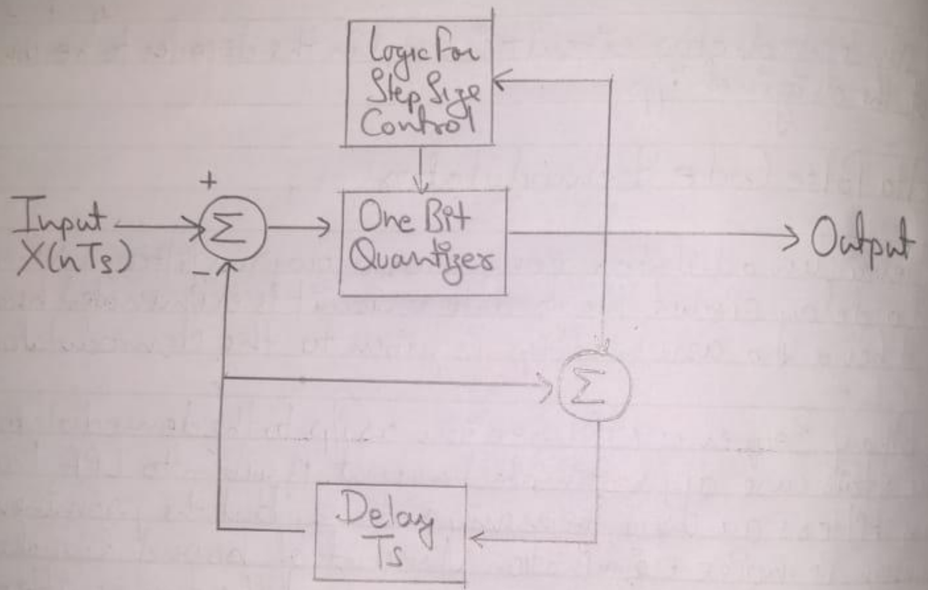

fig → Adaptive Delta Modulation Modulator

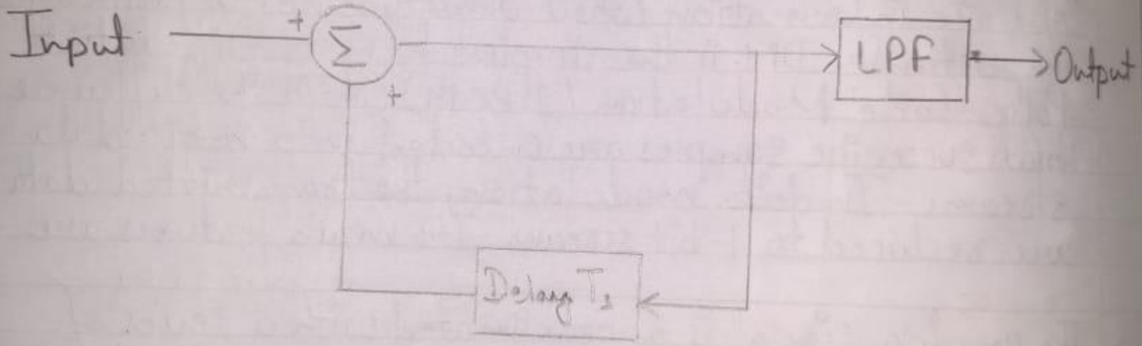

Fig → Adaptive Delta Demodulator Block Diagram

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varies depending upon the i/p signal.

-> The slope overloads and granular noise errors which are present in delta modulation and are not seen in this modulation.

-> The dynamic range of Adaptive Delta Modulation is wider than delta modulation.

-> This modulation utilizes bandwidth more effectively than delta modulation.

Application for ADM

-> This modulation is used for a system which requires improved wireless voice quality as well as speed transfer of bits.

-> In Television Signal Transmission this modulation process is used.

-> This modulation method is used in voice coding.

CONCLUSION

The study of Adaptive Delta Modulation and Demodulation is completed.

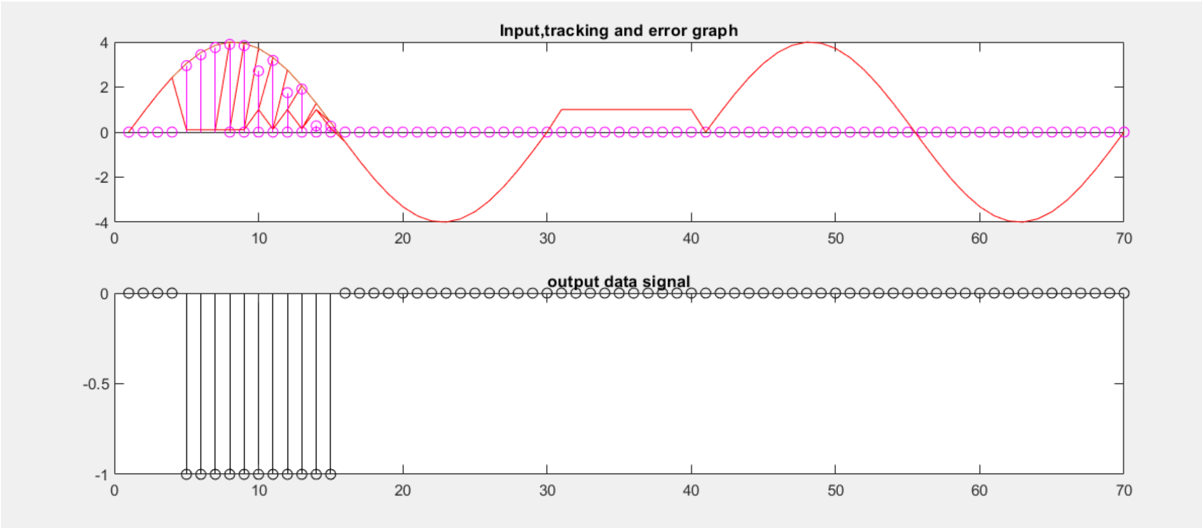
Signature ______

MATLAB PROGRAM TO IMPLEMENT ADAPTIVE DELTA MODULATION

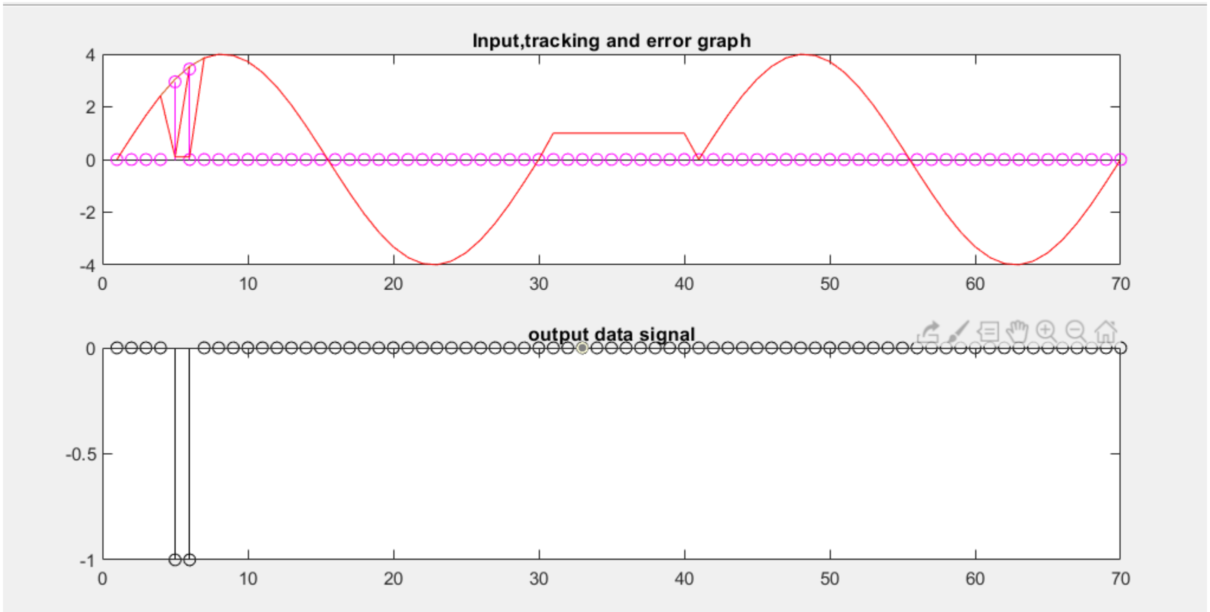
```
t =0:1/29:1;
f=1;
x=4*sin(2*pi*f*t);
x=[x ones(1,10) x];
y = zeros(1,length(x));
d = zeros(1,length(x));
e= zeros(1,length(x));
s=0.1;
for i=5:length(x)
if(x(i)-y(i-1))>=0
      y(i) = x(i)-s;
        d(i)=-1;
elseif(x(i)-y(i-1))<0
y(i)=x(i)-1;
d(i)=-1;
end

if(sum(d(i-4:i)))>3
s=s+0.01;
elseif(sum(d(i-4:i)))<-3
  s=s+0.01;
elseif(sum(d(i-4:i)))==0
s=s-0.01;
else
s=s;
end
pause;
subplot 211;
plot(x);
hold on;
stem(y,'m');
e=x-y;plot(e,'r');

title('Input,tracking and error graph');
subplot 212; stem (d,'k'); title('output data signal')
end
```



Input,tracking and error graph
output data signal



Input,tracking and error graph
output data signal

Practical 10

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AIM

To calculate and compare practical and theoritical values from S/N ratio versus Probability and Error Curve

APPARATUS REQUIRED

System Running MATLAB Software

THEORY

Signal to Noise Ratio (SNR)

In terms of definition, SNR or signal to Noise Ratio is the ratio b/w the desired information or the power of a signal and undesired signal or the power of the background noise.

In other words, SNR is the ratio of signal power to the noise power, and its unit of expression is typically decibels (dB). Also, a ratio greater than 0 dB or higher than 1:1, signifies more signal than noise.

Signal to Noise Ratio Formula

The signal to noise ratio is the ratio b/w the wanted signal and the unwanted background noise. It can be expressed as the most basic form using S/N ratio formula.

$$SNR = \frac{P_{signal}}{P_{noise}}$$

It is more usual to see a signal to noise ratio expressed in a logarithmic basis using decibels with the formula

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$$SNR(dB) = 10\log_{10}\left(\frac{P_{signal}}{P_{noise}}\right)$$

If all levels are expressed in decibels, then the formula can be simplified to the equation below.

$$SNR(dB) = P_{signal}(dB) - P_{noise}(dB)$$

The Power levels may be expressed in levels such as dBm (decibels relative to a milliwatt, or to some other standard by which levels are compared.)

Probability of Error Curve (POE)
Probability Theory is used extensively in the design of modern Communication systems in order to understand the behaviour of noise in these systems and take measures to correct the errors.

The rate at which data can be communicated is called the data rate. The rate at which error occurs in the bits, while transmitting data is called the Bit Error Rate BER.

The probability of the occurrence of BER is the error probability. The increase in SNR decreases the BER; hence the Error Probability also gets decreased. The symbol probabilities, that is, the P_0 and P_1 are the probabilities of the transmission of a symbol '0' and '1'

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The average error probability is precisely the mean value of the errors in the transmission that takes into account the probability of occurrence of each symbol.

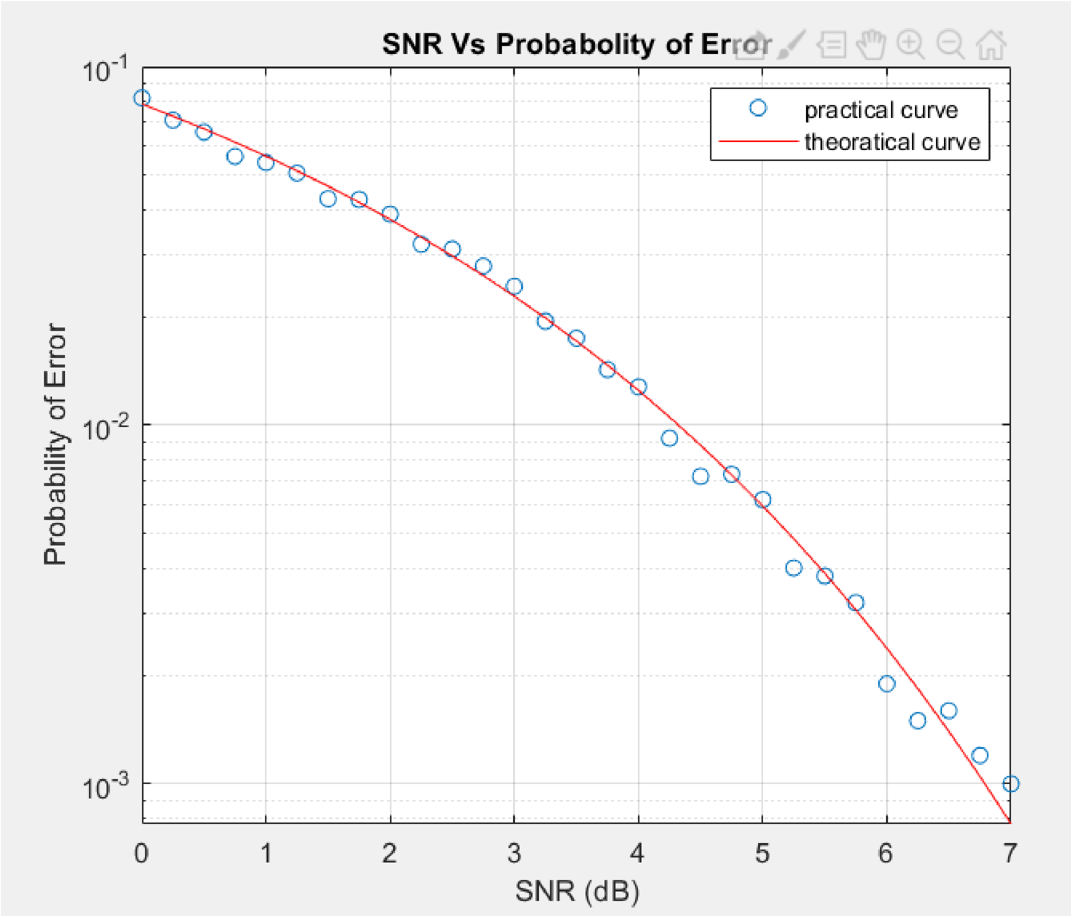
CONCLUSION

Hence, the study of Calculating and comparing the practical and theoretical values from S/N ratio versus Probability of Error curve, via Matlab, has been done successfully.

MATLAB PROGRAM TO DEMONSTRATE SNR VS POE CURVE

```
N=10^4;
%a sequence of 10^4 random bits
sent_bits=randi([0,1],1,N);
%amplitude of S(t)
A=10;
%duration of S(t)
T=10;
%rectangular pulse
S=ones(1,T)*A;
%S(t) Energy
E=norm(S)^2;
%matched filter
h=fliplr(S);
%mapping the bits with S(t)
bits=(2*sent_bits-1);
x=kron(bits,S);
%Noise that gives SNR=0dB to SNR=7dB
SNR_db=0:0.25:7;
n_var=(E/2)*10.^(-SNR_db/10);
for k=1:length(n_var)
%noise
n=randn(1,length(x))*sqrt(n_var(k));
%recieved signal
r=x+n;
%applying the recieved signal to matched filter
z=filter(h,1,r);
%sampling at T & using thresholding operation
z=sign(z(T:T:end));
%recovering the sent bits
recieved_bits=(z+1)/2;
%calculating the bit error
Pb(k)=mean(abs((sent_bits)-(recieved_bits)));
end

%plotting Pb Vs SNR
SNR=10.^(SNR_db/10);
P_theory=(0.5)*erfc(sqrt(2.*SNR)./sqrt(2));
semilogy(SNR_db,Pb,'o',SNR_db,P_theory,'r-')
title ('SNR Vs Probabolity of Error');
xlabel ('SNR (dB)');
ylabel ('Probability of Error');
legend('practical curve','theoratical curve');
grid on;
```



SNR Vs Probabolity of Error
practical curve
theoratical curve
Probability of Error
SNR (dB)
10-1
10-2
10-3
0
1
2
3
4
5
6
7