SIR C.R.REDDY COLLEGE OF ENGINEERING
ELURU-534007

Department of Electronics and Communications

DIGITAL SIGNAL PROCESSING Lab Manual (MTCS - 15)
For I / II M.Tech (Communication Systems), II - Semester

SIR C.R.REDDY COLLEGE OF ENGINEERING
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Digital Signal Processing Lab

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1.SAMPLING &RECONSTRUCTION OF ANALOG SIGNALS

AIM:

Write a MATLAB program for Sampling &reconstruction of analog signals.

PROGRAM:

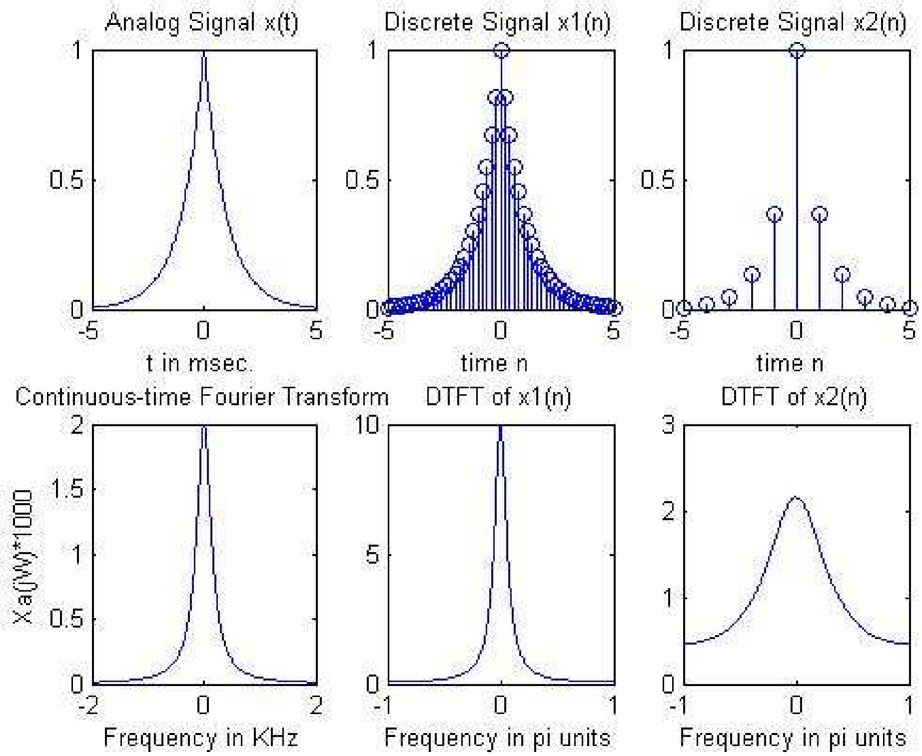
```
%%      Sampling & Reconstruction of Analog Signals
% Analog Signal: x(t)=exp(-a*abs(t)), X(jw)=2a/(a^2+w^2)
clear;
Dt = 0.00005;Fs=1/Dt;
t = -0.005:Dt:0.005;
xa = exp(-1000*abs(t));

% Discrete-time Signal x1(n): sampled above Nyquist rate
Ts1 = 0.0002; Fs1 = 1/Ts1; n1 = -25:1:25;
 nTs1 = n1*Ts1;
xn1 = exp(-1000*abs(nTs1));

% Discrete-time Signal x2(n): sampled below Nyquist rate
Ts2 = 0.001; Fs2 = 1/Ts2; n2 = -5:1:5;
nTs2 = n2*Ts2;
xn2 = exp(-1000*abs(nTs2));

% Continuous-time Fourier Transform
Wmax = 2*pi*2000;
K = 500; k = 0:1:K;
W = k*Wmax/K;

%Xa1 = xa * exp(-j*t'*W) * Dt; Xa1 = real(Xa1); %numerical approx of FT
Xa=2e3./(1e6+W.^2); %exact
W = [-fliplr(W), W(2:501)]; % Omega from -Wmax to Wmax
Xa = [fliplr(Xa), Xa(2:501)]; %CTFT

% Discrete-time Fourier transform(DTFT) of x1(n) & x2(n)
K = 500; k = 0:1:K; %compute DTFT at K points
w = pi*k/K; %freq points
X1 = xn1 * exp(-j*n1'*w);X1 = real(X1);
X2 = xn2 * exp(-j*n2'*w);X2= real(X2);
w = [-fliplr(w), w(2:K+1)]; %freq axis at 2K+1 points
X1 = [fliplr(X1), X1(2:K+1)];X2 = [fliplr(X2), X2(2:K+1)]; %DTFT


% plots
subplot(2,3,1);plot(t*1000,xa);xlabel('t in msec.');title('Analog Signal x(t)')
subplot(2,3,2);stem(n1*Ts1*1000,xn1);xlabel('time n '); title('Discrete Signal x1(n)');
```

```
subplot(2,3,3);stem(n2*Ts2*1000,xn2);xlabel('time n '); title('Discrete Signal x2(n)');
subplot(2,3,4);plot(W/(2*pi*1000),Xa*1000);
xlabel('Frequency in KHz'); ylabel('Xa(jW)*1000');title('Continuous-time Fourier
Transform')
subplot(2,3,5);plot(w/pi,X1);xlabel('Frequency in pi units');title('DTFT of x1(n)');
subplot(2,3,6);plot(w/pi,X2);xlabel('Frequency in pi units');title('DTFT of x2(n)');

% Analog Signal Reconstruction with Low Pass Filter

% Interpolation using sinc function
xa1 = xn1 * sinc(Fs1*(ones(length(nTs1),1)*t-nTs1'*ones(1,length(t))));
error1 = max(abs(xa1 - exp(-1000*abs(t)))) %print error
xa2 = xn2 * sinc(Fs2*(ones(length(nTs2),1)*t-nTs2'*ones(1,length(t))));
error2 = max(abs(xa2 - exp(-1000*abs(t))))

% Plots
figure;
subplot(2,2,1);plot(t*1000,xa);xlabel('t in msec.');title('Original Signal x(t)')
subplot(2,2,2);plot(t*1000,xa1);xlabel('t in msec.'); ylabel('x1(t)')
title('Reconstructed Signal from x1(n) using sinc function'); hold on
subplot(2,2,3);plot(t*1000,xa2);xlabel('t in msec.'); ylabel('x2(t)')
title('Reconstructed Signal from x2(n) using sinc function'); hold on

% Practical Reconstruction with Zero Order Hold/First Order Hold
% using MATLAB stairs(ZOH) and plot(FOH) functions

% Analog Signal reconstruction from x1(n)
figure;
subplot(2,2,1); stairs(nTs1*1000,xn1);xlabel('t in msec.'); ylabel('xa1(t)')
title('Reconstructed Signal from x1(n) using zero-order-hold'); hold on
subplot(2,2,2); plot(nTs1*1000,xn1);xlabel('t in msec.'); ylabel('xa1(t)')
title('Reconstructed Signal from x1(n) using first-order-hold'); hold on

% Analog Signal reconstruction from x2(n)
subplot(2,2,3); stairs(nTs2*1000,xn2);xlabel('t in msec.'); ylabel('xa2(t)')
title('Reconstructed Signal from x2(n) using zero-order-hold'); hold on
subplot(2,2,4); plot(nTs2*1000,xn2);xlabel('t in msec.'); ylabel('xa2(t)')
title('Reconstructed Signal from x2(n) using first-order-hold'); hold on
```

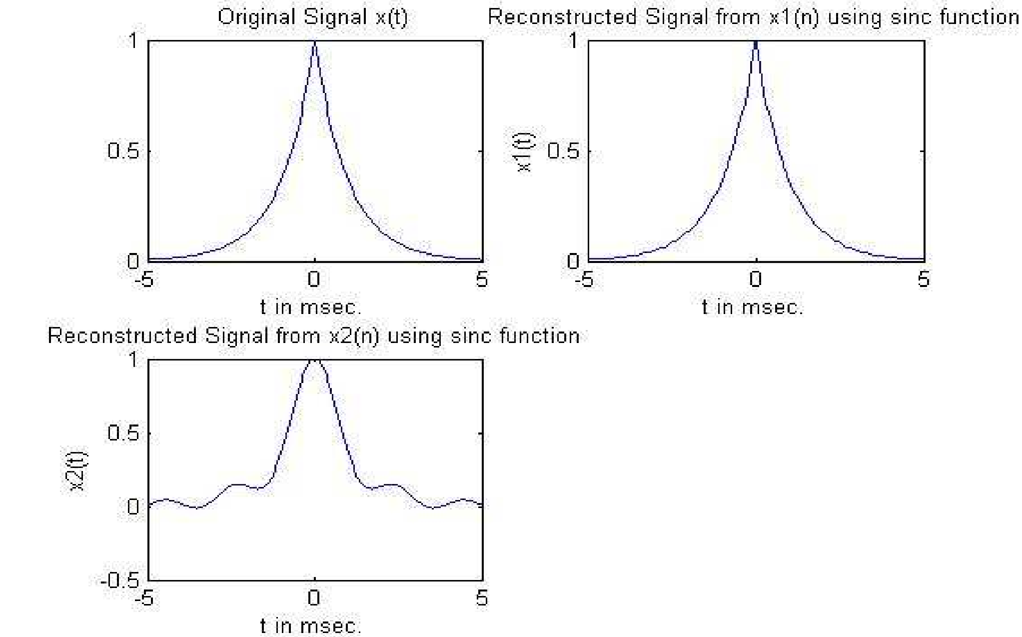
RESULT:

Sampling & reconstruction of analog signals are generated and plotted using MATLAB.

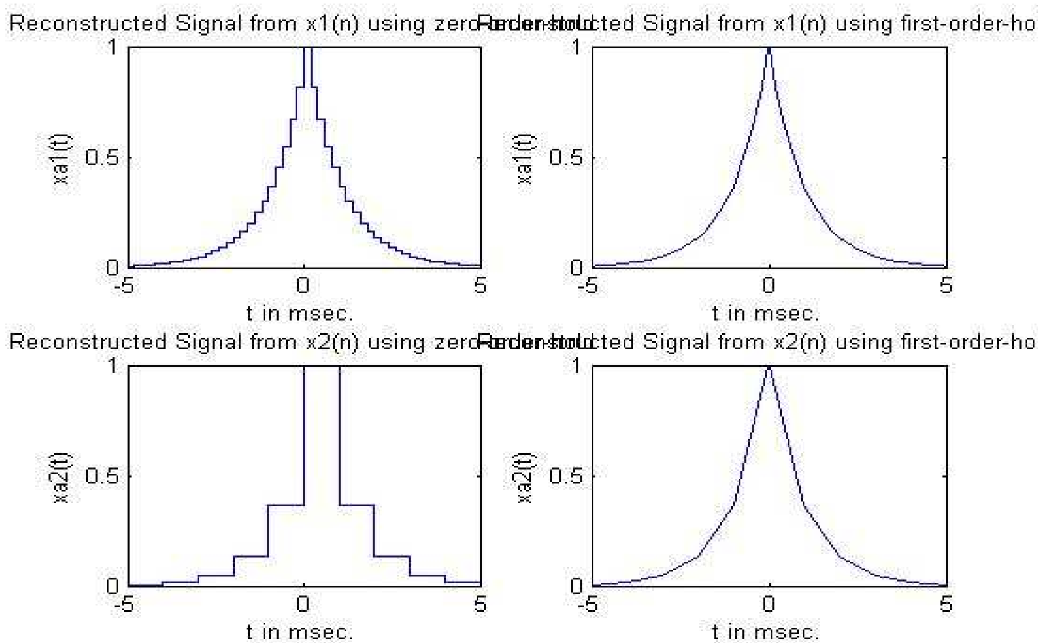
Wave Forms:



Analog Signal x(t)
Discrete Signal x1(n)
Discrete Signal x2(n)
1
0.5
0
-5
0
5
t in msec.
time n
time n
Continuous-time Fourier Transform
DTFT of x1(n)
DTFT of x2(n)
Xa(jW)*1000
2
1.5
1
0.5
10
3
-2
-1
Frequency in KHz
Frequency in pi units
Frequency in pi units



Original Signal x(t)
t in msec.
Reconstructed Signal from x1(n) using sinc function
x1(t)
t in msec.
Reconstructed Signal from x2(n) using sinc function
x2(t)
t in msec.



Reconstructed Signal from x1(n) using zer
xa1(t)
t in msec.
ted Signal from x1(n) using first-order-hol
xa1(t)
t in msec.
Reconstructed Signal from x2(n) using zer
xa2(t)
t in msec.
ted Signal from x2(n) using first-order-hol
xa2(t)
t in msec.

2. FILTERING OF SIGNALS

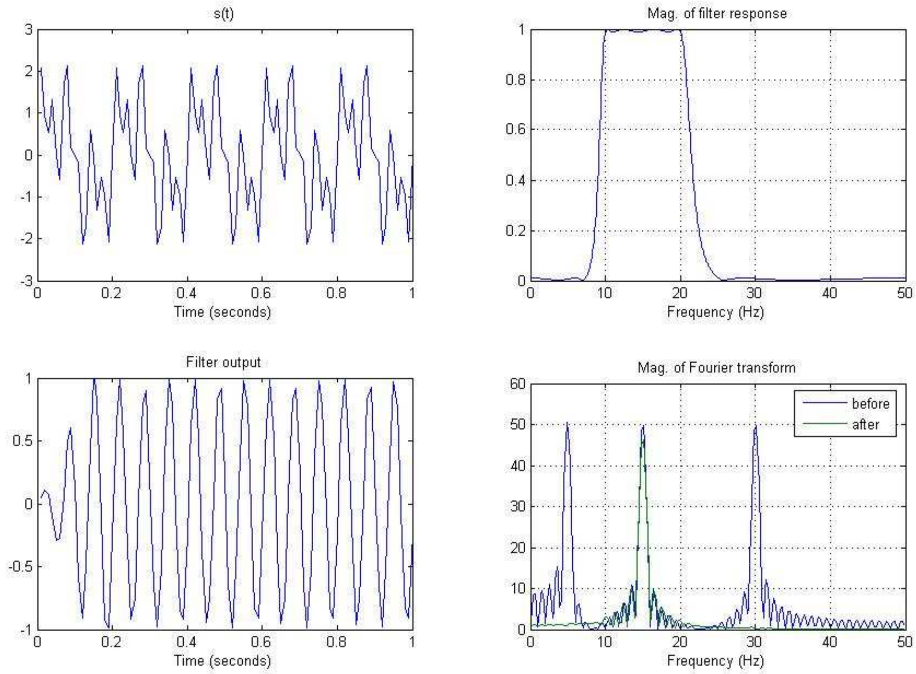
AIM:
Write a MATLAB program for filtering of signals.

PROGRAM:

```
%%      Filtering of Signals

Fs = 100; % sampling freq
t = (1:100)/Fs; % time axis
s1 = sin(2*pi*t*5); s2=sin(2*pi*t*15); s3=sin(2*pi*t*30); % 3 freq are present
s = s1+s2+s3; % signal
subplot(2,2,1),plot(t,s);xlabel('Time (seconds)');title('s(t)');%ylabel('Time waveform');

% design a digital filter(BP) to select 15Hz component
[b,a] = ellip(4,0.1,40,[10 20]*2/Fs); % Elliptic filter
[H,w] = freqz(b,a,512); % compute frequency response
subplot(2,2,2),plot(w*Fs/(2*pi),abs(H));title('Mag. of filter response');
xlabel('Frequency (Hz)');
grid;

% send signal through filter
sf = filter(b,a,s);
subplot(2,2,3), plot(t,sf); xlabel('Time (seconds)'); title('Filter output');
axis([0 1 -1 1]);
S = fft(s,512); SF = fft(sf,512); w = (0:255)/256*(Fs/2); % compute spectrum
subplot(2,2,4),plot(w,abs([S(1:256)' SF(1:256)']));
title('Mag. of Fourier transform');
xlabel('Frequency (Hz)');
grid;
legend({'before','after'})
```

RESULT:
Filtering of signals is performed using MATLAB.

Wave Forms:


s(t)
Time (seconds)
Mag. of filter response
Frequency (Hz)
Filter output
Time (seconds)
Mag. of Fourier transform
before
after
Frequency (Hz)

3. SPECTRUM ANALYSIS USING FFT

AIM:

Write a MATLAB program for spectrum analysis using fft.

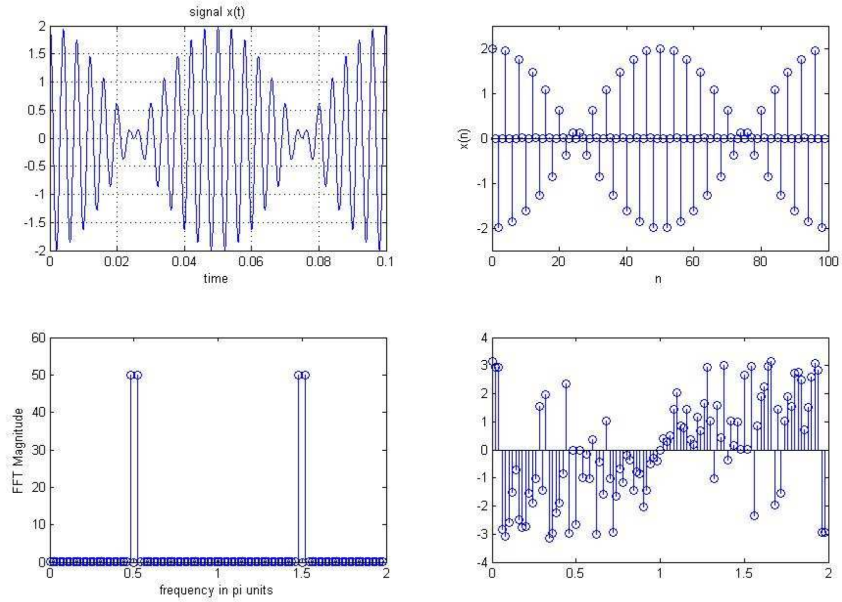
PROGRAM:

```
%%      Spectrum Analysis using FFT

% Nyquist freq=2*Fh, Fs>=2*Fh
% normalized digital freq=(2*pi*f)/Fs
% FFT resolution=Fs/N
% consider zero-padding for better display of FFT
% x(t)=cos(w1*t)+cos(w2*t)

clc;clear all; close all;
f1=240;f2=260;Fs=1000;Ts=1/Fs;Np=100;

% analog signal
t=0:0.0001:Ts*Np;xa=cos(2*pi*f1*t)+cos(2*pi*f2*t);

% discrete signal
n=[0:Np-1];
x=cos(2*pi*f1*n*Ts)+cos(2*pi*f2*n*Ts);
X=fft(x,length(x));magX=abs(X);
w=2*pi/Np*n;
figure;subplot(2,2,1);plot(t,xa);title('signal x(t)');xlabel('time');grid
subplot(2,2,2);stem(n,x);;xlabel('n');ylabel('x(n)')
axis([0,Np,-2.5,2.5])
subplot(2,2,3);stem(w/pi,magX);;xlabel('frequency in pi units');
ylabel('FFT Magnitude');
subplot(2,2,4);stem(w/pi,angle(X));
xlabel('frequency in pi units');
ylabel('FFT Phase');
```

RESULT:

Spectrum analysis using fft is performed using MATLAB.

Wave Forms:



4. PULSE CODE MODULATION

AIM:

Write a MATLAB program to generate pulse code modulation

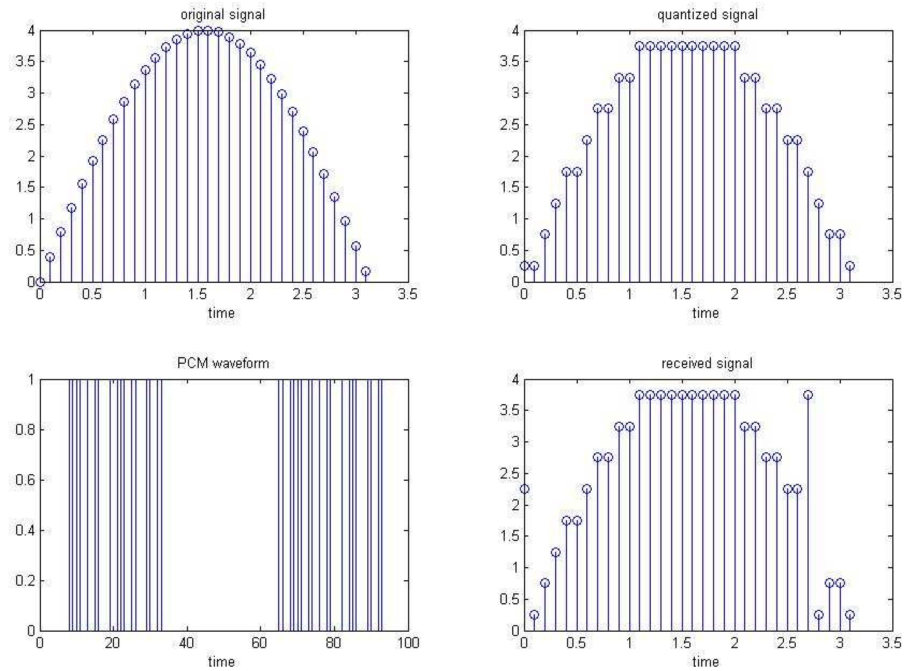
PROGRAM:

```
%%      Pulse Code Modulation (PCM)

P=5; %percentage of errors in transmission

% sampling
t = [0:0.1:1*pi]; % Times at which to sample the sine function
sig = 4*sin(t); % Original signal, a sine wave
%sig=exp(-1/3*t);

% quantization
Vh=max(sig);Vl=min(sig);
N=3;M=2^N;S=(Vh-Vl)/M; %design N-bit uniform quantizer with stepsize=S
partition = [Vl+S:S:Vh-S]; % Length M-1, to represent M intervals
codebook = [Vl+S/2:S:Vh-S/2]; % Length M, one entry for each interval

% partition = [-1:.2:1]; % Length 11, to represent 12 intervals
% codebook = [-1.2:.2:1]; % Length 12, one entry for each interval
[index,quantized_sig,distor] = quantiz(sig,partition,codebook); % Quantize.

% binary encoding
codedsig=de2bi(index,'left-msb');codedsig=codedsig';
txbits=codedsig(:); %serial transmit
errvec=randsrc(length(txbits),1,[0 1;(1-P/100) P/100]); %error vector

%rxbits=xor(txbits,errvec);
rxbits=rem(txbits+errvec,2); %bits received

rxbits=reshape(rxbits,N,length(sig));rxbits=rxbits';
index1=bi2de(rxbits,'left-msb'); %decode
reconstructedsig=codebook(index1+1); %re-quantize

%plot(t,sig,'x',t,quantized_sig,'.')
%plot(t,sig,'x-',t,quantized_sig,'.--',t,reconstructedsig,'d-')
%figure,stem(t,sig,'x-');hold;stem(t,quantized_sig,'.--');stem(t,reconstructedsig,'d-');
figure,subplot(2,2,1); stem(t,sig); xlabel('time'); title('original signal');
subplot(2,2,2); stem(t,quantized_sig); xlabel('time'); title('quantized signal');
tt=[0:N*length(t)-1];
subplot(2,2,3);stairs(tt,txbits); xlabel('time'); title('PCM waveform');
subplot(2,2,4);stem(t,reconstructedsig); xlabel('time'); title('received signal');
```

RESULT:

Pulse code modulation is performed using MATLAB.

Wave Forms:



original signal
quantized signal
PCM waveform
received signal
time
time
time
time

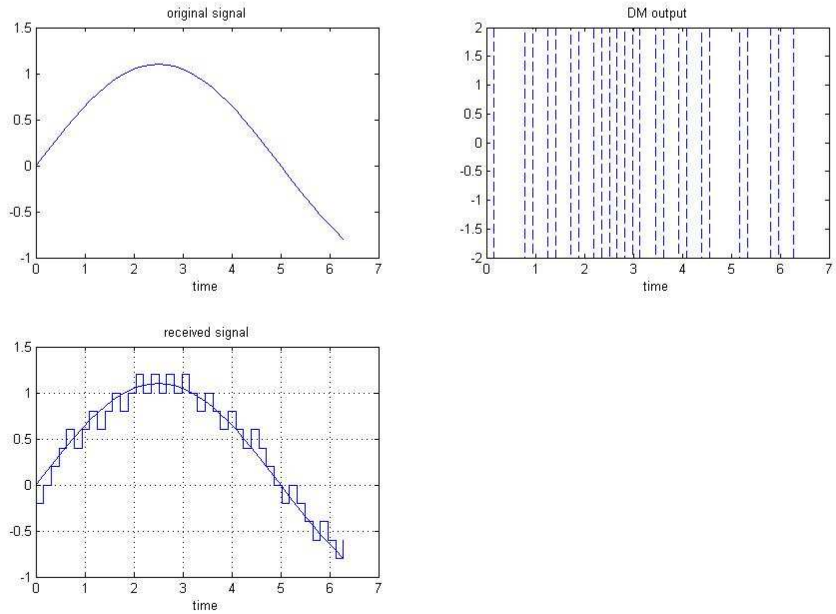
5. DELTA MODULATION

AIM:
Write a MATLAB program to perform delta modulation

PROGRAM:
```
%%      Delta Modulation (DM)

%delta modulation = 1-bit differential pulse code modulation (DPCM)
predictor = [0 1]; % y(k)=x(k-1)
%partition = [-1:.1:.9];codebook = [-1:.1:1];
step=0.2; %SFs>=2pifA
partition = [0];codebook = [-1*step step]; %DM quantizer

t = [0:pi/20:2*pi];
x = 1.1*sin(2*pi*0.1*t); % Original signal, a sine wave
%t = [0:0.1:2*pi];x = 4*sin(t);
%x=exp(-1/3*t);
%x = sawtooth(3*t); % Original signal

% Quantize x(t) using DPCM.
encodedx = dpcmenco(x,codebook,partition,predictor);

% Try to recover x from the modulated signal.
decodedx = dpcmdeco(encodedx,codebook,predictor);
distor = sum((x-decodedx).^2)/length(x) % Mean square error

% plots

figure,subplot(2,2,1);plot(t,x);xlabel('time');title('original signal');
subplot(2,2,2);stairs(t,10*codebook(encodedx+1),'--');xlabel('time');title('DM output');
subplot(2,2,3);plot(t,x);hold;stairs(t,decodedx);grid;xlabel('time');title('received signal');
```

RESULT:
Delta modulation is performed using MATLAB.

Wave Forms:



6. CDMA SIGNAL DETECTION IN GAUSSIAN NOISE USING MATCHED FILTER RECEIVER AND DECORRELATOR

AIM:

Write a MATLAB program to CDMA signal detection in Gaussian noise using matched filter receiver and decorrelator

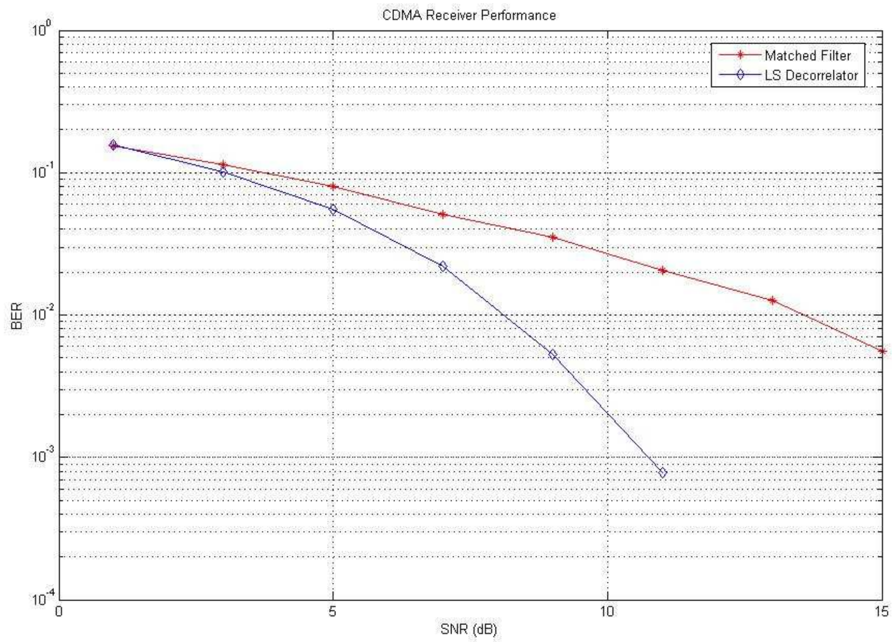
PROGRAM:

```
%% CDMA Signal Detection in Gaussian Noise using MF receiver and
%% De-correlator(faster code)

clear; tic,
K=8; N=31; M=10000; % no of bits
bsize=1000;nblocks=50; Nb=M/bsize;
snr_db=[1:2:15];
S2=randsrc(N,K); S=(1/sqrt(N))*S2;

R1=S'*S;s1=S(:,1); % user-1 sequence
LS=inv(R1)*S'; % Least-Squares detector

snr=10.^(0.1*snr_db); % linear snr
ber1=zeros(1,length(snr)); ber2=zeros(1,length(snr));
B=[];T=[];R=[]; %preallocate

A1=1;           % user 1 amplitude=fixed,vary noise var
A2=A1;          % user 2 relative amplitude
A3=A1; A4=A1; A5=A1; A6=A1; A=diag([A1,A2,A3,A4,A5,A6]);
A=eye(K);
H=S*A; % channel matrix
   for i=1:length(snr)
  var=1/snr(i);err1=0;err2=0; % initialize
  for k=1:nblocks %block processing
W=sqrt(var)*randn(N,bsize);
B=randsrc(K,bsize);    % user data matrix
T=H*B;                 % transmitted CDMA signal block
R=T+W;                 % received signal  block

z1=s1'*R;b1emf=sign(A1*z1); err1=err1+sum(b1emf ~= B(1,:)); %matched filter
z2=LS*R;b1ed=sign(A1*z2); err2=err2+sum(b1ed(1,:) ~= B(1,:)); %Decorrelator

end; % k loop
ber1(i)=err1/(bsize*nblocks); ber2(i)=err2/(bsize*nblocks);
end;toc
% plot
figure,semilogy(snr_db,ber1,'-r*'); hold on;
semilogy(snr_db,ber2,'-bd'); grid
xlabel('SNR (dB)'); ylabel('BER'); title('CDMA Receiver Performance')
legend('Matched Filter','LS Decorrelator');
```

RESULT:

CDMA signal detection in Gaussian noise using matched filter receiver and decorrelator is performed using MATLAB.

Wave Forms:



7. ADAPTIVE CHANNEL EQUALIZATION USING LMS ALGORTHIM

AIM:

Write a MATLAB program to Adaptive channel equalization using LMS algorthim.

PROGRAM:

```
%%      Channel Equalization using Adaptive Filters
%%      1. Basic Procedure for Equalizing a Signal

% Build a set of test data
x = pskmod(randint(1000,1),2); % BPSK symbols
rxsig = conv(x,[1 0.8 0.3]);  % Received signal with ISI
% Create an equalizer object.
eqlms = lineareq(8,lms(0.03));
% Change the reference tap index in the equalizer.
eqlms.RefTap = 4;
% Apply the equalizer object to a signal.
y = equalize(eqlms,rxsig,x(1:200));

%%      2. Equalizing Using a Training Sequence

% Set up parameters and signals.
M = 4; % Alphabet size for modulation
msg = randint(1500,1,M); % Random message
modmsg = pskmod(msg,M); % Modulate using QPSK.
trainlen = 500; % Length of training sequence
chan = [.986; .845; .237; .123+.31i]; % Channel coefficients
chan = [1 0.8 0.3];

filtmsg = filter(chan,1,modmsg); % Introduce channel distortion.

% Equalize the received signal.
eq1 = lineareq(8, lms(0.01)); % Create an equalizer object.
eq1.SigConst = pskmod([0:M-1],M); % Set signal constellation.
% Change the reference tap index in the equalizer.
eq1.RefTap = 4;

[symbolest,yd,e] = equalize(eq1,filtmsg,modmsg(1:trainlen)); % Equalize.
plot(20*log10(abs(e(1:250))));

% Plot signals.
h = scatterplot(filtmsg,1,trainlen,'bx'); hold on;
scatterplot(symbolest,1,trainlen,'g.',h);
scatterplot(eq1.SigConst,1,0,'k*',h);
legend('Filtered signal','Equalized signal',...
   'Ideal signal constellation');
```

```
hold off;
% Compute error rates with and without equalization.
demodmsg_noeq = pskdemod(filtmsg,M); % Demodulate unequalized signal.
demodmsg = pskdemod(yd,M); % Demodulate detected signal from equalizer.
[nnoeq,rnoeq] = symerr(demodmsg_noeq(trainlen+1:end),...
   msg(trainlen+1:end));
[neq,req] = symerr(demodmsg(trainlen+1:end),...
   msg(trainlen+1:end));
disp('Symbol error rates with and without equalizer:')
disp([req rnoeq])

%%      3. Equalizing in Decision-Directed Mode

M = 4; % Alphabet size for modulation
msg = randint(1500,1,M); % Random message
modmsg = pskmod(msg,M); % Modulate using QPSK.
trainlen = 500; % Length of training sequence
chan = [.986; .845; .237; .123+.31i]; % Channel coefficients
filtmsg = filter(chan,1,modmsg); % Introduce channel distortion.

% Set up equalizer.
eqlms = lineareq(8, lms(0.01)); % Create an equalizer object.
eqlms.SigConst = pskmod([0:M-1],M); % Set signal constellation.
% Maintain continuity between calls to equalize.
eqlms.ResetBeforeFiltering = 0;

% Equalize the received signal, in pieces.
% 1. Process the training sequence.
s1 = equalize(eqlms,filtmsg(1:trainlen),modmsg(1:trainlen));
% 2. Process some of the data in decision-directed mode.
s2 = equalize(eqlms,filtmsg(trainlen+1:800));
% 3. Process the rest of the data in decision-directed mode.
s3 = equalize(eqlms,filtmsg(801:end));
s = [s1; s2; s3]; % Full output of equalizer

%%      4. Delays from Equalization

M = 2; % Use BPSK modulation for this example.
msg = randint(1000,1,M); % Random data
modmsg = pskmod(msg,M); % Modulate.
trainlen = 100; % Length of training sequence
trainsig = modmsg(1:trainlen); % Training sequence

% Define an equalizer and equalize the received signal.
eqlin = lineareq(3,normlms(.0005,.0001),pskmod(0:M-1,M));
eqlin.RefTap = 2; % Set reference tap of equalizer.
[eqsig,detsym] = equalize(eqlin,modmsg,trainsig); % Equalize.

detmsg = pskdemod(detsym,M); % Demodulate the detected signal.
```

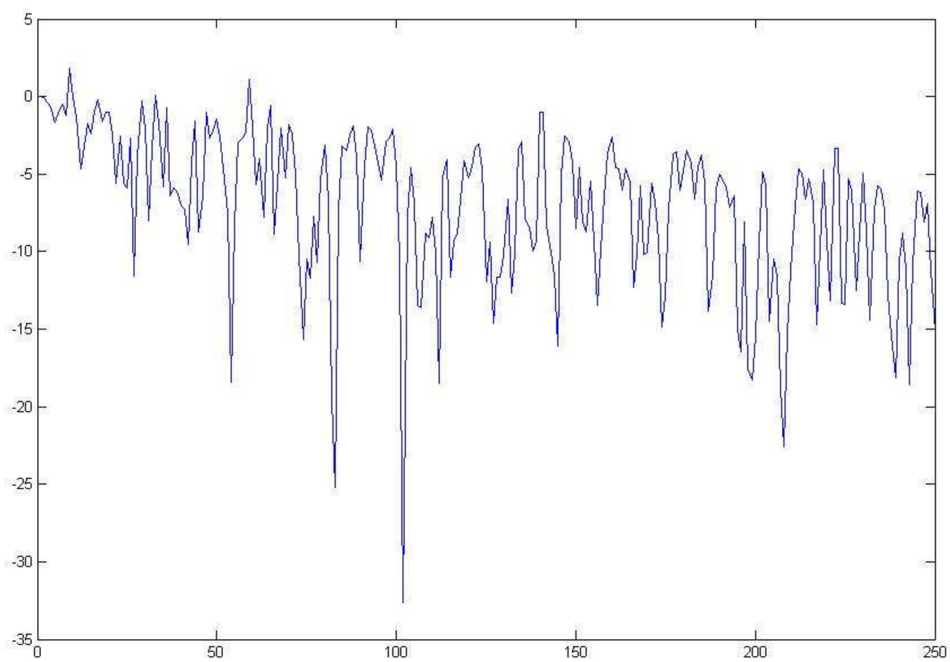
```
% Compensate for delay introduced by RefTap.
D = (eqlin.RefTap -1)/eqlin.nSampPerSym;
trunc_detmsg = detmsg(D+1:end); % Omit first D symbols of equalized data.
trunc_msg = msg(1:end-D); % Omit last D symbols.

% Compute bit error rate, ignoring training sequence.
[numerrs,ber] = biterr(trunc_msg(trainlen+1:end),...
  trunc_detmsg(trainlen+1:end))

%%      5. Equalizing Using a Loop

% Set up parameters.
M = 16; % Alphabet size for modulation
sigconst = qammod(0:M-1,M); % Signal constellation for 16-QAM
chan = [1 0.45 0.3+0.2i]; % Channel coefficients

% Set up equalizers.
eqrls = lineareq(6, rls(0.99,0.1)); % Create an RLS equalizer object.
eqrls.SigConst = sigconst; % Set signal constellation.
eqrls.ResetBeforeFiltering = 0; % Maintain continuity between iterations.

eqlms = lineareq(6, lms(0.003)); % Create an LMS equalizer object.
eqlms.SigConst = sigconst; % Set signal constellation.
eqlms.ResetBeforeFiltering = 0; % Maintain continuity between iterations.

eq_current = eqrls; % Point to RLS for first iteration.

% Main loop
for jj = 1:4
  msg = randint(500,1,M); % Random message
  modmsg = qammod(msg,M); % Modulate using 8-QAM.

% Set up training sequence for first iteration.
  if jj == 1
    ltr = 200; trainsig = modmsg(1:ltr);
  else
    % Use decision-directed mode after first iteration.
    ltr = 0; trainsig = [];
  end

  % Introduce channel distortion.
  filtmsg = filter(chan,1,modmsg);

  % Equalize the received signal.
  [s,sd,e] = equalize(eq_current,filtmsg,trainsig);

  % Plot signals.
  h = scatterplot(filtmsg(ltr+1:end),1,0,'bx'); hold on;
  scatterplot(s(ltr+1:end),1,0,'g.',h);
  scatterplot(sigconst,1,0,'k*',h);
```

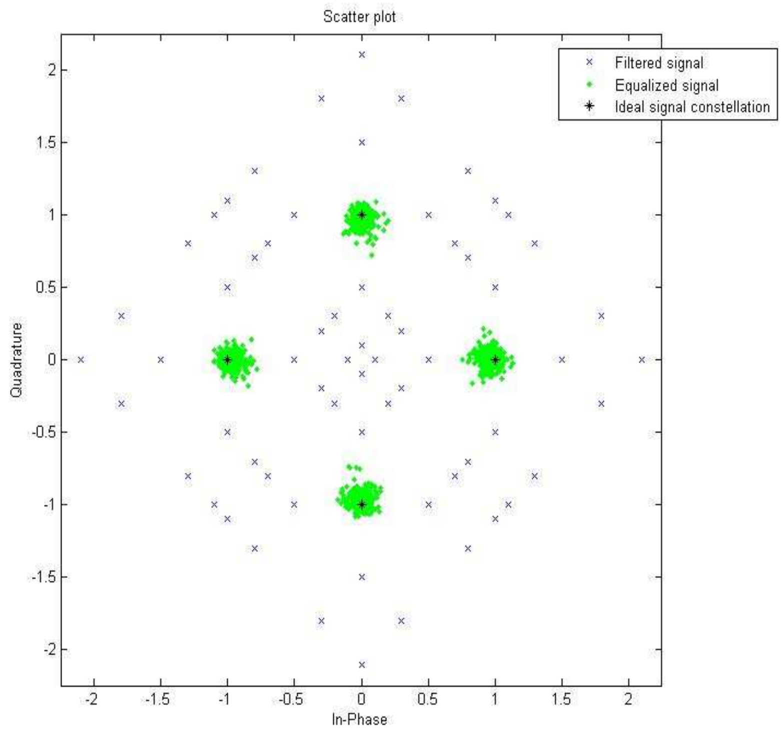
```
   legend('Received signal','Equalized signal','Signal constellation');
   title(['Iteration #' num2str(jj) ' (' eq_current.AlgType ')']);
   hold off;

   % Switch from RLS to LMS after second iteration.
   if jj == 2
      eqlms.WeightInputs = eq_current.WeightInputs; % Copy final inputs.
      eqlms.Weights = eq_current.Weights; % Copy final weights.
      eq_current = eqlms; % Make eq_current point to eqlms.
   end
end
figure,plot(20*log10(abs(e(1:500))));

%%
```

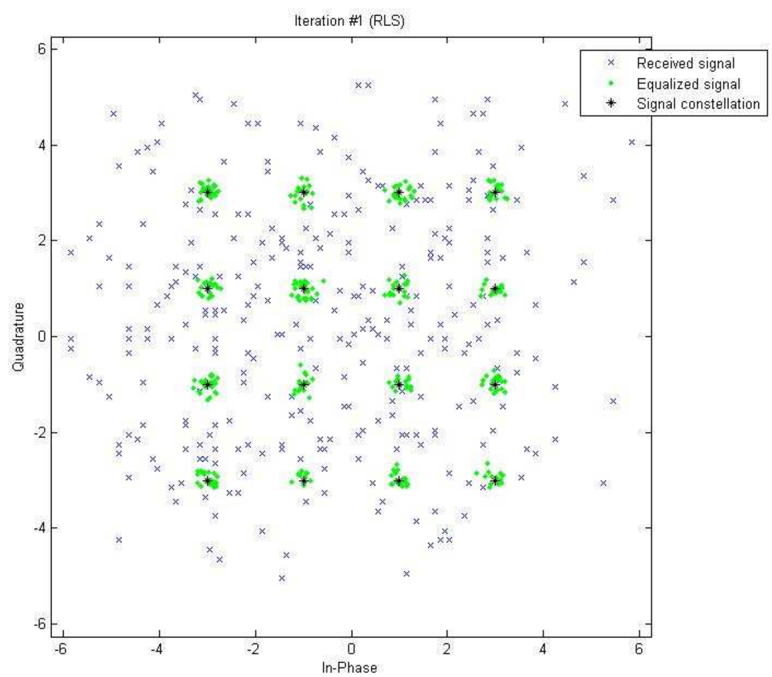
RESULT:

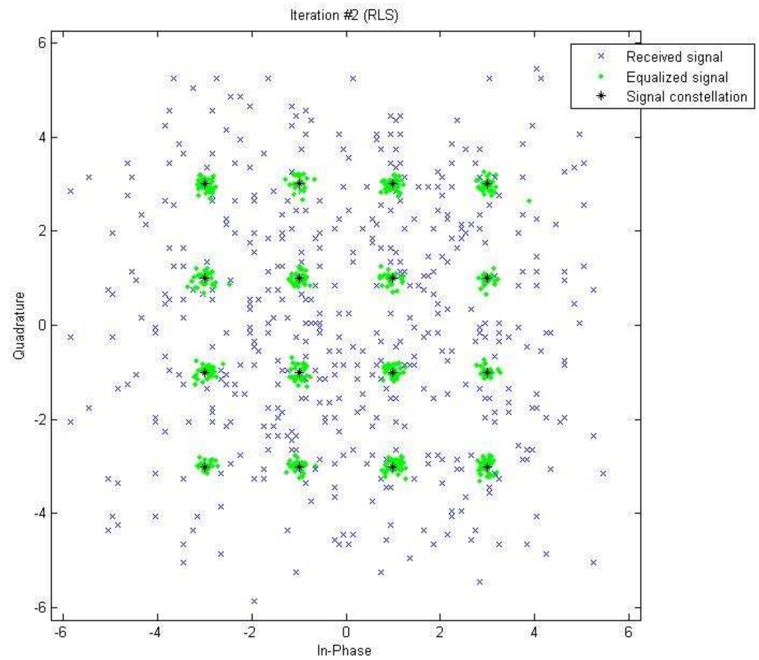
Adaptive channel equalization using LMS algorithm is performed using MATLAB.

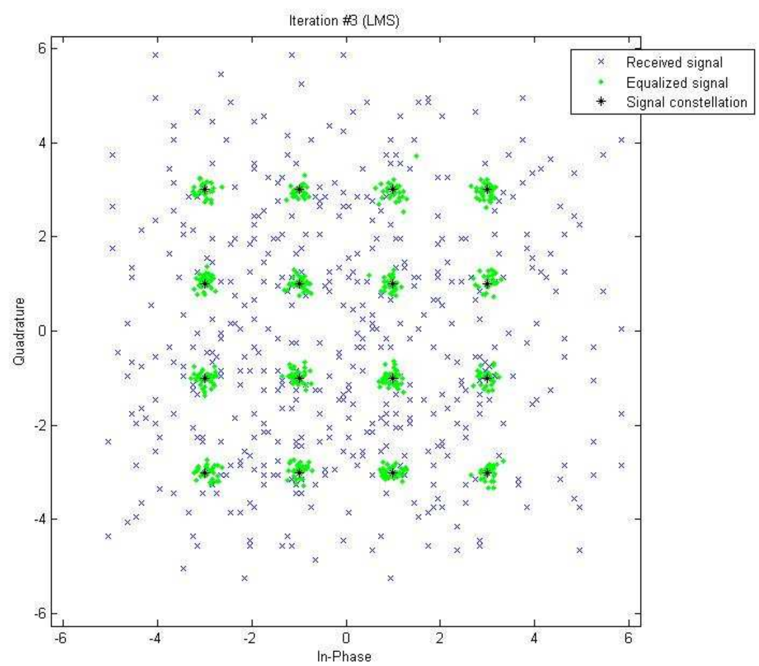
Wave Forms:

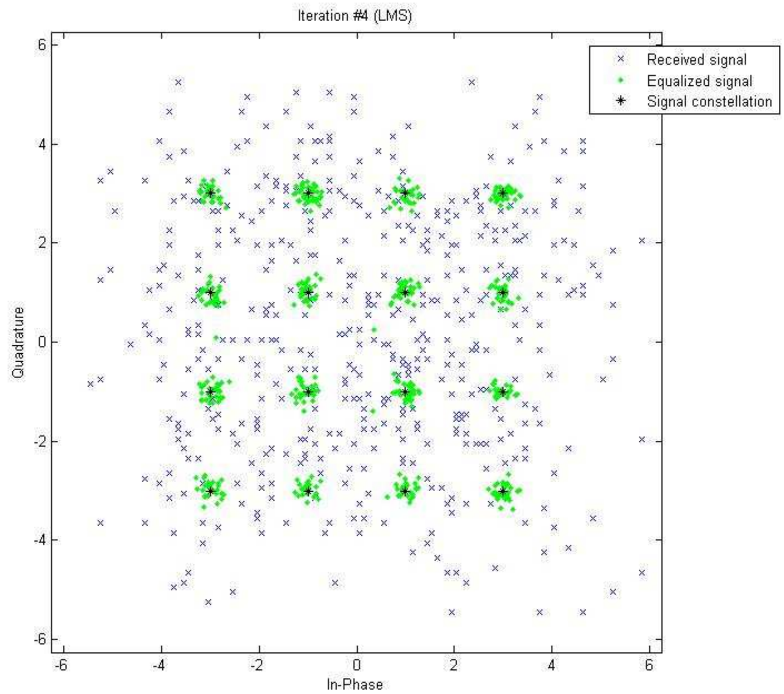


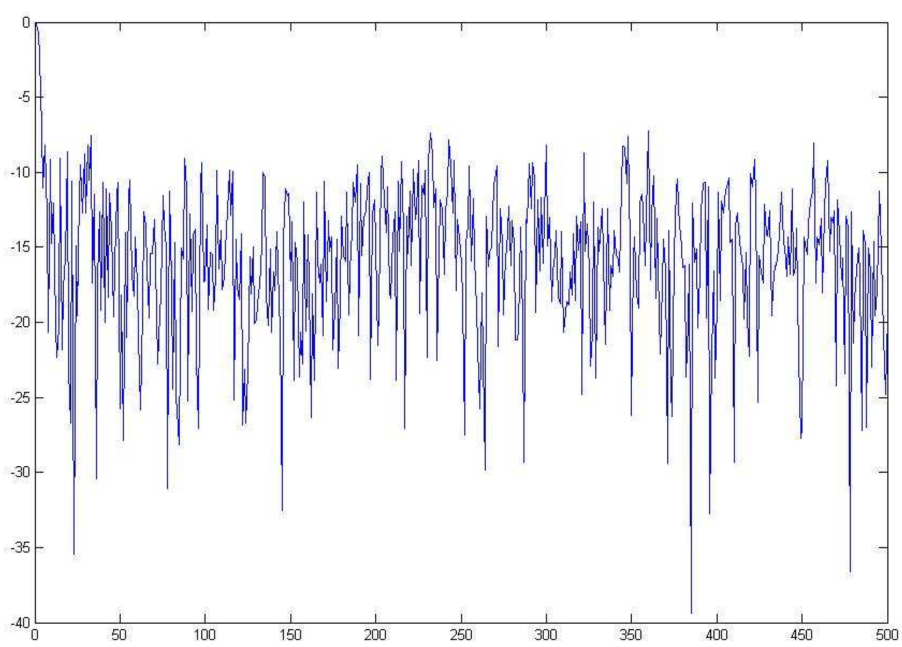

Scatter plot
Filtered signal
Equalized signal
Ideal signal constellation
Quadrature
In-Phase


Iteration #1 (RLS)
Received signal
Equalized signal
Signal constellation
Quadrature
In-Phase


Iteration #2 (RLS)
Received signal
Equalized signal
Signal constellation
Quadrature
In-Phase
6
4
2
0
-2
-4
-6


Iteration #3 (LMS)
Received signal
Equalized signal
Signal constellation
Quadrature
In-Phase
6
4
2
0
-2
-4
-6


Iteration #4 (LMS)
Received signal
Equalized signal
Signal constellation
Quadrature
In-Phase



8. CONVOLUTION CODING AND VITERBI DECODING

AIM:
Write a MATLAB program to Perform Turbo coding and decoding.

PROGRAM:

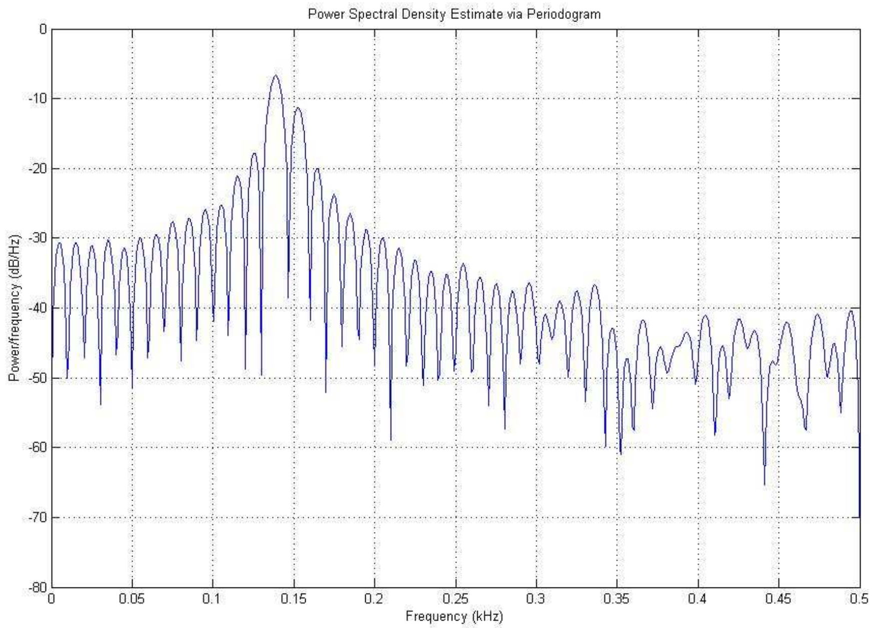
```
%%      Convolutional Coding and Viterbi Decoding Examples

% BSC channel
t = poly2trellis([4 3],[4 5 17;7 4 2]); % Trellis
msg = ones(10000,1); % Data to encode
code = convenc(ones(10000,1),t); % Encode using convolutional code.
[ncode,err] = bsc(code,.01); % Introduce errors in code.
numchanerrs = sum(sum(err)) % Number of channel errors
dcode = vitdec(ncode,t,2,'trunc','hard'); % Decode.
[numsyserrs,ber] = biterr(dcode,msg) % Errors after decoding

%%

t = poly2trellis([4 3],[4 5 17;7 4 2]); % Define trellis.
code = convenc(ones(100,1),t); % Encode a string of ones.
tb = 2; % Traceback length for decoding
decoded = vitdec(code,t,tb,'trunc','hard'); % Decode.

%%

t = poly2trellis(7,[171 133]); % Define trellis.
msg = randint(4000,1,2,139); % Random data
code = convenc(msg,t); % Encode the data.
ncode = awgn(code,6,'measured',244); % Add noise.

% Quantize to prepare for soft-decision decoding.
qcode = quantiz(ncode,[0.001,.1,.3,.5,.7,.9,.999]);
tblen = 48; delay = tblen; % Traceback length
decoded = vitdec(qcode,t,tblen,'cont','soft',3); % Decode.

% Compute bit error rate.
[number,ratio] = biterr(decoded(delay+1:end),msg(1:end-delay));

%%
trel = poly2trellis(3,[6 7]); % Define trellis.
msg = randint(100,1,2,123); % Random data
code = convenc(msg,trel); % Encode.
ncode = rem(code + randerr(200,1,[0 1;.95 .05]),2); % Add noise.
tblen = 3; % Traceback length

decoded1 = vitdec(ncode,trel,tblen,'cont','hard'); %Hard decision

% Use unquantized decisions.
```

```
ucode = 1-2*ncode; % +1 & -1 represent zero & one, respectively.
decoded2 = vitdec(ucode,trel,tblen,'cont','unquant');

% To prepare for soft-decision decoding, map to decision values.
[x,qcode] = quantiz(1-2*ncode,[-.75 -.5 -.25 0 .25 .5 .75],...
[7 6 5 4 3 2 1 0]); % Values in qcode are between 0 and 2^3-1.
decoded3 = vitdec(qcode',trel,tblen,'cont','soft',3);

% Compute bit error rates, using the fact that the decoder output is delayed by tblen
symbols.
[n1,r1] = biterr(decoded1(tblen+1:end),msg(1:end-tblen));
[n2,r2] = biterr(decoded2(tblen+1:end),msg(1:end-tblen));
[n3,r3] = biterr(decoded3(tblen+1:end),msg(1:end-tblen));
disp(['The bit error rates are:   ',num2str([r1 r2 r3])])

%%      Gibb's Phenomenon

% periodic signal x(t)=-1 for t=(-0.5,0) and x(t)=1 for t=(0,0.5), period T=1
% Fourier coeffs=4/(pi*k)

% t = [-500:500] * 0.001;
t1 = [-0.5:0.001:0.5]*3;t2 = [-0.5:0.0001:0.5];t3 = [-0.5:0.00001:0.5];
t=t1; %time scale
total = zeros(size(t));
for k = 1 : 2 : 200
total = total + (4/pi) * sin(2*pi*k*t) / k;
end
plot(t,total);grid minor;

%%      Power Spectrum Estimation

randn('state',0)
fs = 1000;          % Sampling frequency
t = (0:fs/10)/fs;    % One-tenth of a second worth of samples
A = [1 2];          % Sinusoid amplitudes
f = [150;140];       % Sinusoid frequencies
xn = A*sin(2*pi*f*t) + 0.1*randn(size(t)); % sum of 2 freq + noise
Hs = spectrum.periodogram; % define periodogram object
psd(Hs,xn,'Fs',fs,'NFFT',1024) % compute PSD estimate via periodogram
```

RESULT:

Convolution coding and Viterbi decoding is performed using MATLAB.

Wave Forms:



Power Spectral Density Estimate via Periodogram
Power/frequency (dB/Hz)
Frequency (kHz)
0
-10
-20
-30
-40
-50
-60
-70
-80
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
0.45
0.5

9. DIGITAL MODULATION AND DEMODULATION

AIM:

Write a MATLAB program to Perform Digital modulation and demodulation.

PROGRAM:

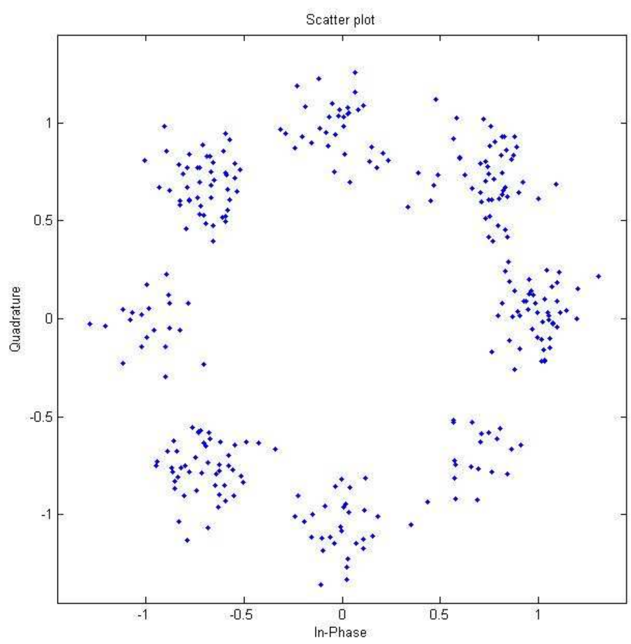
```
%%      Digital Modulation & Demodulation

clc;
clear all;
close all;
M = 8; % Use 16-ary modulation.
Fd = 1; % Assume the original message is sampled
% at a rate of 1 sample per second.
Fs = 3; % The modulated signal will be sampled
% at a rate of 3 samples per second.
x = randint(100,1,M); % Random digital message
% Use M-ary PSK modulation to produce y.
y = dmodce(x,Fd,Fs,'psk',M);
% Add some Gaussian noise.
ynoisy = y + .14*randn(300,1) + .14*j*randn(300,1);
% Create scatter plot from noisy data.
scatterplot(ynoisy,1,0,'b.');
% Demodulate y to recover the message.
z = ddemodce(ynoisy,Fd,Fs,'psk',M);
s = symerr(x,z) % Check symbol error rate.

% Create a random digital message
M = 16; % Alphabet size
x = randint(5000,1,M); % Message signal

% Use 16-QAM modulation.
y = qammod(x,M);

% Transmit signal through an AWGN channel.
ynoisy = awgn(y,15,'measured');

% Create scatter plot from noisy data.
scatterplot(ynoisy);

% Demodulate to recover the message.
z = qamdemod(ynoisy,M);

% Check symbol error rate.
[num,rt]= symerr(x,z)

M = 16; % Alphabet size
x = randint(5000,1,M); % Message signal
```

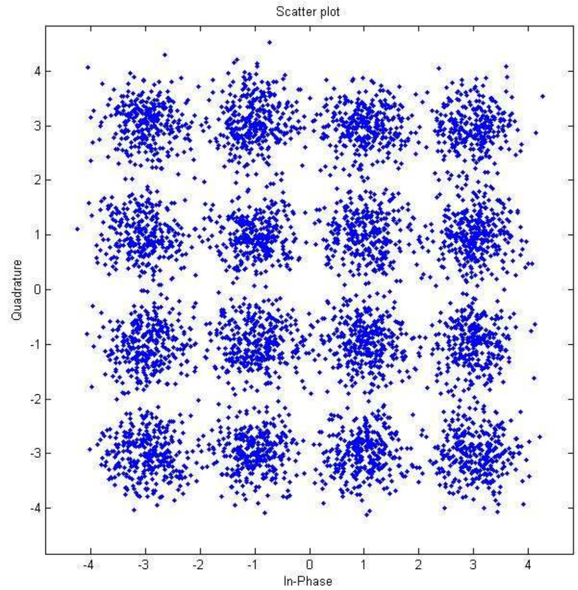
```
Nsamp = 4; % Oversampling rate

% Use 16-QAM modulation.
y = qammod(x,M);

% Follow with rectangular pulse shaping.
ypulse = rectpulse(y,Nsamp);

% Transmit signal through an AWGN channel.
ynoisy = awgn(ypulse,15,'measured');

% Downsample at the receiver.
ydownsamp = intdump(ynoisy,Nsamp);

% Demodulate to recover the message.
z = qamdemod(ydownsamp,M);
scatterplot(x,z);
```

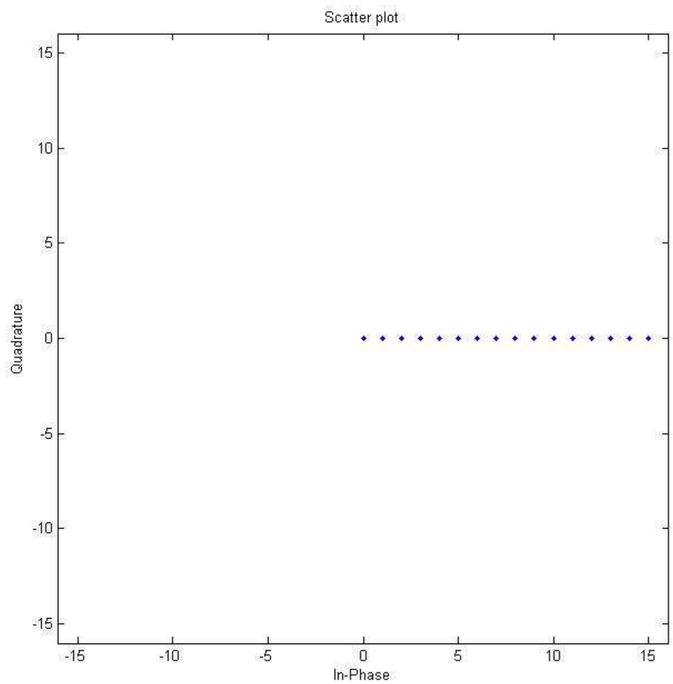
RESULT:

Digital modulation and demodulation is performed using MATLAB.

Wave Forms:




Scatter plot
Quadrature
In-Phase
4
3
2
1
0
-1
-2
-3
-4
-4
-3
-2
-1
0
1
2
3
4


Scatter plot
Quadrature
In-Phase
15
10
5
0
-5
-10
-15
-15
-10
-5
0
5
10
15

10. HUFFMAN SOURCE CODING

AIM:

Write a MATLAB program to Perform Huffman source coding.

PROGRAM:

```
%%      Huffman Source Coding

 %%  Create a Huffman code dictionary using HUFFMANDICT.
 [dict,avglen] = huffmandict([1:6],[.5 .125 .125 .125 .0625 .0625]);
%[dict,avglen] = huffmandict({'x1','x2','x3','x4','x5','x6'},[.5 .125 .125 .125 .0625 .0625]);
         % Define a signal of valid letters.
           sig = [ 2 1 4 2 1 1 5 4 ];
         % Encode the signal using the Huffman code dictionary.
           sig_encoded = huffmanenco(sig,dict);
%codetable = cell2mat(dict);   bits=ceil(log2(17));
          symbols = [1:6] % Alphabet vector
          prob = [.3 .3 .2 .1 .1 0.0] % Symbol probability vector
          [dict,avglen] = huffmandict(symbols,prob)

          prob(prob==0) = [];       % remove zero entries in prob
          H = -sum(prob.*log2(prob));eff1=H/avglen*100;
                    bits=ceil(log2(length(symbols)));eff2=H/bits*100;
           % Pretty print the dictionary.
          temp = dict;
          for i = 1:length(temp)
          temp{i,2} = num2str(temp{i,2});
          end
          temp;
         letters = [1:6]; % Distinct symbols the data source can produce
         p = [.5 .125 .125 .125 .0625 .0625]; % Probability distribution
         [dict,avglen] = huffmandict(letters,p); % Get Huffman code.
         % Pretty print the dictionary.
          codetable = dict;
          for i = 1:length(codetable)
          codetable{i,2} = num2str(codetable{i,2});
          end
         codetable
         sig  = randsrc(1,20,[letters; p]); % Create data using p.
         comp = huffmanenco(sig,dict);  % Encode the data.
         deco = huffmandeco(comp,dict); % Decode the encoded signal.
         equal = isequal(sig,deco); % Check whether the decoding is correct.
```

RESULT:

Huffman source coding is performed using MATLAB.

OUTPUT :

```
symbols =

     1     2     3     4     5     6


prob =

    0.3000    0.3000    0.2000    0.1000    0.1000         0


dict =

    [1]    [1x2 double]
    [2]    [1x2 double]
    [3]    [1x2 double]
    [4]    [1x4 double]
    [5]    [1x3 double]
    [6]    [1x4 double]


avglen =

    2.3000


codetable =

    [1]    '0'
    [2]    '1  0  0'
    [3]    '1  1  1'
    [4]    '1  1  0'
    [5]    '1  0  1  1'
    [6]    '1  0  1  0'
```

11. OVERALL COMMUNICATION SYSTEM

AIM:

Write a MATLAB program to Perform Overall communication system Simulation.

PROGRAM:

```
%%      Overall Communication System Simulation

% This example, described in the Getting Started chapter of the
% Communications Toolbox documentation, aims to solve the following
% problem:
%
% Modify the filtering example (COMMDOC_RRC) so that it
% includes convolutional coding and decoding, given the
% constraint lengths and generator polynomials of the
% convolutional code.

% Copyright 1996-2004 The MathWorks, Inc.
% $Revision: 1.1.6.2 $ $Date: 2004/03/30 13:01:48 $


%%      Setup
% Define parameters.
M = 16; % Size of signal constellation
k = log2(M); % Number of bits per symbol
n = 5e5; % Number of bits to process
nsamp = 4; % Oversampling rate

%%      Signal Source
% Create a binary data stream as a column vector.
x = randint(n,1); % Random binary data stream

% Plot first 40 bits in a stem plot.
stem(x(1:40),'filled');
title('Random Bits');
xlabel('Bit Index'); ylabel('Binary Value');

%%      Channel Encoder
% Define a convolutional coding trellis and use it
% to encode the binary data.
t = poly2trellis([5 4],[23 35 0; 0 5 13]); % Trellis
code = convenc(x,t); % Encode.
coderate = 2/3;

%%      Bit-to-Symbol Mapping
% Convert the bits in x into k-bit symbols, using
% Gray coding.
 % A. Define a vector for mapping bits to symbols using
```

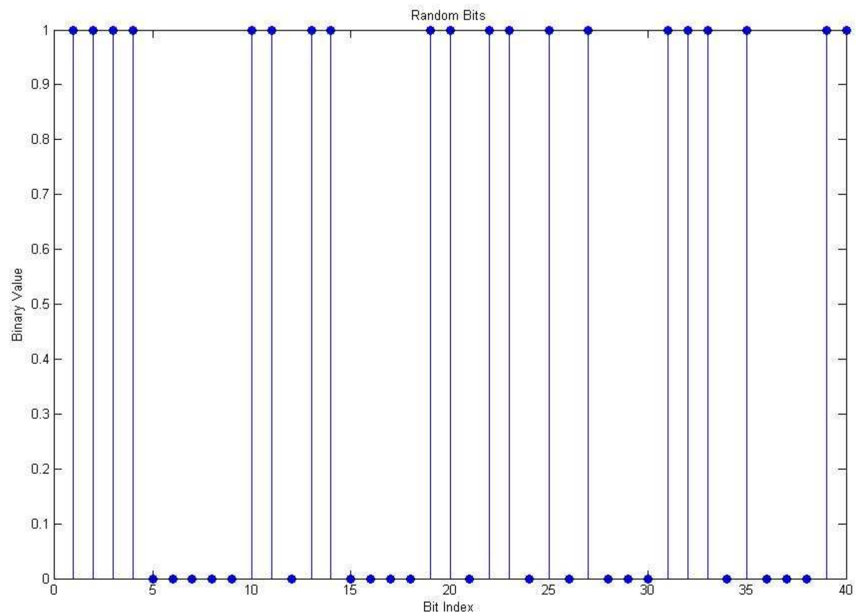
```
% Gray coding. The vector is specific to the arrangement
% of points in a 16-QAM constellation.
mapping = [0 1 3 2 4 5 7 6 12 13 15 14 8 9 11 10].';

% B. Do ordinary binary-to-decimal mapping.
xsym = bi2de(reshape(code,k,length(code)/k).','left-msb');

% C. Map from binary coding to Gray coding.
xsym = mapping(xsym+1);

%%      Stem Plot of Symbols
% Plot first 10 symbols in a stem plot.
figure; % Create new figure window.
stem(xsym(1:10));
title('Random Symbols');
xlabel('Symbol Index'); ylabel('Integer Value');

%%      Modulation
% Modulate using 16-QAM.
y = qammod(xsym,M);

%%      Filter Definition
% Define filter-related parameters.
filtorder = 40; % Filter order
delay = filtorder/(nsamp*2); % Group delay (# of input samples)
rolloff = 0.25; % Rolloff factor of filter

% Create a square root raised cosine filter.
rrcfilter = rcosine(1,nsamp,'fir/sqrt',rolloff,delay);

% Plot impulse response.
figure; impz(rrcfilter,1);

%%      Transmitted Signal
% Upsample and apply square root raised cosine filter.
ytx = rcosflt(y,1,nsamp,'filter',rrcfilter);

% Create eye diagram for part of filtered signal.
eyediagram(ytx(1:2000),nsamp*2);

%%      Channel
% Send signal over an AWGN channel.
EbNo = 10; % In dB
snr = EbNo + 10*log10(k*coderate)-10*log10(nsamp);
ynoisy = awgn(ytx,snr,'measured');

%%      Received Signal
% Filter received signal using square root raised cosine filter.
yrx = rcosflt(ynoisy,1,nsamp,'Fs/filter',rrcfilter);
yrx = downsample(yrx,nsamp); % Downsample.
```

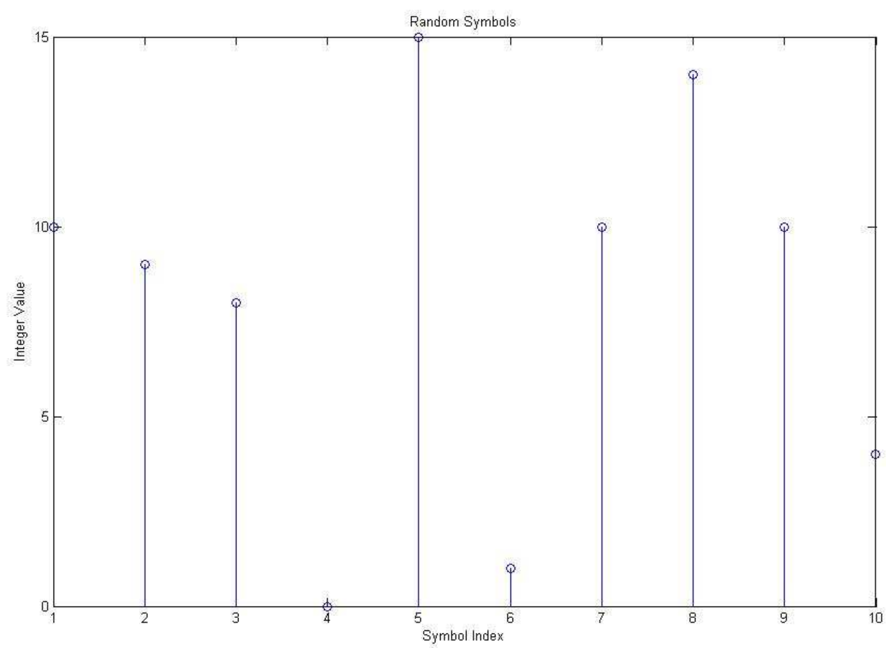
```
yrx = yrx(2*delay+1:end-2*delay); % Account for delay.

%%        Scatter Plot
% Create scatter plot of received signal before and
% after filtering.
h = scatterplot(sqrt(nsamp)*ynoisy(1:nsamp*5e3),nsamp,0,'g.');
hold on;
scatterplot(yrx(1:5e3),1,0,'kx',h);
title('Received Signal, Before and After Filtering');
legend('Before Filtering','After Filtering');
axis([-5 5 -5 5]); % Set axis ranges.

%%        Demodulation
% Demodulate signal using 16-QAM.
zsym = qamdemod(yrx,M);

%%        Symbol-to-Bit Mapping
% Undo the bit-to-symbol mapping performed earlier.

% A. Define a vector that inverts the mapping operation.
[dummy demapping] = sort(mapping);
% Initially, demapping has values between 1 and M.
% Subtract 1 to obtain values between 0 and M-1.
demapping = demapping - 1;

% B. Map between Gray and binary coding.
zsym = demapping(zsym+1);

% C. Do ordinary decimal-to-binary mapping.
z = de2bi(zsym,'left-msb');
% Convert z from a matrix to a vector.
z = reshape(z.',prod(size(z)),1);

%%        Channel Decoder
% Decode the convolutional code.
tb = 16; % Traceback length for decoding
z = vitdec(z,t,tb,'cont','hard'); % Decode.

%%        BER Computation
% Compare x and z to obtain the number of errors and
% the bit error rate. Take the decoding delay into account.
decdelay = 2*tb; % Decoder delay, in bits
[number_of_errors,bit_error_rate] = ...
   biterr(x(1:end-decdelay),z(decdelay+1:end))
```

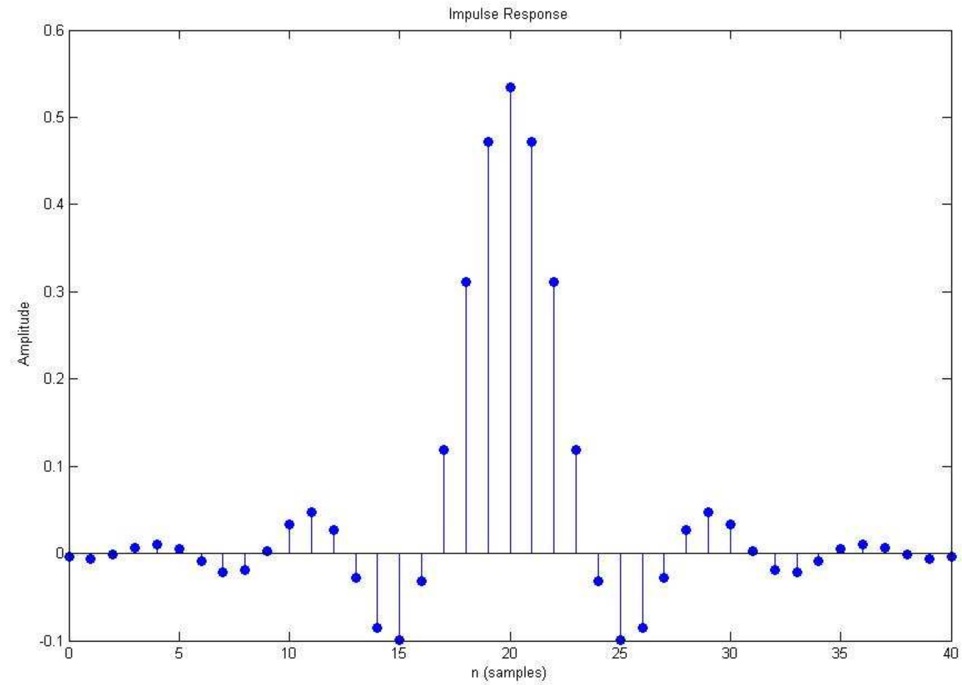
RESULT:

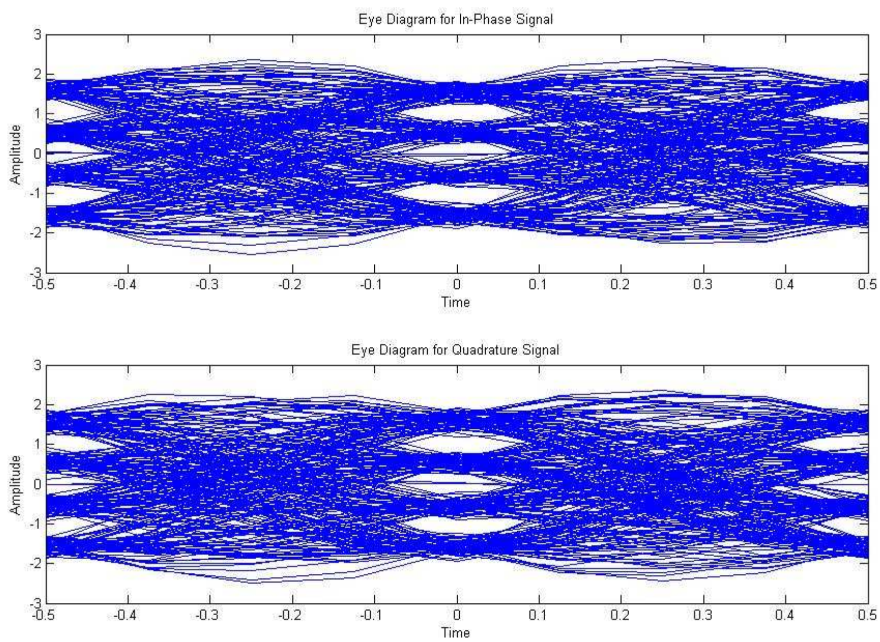
Overall communication system simulation is performed using MATLAB.

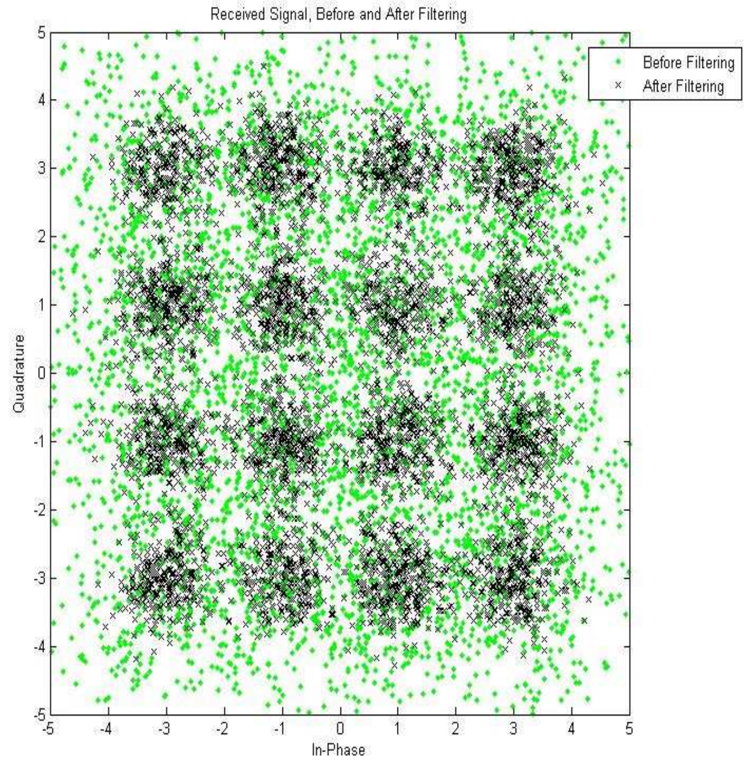
Wave Forms:


Random Bits
Binary Value
Bit Index
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
0
5
10
15
20
25
30
35
40


Random Symbols
Integer Value
Symbol Index
0
5
10
15
1
2
3
4
5
6
7
8
9
10


Impulse Response
Amplitude
n (samples)
0.6
0.5
0.4
0.3
0.2
0.1
0
-0.1
0
5
10
15
20
25
30
35
40


Eye Diagram for In-Phase Signal
Amplitude
Time
Eye Diagram for Quadrature Signal
Amplitude
Time
3
2
1
0
-1
-2
-3
-0.5
-0.4
-0.3
-0.2
-0.1
0
0.1
0.2
0.3
0.4
0.5


Received Signal, Before and After Filtering
Before Filtering
After Filtering
Quadrature
In-Phase
5
4
3
2
1
0
-1
-2
-3
-4
-5
-5
-4
-3
-2
-1
0
1
2
3
4
5

12. FADING CHANNEL SIMULATION

AIM:

Write a MATLAB program to Perform Fading channel Simulation.

PROGRAM:

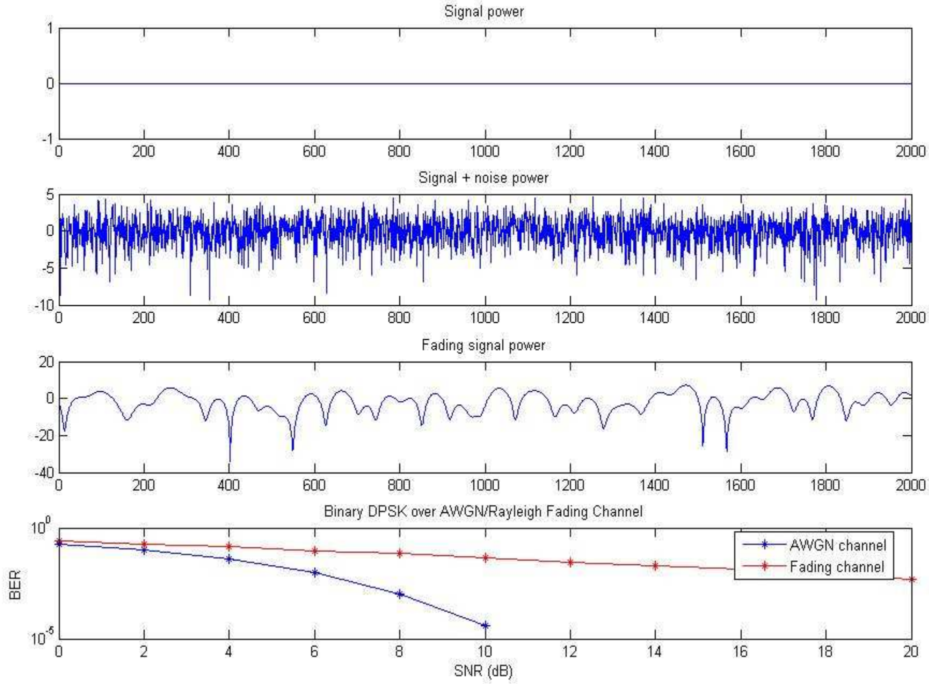
```
%%      Fading Channel Simulation

c = rayleighchan(1/10000,100); % Create channel
c % Display all properties of the channel object.
db = 10; % noise level
sig = j*ones(2000,1); % Signal
% sig = pskmod(randint(1000,1),4); % BPSK symbols

noisysig = awgn(sig,db,'measured'); % add noise
fadingsig = filter(c,sig); % Pass signal through fading channel.
noisyfadingsig = awgn(fadingsig,db,'measured'); % fading + noise
figure,
subplot(4,1,1);plot(20*log10(abs(sig))); % Plot power of signal, versus sample number.
title('Signal power');

subplot(4,1,2);plot(20*log10(abs(noisysig))); % Plot power of noisy-signal
title('Signal + noise power');

subplot(4,1,3);plot(20*log10(abs(fadingsig))); % Plot power of faded signal
title('Fading signal power');

subplot(4,1,4);plot(20*log10(abs(noisyfadingsig))); % Plot power of noisy-fading-signal
title('Fading signal + noise power');

%%      BER of DBPSK over AWGN/frequency-flat Rayleigh fading channel

% Create Rayleigh fading channel object.
chan = rayleighchan(1/10000,100);

% Generate data and apply fading channel.
M = 2; % DBPSK modulation order
tx = randint(50000,1,M);  % Random bit stream
dpskSig = dpskmod(tx,M);  % DPSK signal
fadedSig = filter(chan,dpskSig); % Effect of channel

% Compute error rate for different values of SNR.

SNR = 0:2:20; % Range of SNR values, in dB.

for n = 1:length(SNR)
   rxSig1 = awgn(dpskSig,SNR(n)); % Add Gaussian noise.
   rxSig2 = awgn(fadedSig,SNR(n)); % Add Gaussian noise after fading
```

```
   rx1 = dpskdemod(rxSig1,M); % Demodulate.
   rx2 = dpskdemod(rxSig2,M); % Demodulate.

   % Compute error rate.
   % Ignore first sample because of DPSK initial condition.
   [nErrors1, BER1(n)] = biterr(tx(2:end),rx1(2:end));
   [nErrors2, BER2(n)] = biterr(tx(2:end),rx2(2:end));

end

% Compute theoretical performance results, for comparison.
BERtheory = berfading(SNR,'dpsk',M,1);

% Plot BER results.
semilogy(SNR,BER1,'b*-',SNR,BER2,'r*-');legend('AWGN channel','Fading channel');

% semilogy(SNR,BERtheory,'b-',SNR,BER1,'r*',SNR,BER2,'r*');
% legend('Theoretical BER','Empirical BER');

xlabel('SNR (dB)'); ylabel('BER');
title('Binary DPSK over AWGN/Rayleigh Fading Channel');

%%      Working with Delays

M = 2; % DQPSK modulation order
bitRate = 50000;

% Create Rayleigh fading channel object.
ch = rayleighchan(1/bitRate,4,[0 0.5/bitRate],[0 -10]);
delay = ch.ChannelFilterDelay;

tx = randint(50000,1,M); % Random bit stream
dpskSig = dpskmod(tx,M); % DPSK signal
fadedSig = filter(ch,dpskSig); % Effect of channel
rx = dpskdemod(fadedSig,M); % Demodulated signal

% Compute bit error rate, taking delay into account.
% Remove first sample because of DPSK initial condition.
tx = tx(2:end); rx = rx(2:end);
% Truncate to account for channel delay.
tx_trunc = tx(1:end-delay); rx_trunc = rx(delay+1:end);
[num,ber] = biterr(tx_trunc,rx_trunc) % Bit error rate

%%
```

RESULT:

Fading channel simulation is performed using MATLAB.

Wave Forms:



Signal power
Signal + noise power
Fading signal power
Binary DPSK over AWGN/Rayleigh Fading Channel
AWGN channel
Fading channel
BER
SNR (dB)