YANKEE BUSH SOFTWARE LLC

Low Pass & High Pass Filter Design

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Take

Com
s(n).
rand

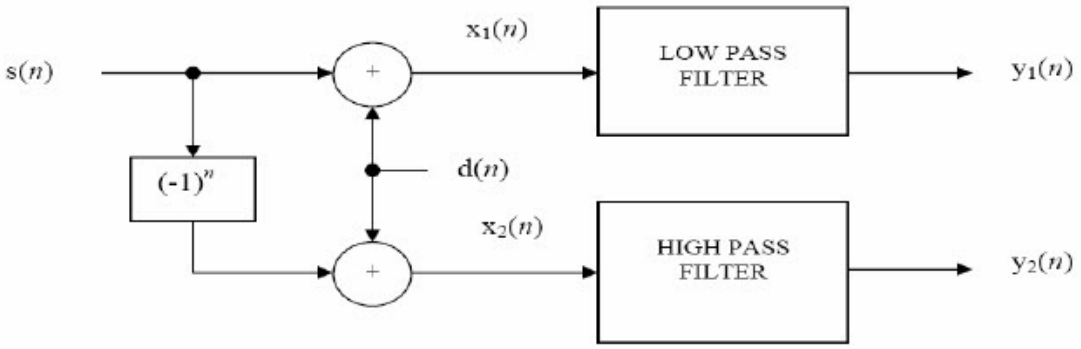
s(n)

Problem Description

Take a block of length L = 1000 from the sequence,

$$s(n) = 0.4^n [\cos(\pi n/4) + \sin(3\pi n/4)]u(n)$$

Configure the following system to pass a select band of frequencies from the sequence $s(n)$. Here, $u(n)$ is the unit step sequence and $d(n)$ is a sequence drawn from uniform random noise lying between [-1,1].



Design the filters to achieve the following:

1. Pass frequencies in the range from $[0, \frac{\pi}{3}]$ using the lowpass filter
2. Pass frequencies in the range from $[\frac{\pi}{2}, \pi]$ using the highpass filter

Plot the following sequences.

1. $s(n)$, $d(n)$, $y_1(n)$, $y_2(n)$: length of each sequence = 1000
2. $S(k)$, $D(k)$, $Y_1(k)$, $Y_2(k)$: 1024-point DFT of each sequence (magnitude, phase)

Matlab(R) Source Code

```
clc;
clear all;
% unit step function
u=ones(1000,1);
% input sequence
for n=1:1000
        s(n)=0.4^n*(cos(pi*n/4)+sin(3*pi*n/4))*u(n);
end
% noise sequence
d=randn(1,1000);
%d2=randsrc(1,1000,[-1,1]);
x1=d+s;
%x2=d2+s;
% butterworth       low pass filter design
[n,w]=buttord(0.3,0.4,4,60);
[b a]=butter(n,w);
y1=filter(b,a,x1);
%freqz(b,a);
for n=1:1000
        xx(n)=((-1)^n)*s(n);
end xx2=xx+d;
```

```
%high pass filter design
[nh wh]=buttord(0.5,0.12,4,60);
[b a]=butter(nh,wh);
y2=filter(b,a,xx2);
%freqz(b,a);
% plotting of sequences s(n),y1(n),y2(n),d(n)
figure(1)
plot(s);
title('input sequence s(n)');
xlabel('length of sequence');
ylabel('magnitude');
hold on;

figure(2)
plot(d);
title('noise sequence');
xlabel('length of sequence');
ylabel('magnitude');
hold on;

figure(3) plot(y1)
title('low pass filtered output sequence y1(n)');
xlabel('length of sequence');
```

```
ylabel('magnitude');
hold on;

figure(4)
plot(y2)
title('high pass filtered sequence y2(n)');
xlabel('length of sequence');
ylabel('magnitude');
hold on;
% calculation of 1024 point dft of sequences s(n),y1(n),y2(n),d(n) and
% plotting of magnitude and phase plot of dft sequences

% input sequence sk=fft(s,1024);
pss= sk.*conj(sk)/1024; f = 1000*(0:512)/1024;
figure(5)
plot(f,pss(1:513))
title('Frequency content of s')
xlabel('frequency (Hz)')
hold on;
psk = unwrap(angle(sk));
f1 = (0:length(sk)-1)'/length(sk)*1000;
figure(6)
plot(f1,psk)
```

```
title('phase plot s')
ylabel('phase (radian)')
hold on;
% noise sequence
dk=fft(d,1024);
dss=dk.*conj(dk)/1024;
figure(7)
plot(f,dss(1:513))
title('Frequency content of d')
xlabel('frequency (Hz)')
hold on;
dsk = unwrap(angle(dk));
f1 = (0:length(dk)-1)'/length(dk)*1000;
figure(71)
plot(f1,dsk)
title('phase plot d')
ylabel('phase (radian)')
hold on;
% low pass filtered output
y1(n) y1k=fft(y1,1024);
Py1 = y1k.* conj(y1k) / 1024;
p = unwrap(angle(y1k));
```

```
f = 1000*(0:512)/1024;
figure(8)
plot(f,Py1(1:513))
title('Frequency content of y1')
xlabel('frequency (Hz)')
hold on;
f1 = (0:length(y1k)-1)'/length(y1k)*1000;
figure(9)
plot(f1,p)
title('phase plot y1')
ylabel('phase (radian)')
hold on;
% high pass filtered output
y2k=fft(y2,1024);
Py2 = y2k.* conj(y2k) / 1024;
p2 = unwrap(angle(y2k));
f = 1000*(0:512)/1024;
figure(10)
plot(f,Py2(1:513))
title('Frequency content of y2')
xlabel('frequency (Hz)')
hold on;
```

```
f1 = (0:length(y2k)-1)'/length(y2k)*1000;

figure(11)

plot(f1,p2)

title('phase plot y2')

ylabel('phase (radian)')

xlabel('frequency(Hz)');

hold on;

hold off;
```

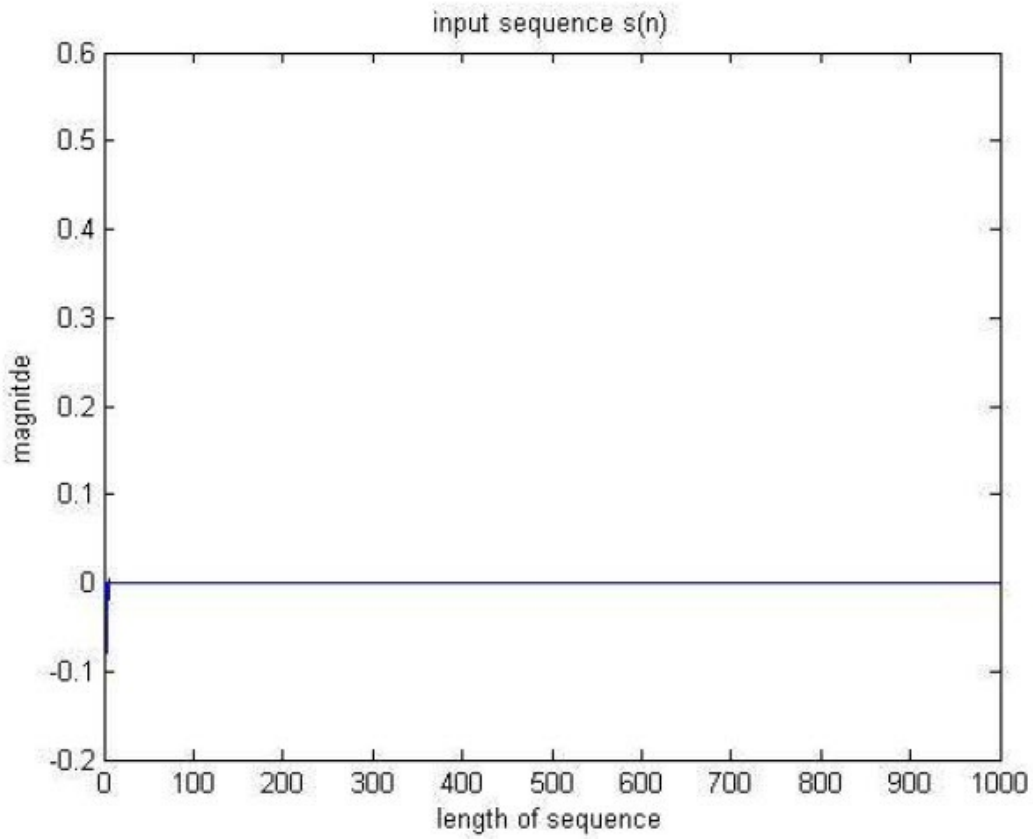
INP

magnitde

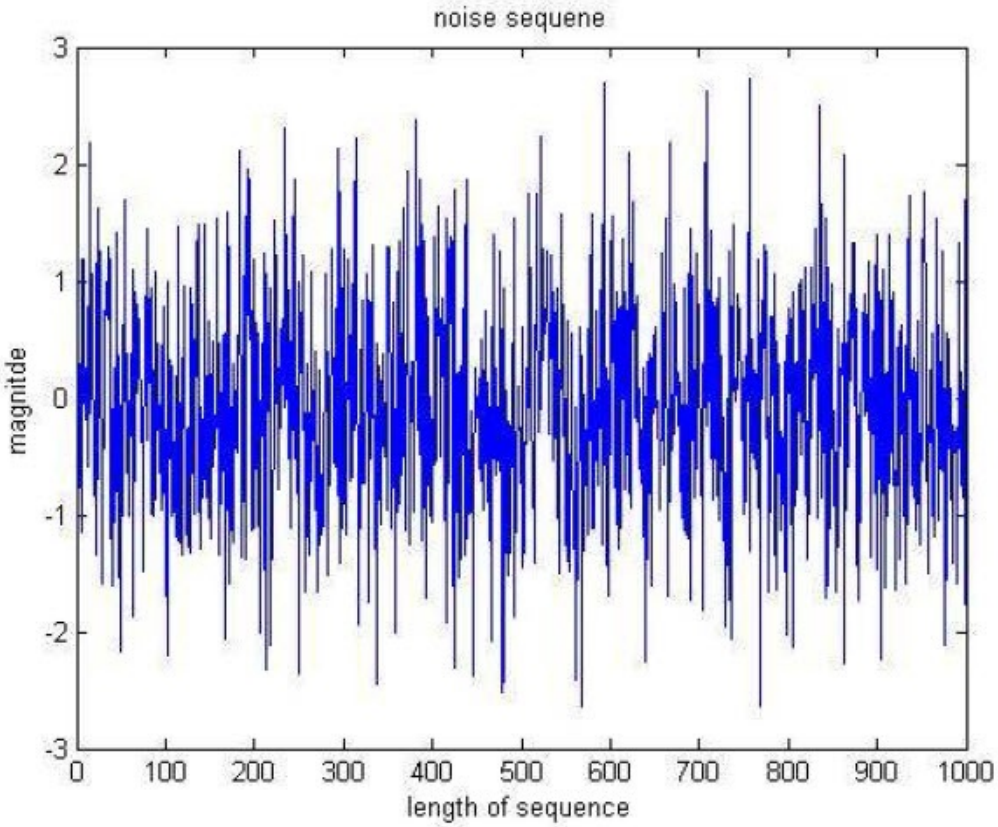
Simulations

Nois

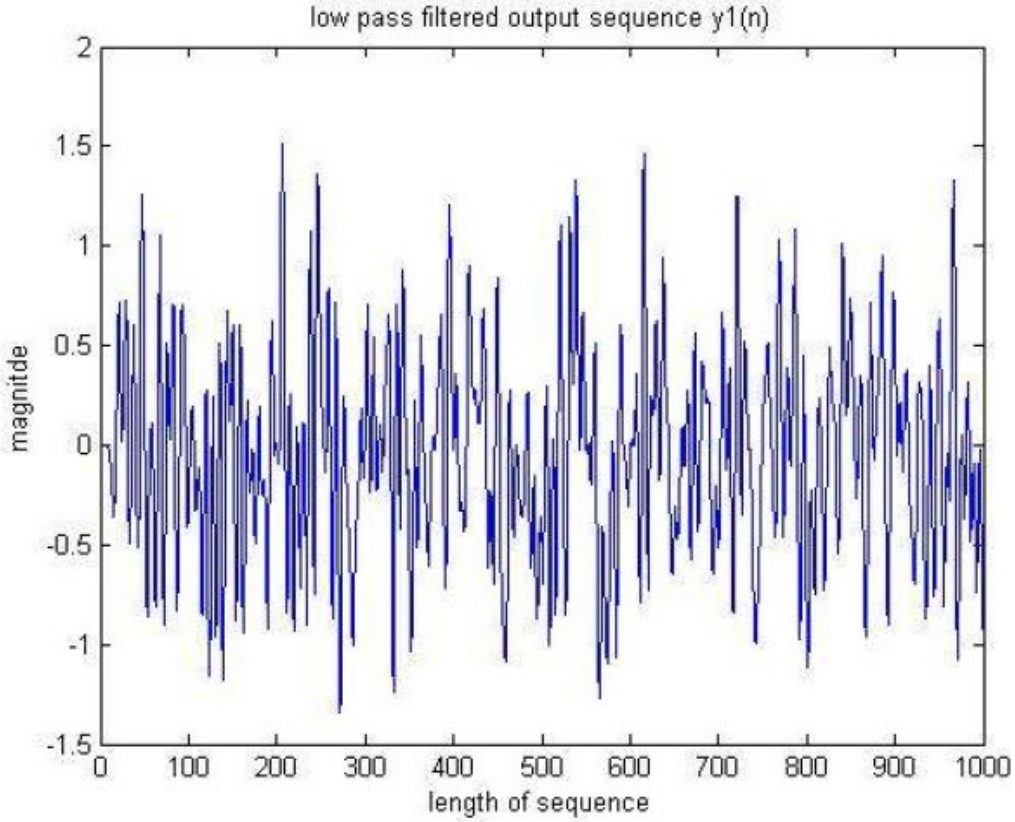
INPUT SEQUENCE s(n) (length of input sequence =1000)



Noise Sequence (length of sequence =1000)



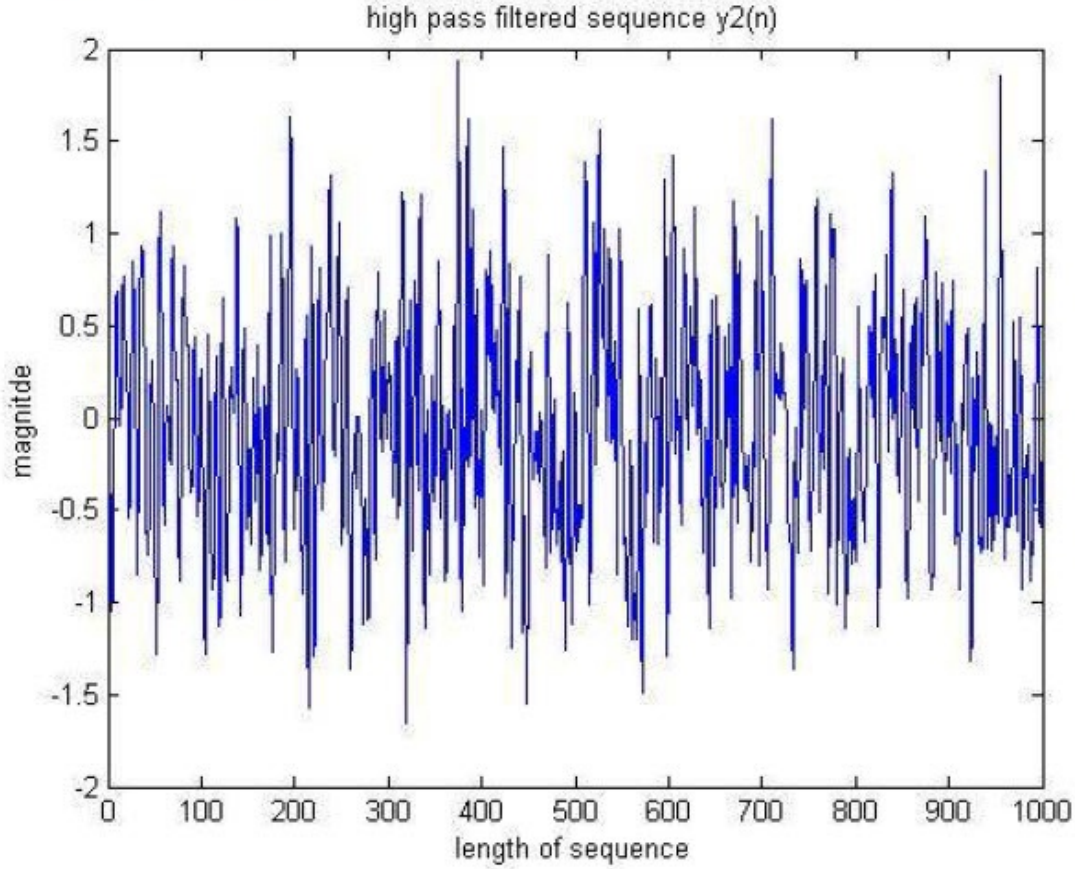
Y1(n) (length of sequence =1000)



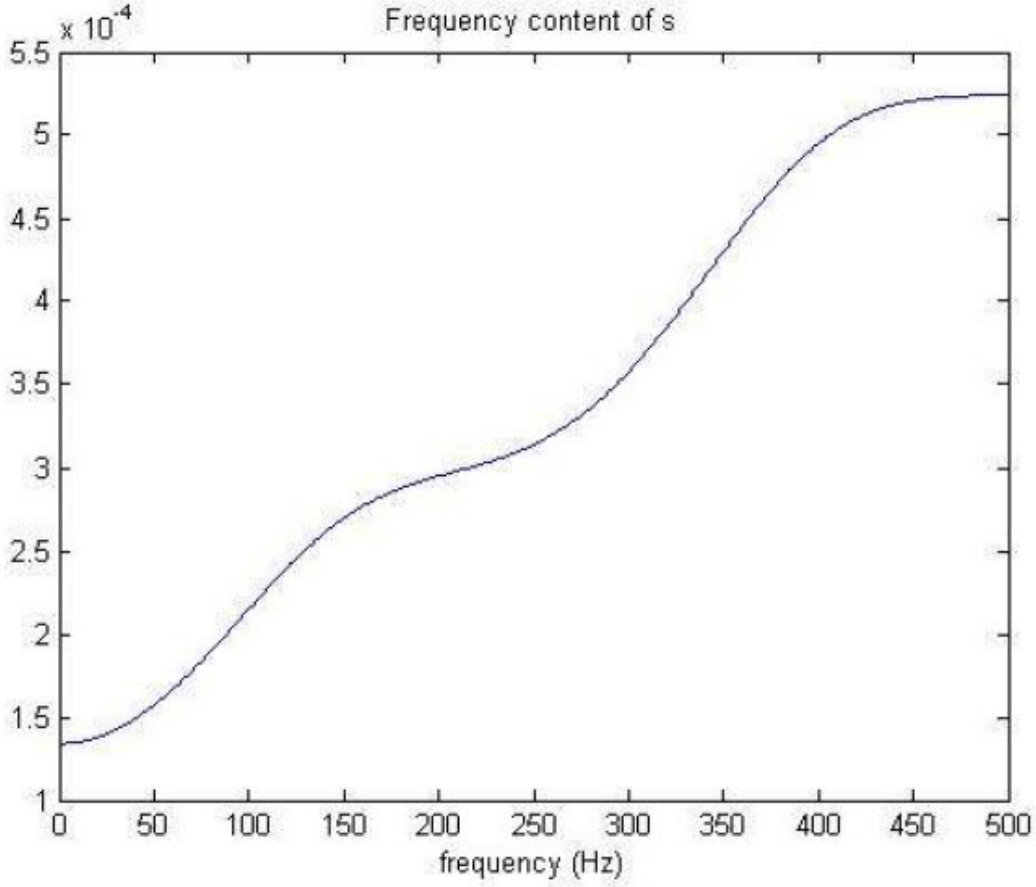
Y2

Y2(n) (length of sequence =1000)

(M
:10

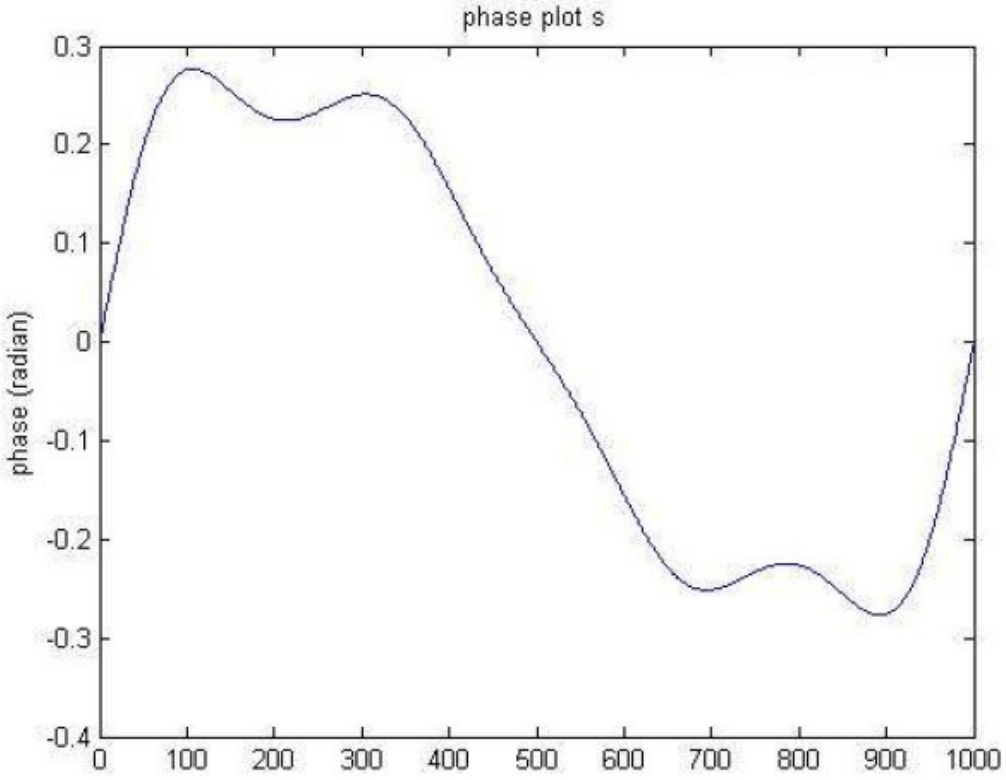


(Magnitude) Frequency Content of S :1024 DFT of each sequence

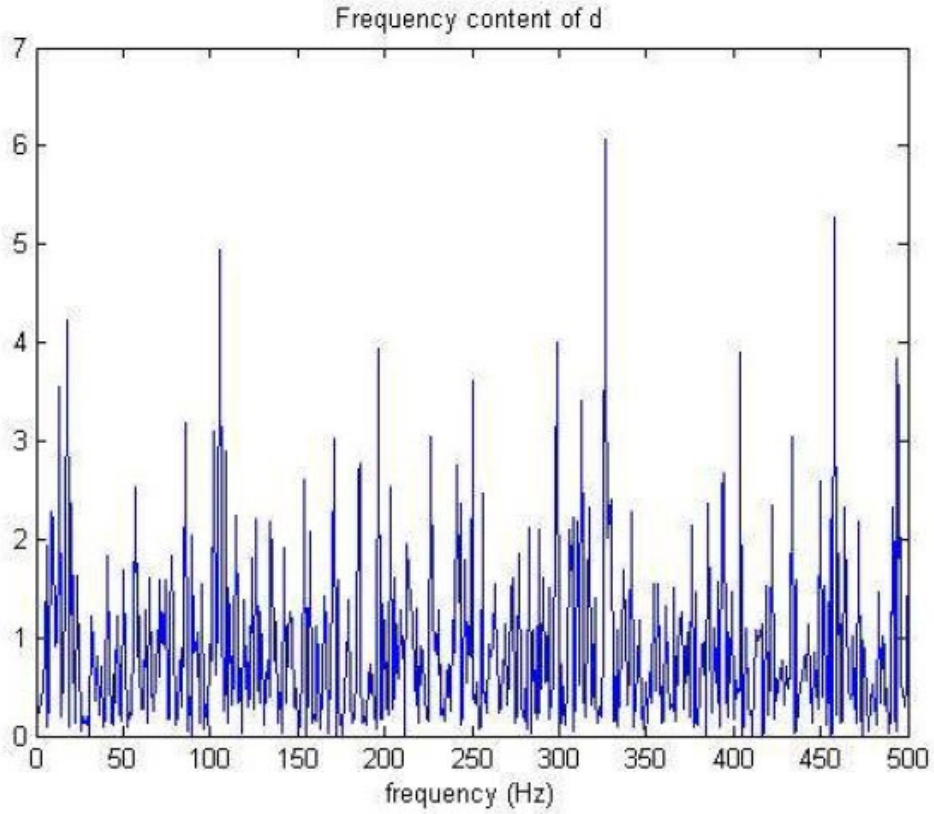


Ph

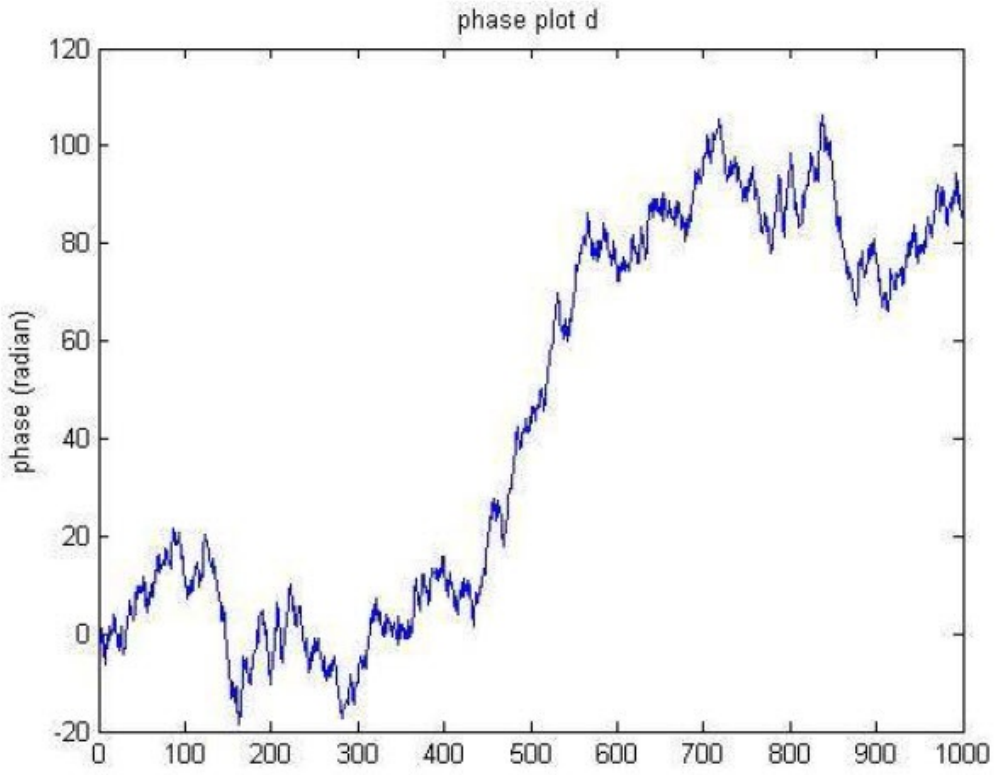
Phase Plot of S:1024 DFT of each sequence



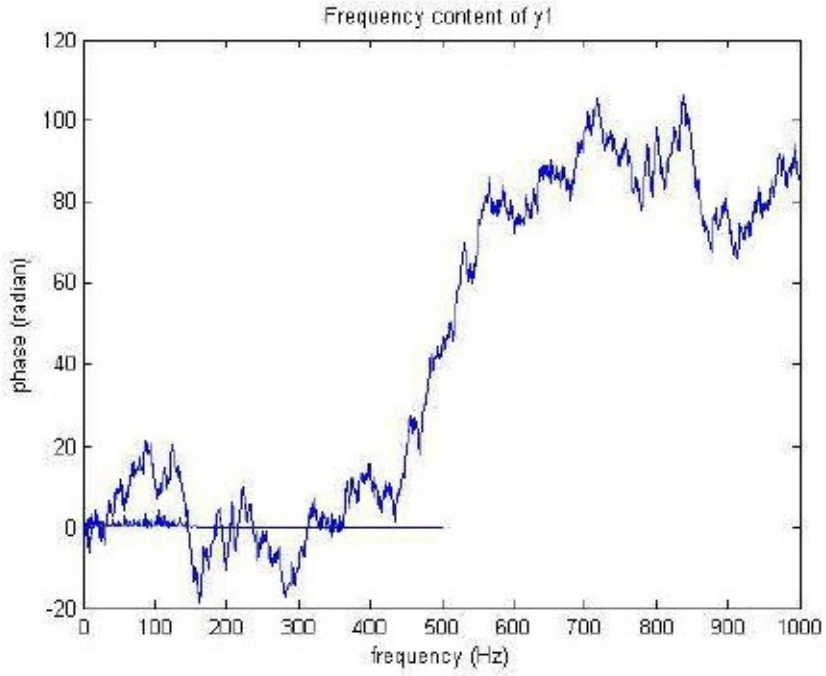
(Magnitude)Frequency Content of D:1024 DFT of each sequence



Phase Plot of D:1024 DFT of each sequence

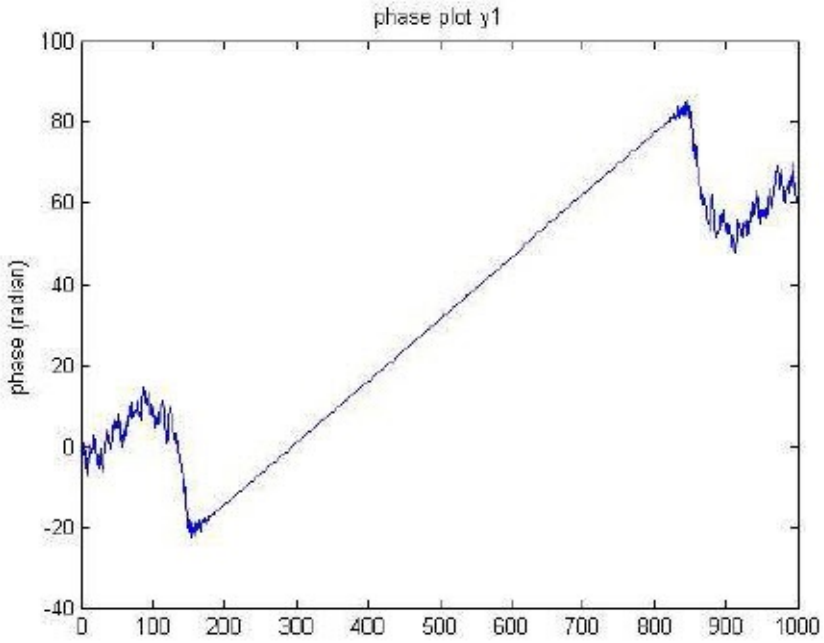


(Magnitude)Frequency Content of y1:1024
DFT of each sequence

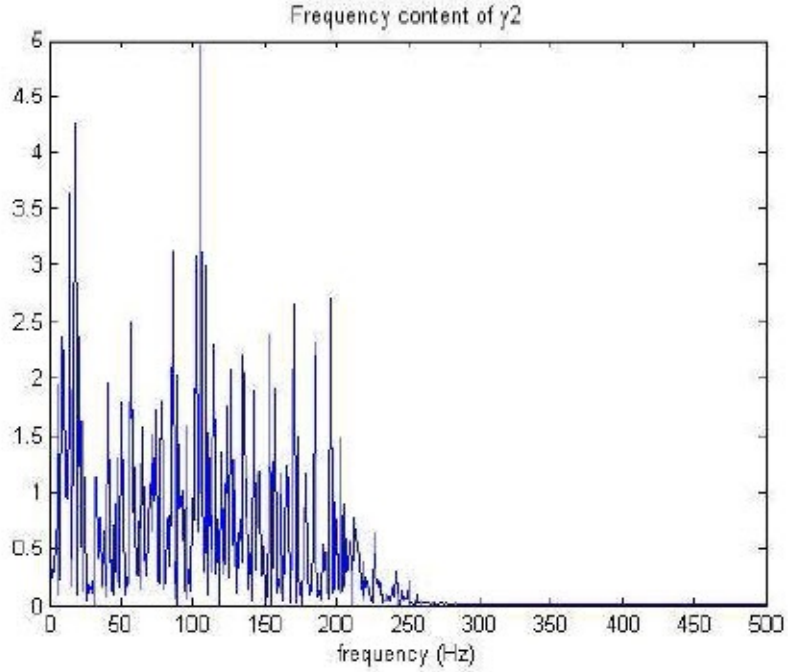


Pha
seq

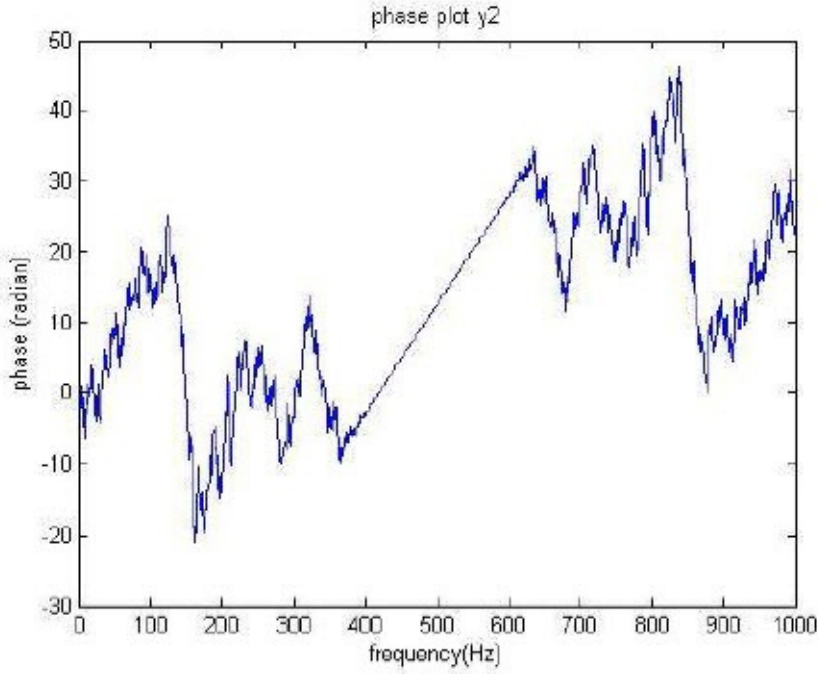
Phase Plot of y1:1024 DFT of each sequence



(Magnitude)Frequency Content of y2:1024
DFT of each sequence



Phase Plot of y2:1024 DFT of each sequence



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