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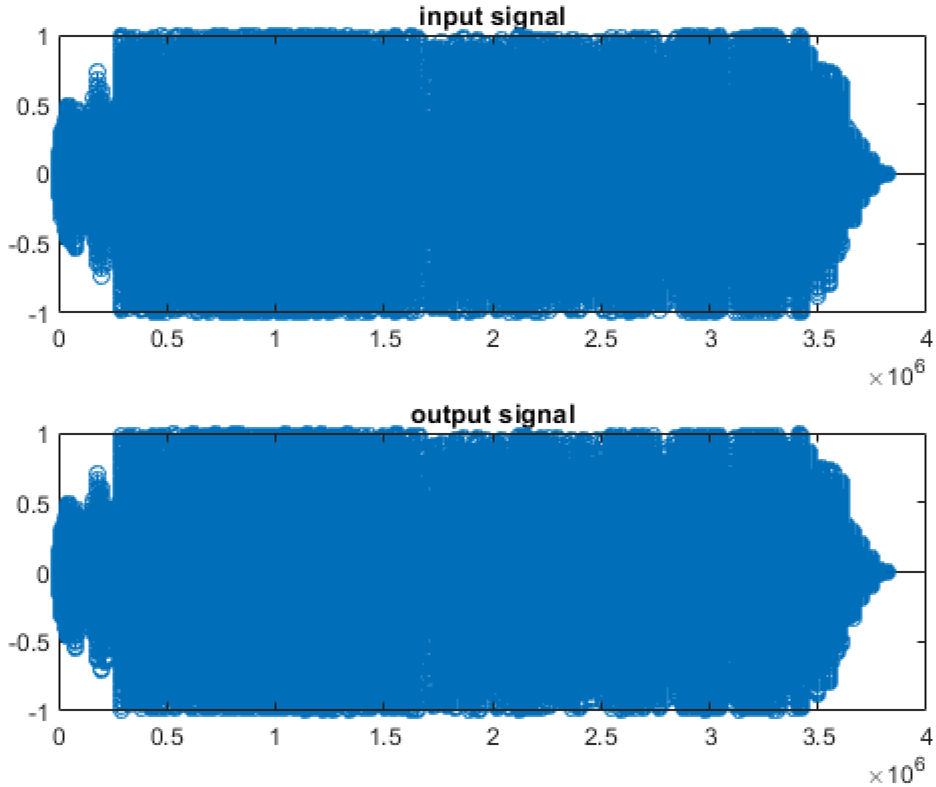
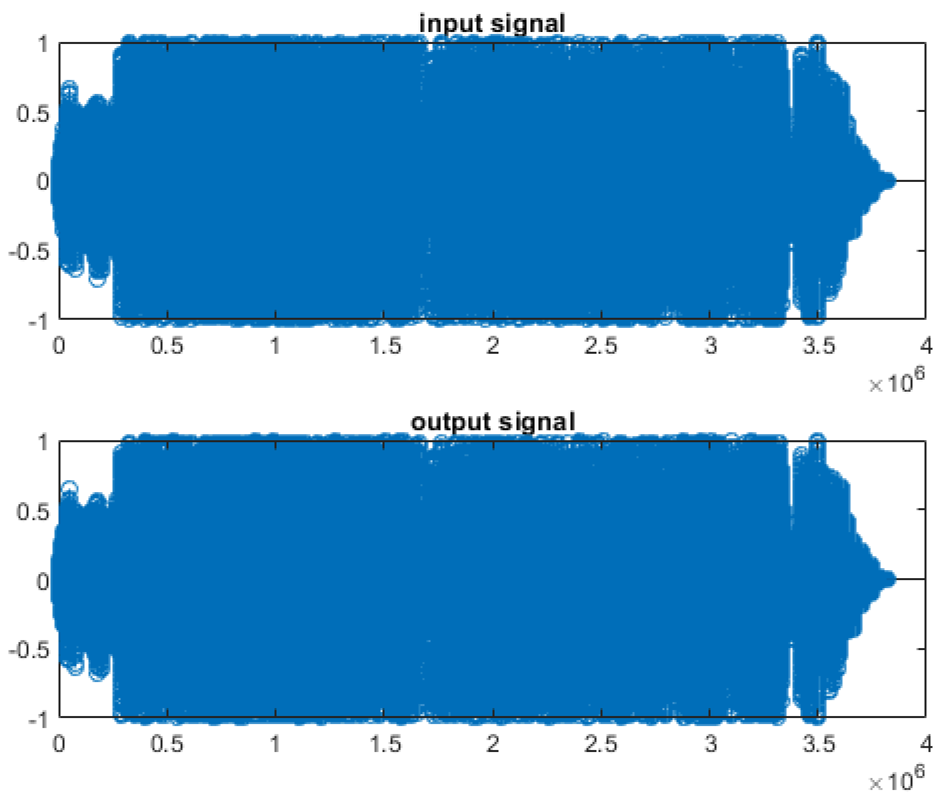
System 1 : $y[n] = 0.5x[n] + 0.5x[n-1]$

Using recursive method.

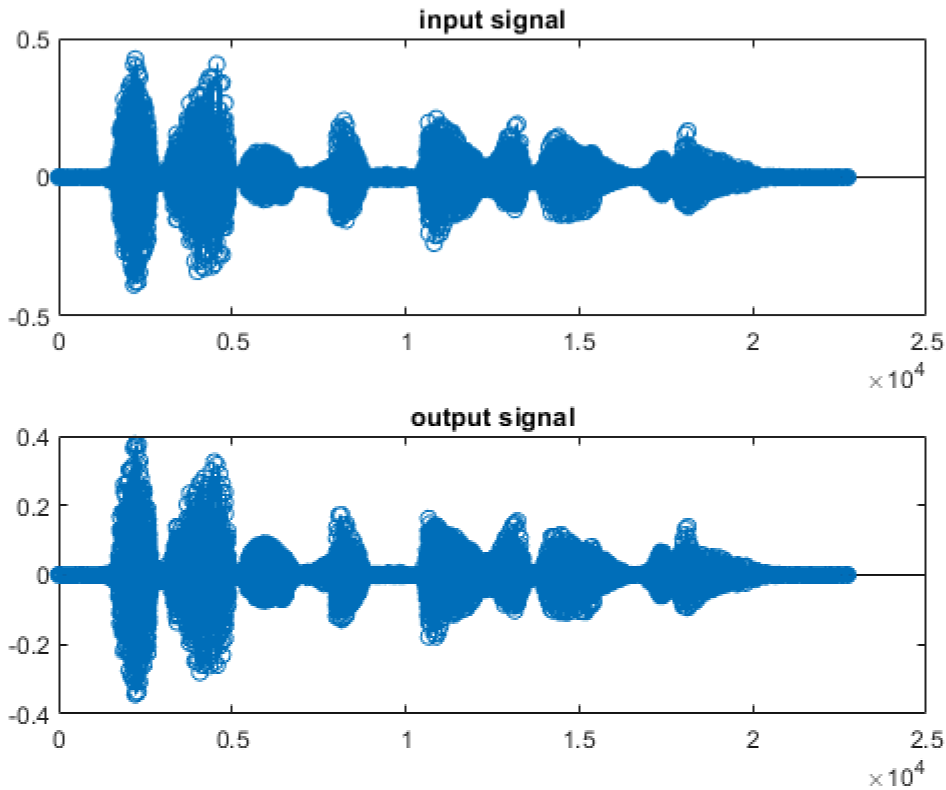
```
[x1,fs_x1] = audioread('inputs/x1.wav');
x1_1 = x1(:,1);
x1_2 = x1(:,2);
[x2,fs_x2] = audioread('inputs/x2.wav');
[x3,fs_x3] = audioread('inputs/x3.wav');
[x4,fs_x4] = audioread('inputs/x4.wav');
[x5,fs_x5] = audioread('inputs/x5.wav');

% Input 1 channel 1 to System 1
    y1_1 = dt_1(x1_1);
    figure();
    subplot 211
    stem(1:length(x1_1),x1_1); title('input signal');
    subplot 212
    stem(1:length(y1_1),y1_1); title('output signal');

% Input 1 channel 2 to System 1
    y1_2 = dt_1(x1_2);
    figure();
    subplot 211
    stem(1:length(x1_2),x1_2); title('input signal');
    subplot 212
    stem(1:length(y1_2),y1_2); title('output signal');
y1_1_both = [y1_1(:),y1_2(:)]; %combine two channels.
% soundsc(x1,fs_x1); %x1 original audio
% soundsc(y1_1_both,fs_x1); %x1 output audio through system 1
```


input signal
1
0.5
0
-0.5
-1
0
0.5
1
1.5
2
2.5
3
3.5
4
×10^6
output signal
1
0.5
0
-0.5
-1
0
0.5
1
1.5
2
2.5
3
3.5
4
×10^6


input signal
1
0.5
0
-0.5
-1
0
0.5
1
1.5
2
2.5
3
3.5
4
×10^6
output signal
1
0.5
0
-0.5
-1
0
0.5
1
1.5
2
2.5
3
3.5
4
×10^6

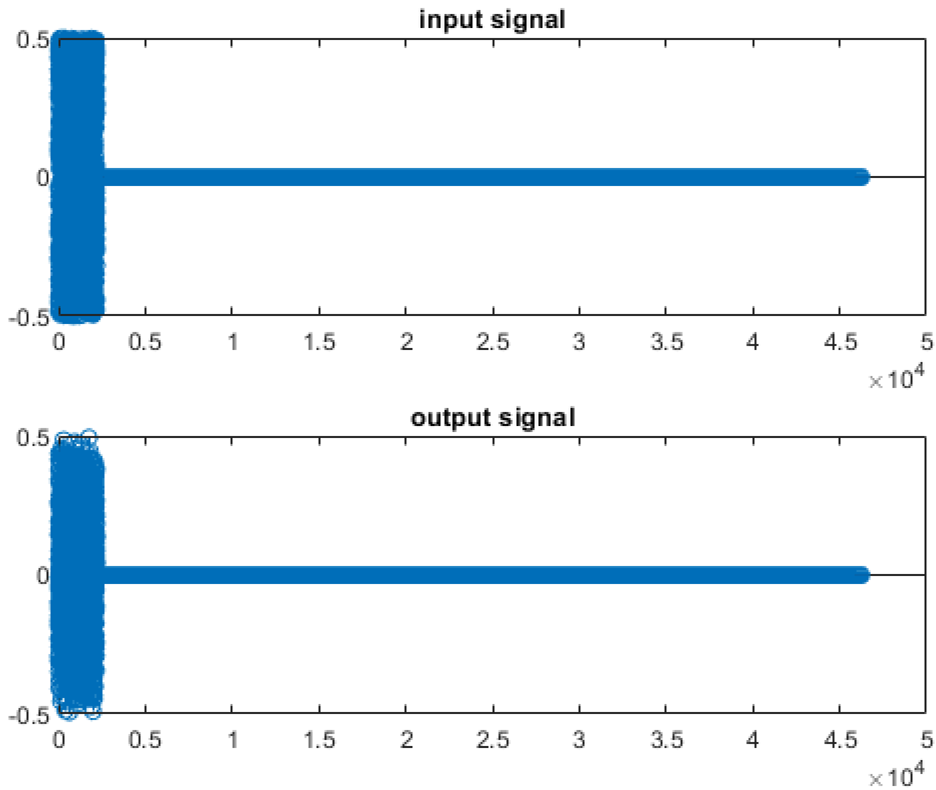
OBSERVATION: Although there is a clear small difference between the input and output of audio x1 as per the system 1 generated graphs, it cannot be sense through bare hearing.

```
% Input 2 to System 1
y1_3 = dt_1(x2);
figure();
subplot 211
stem(1:length(x2),x2); title('input signal');
subplot 212
stem(1:length(y1_3),y1_3); title('output signal');
% soundsc(x2,fs_x2); %x2 original audio
% soundsc(y1_3,fs_x2); %x2 output audio through system 1
```



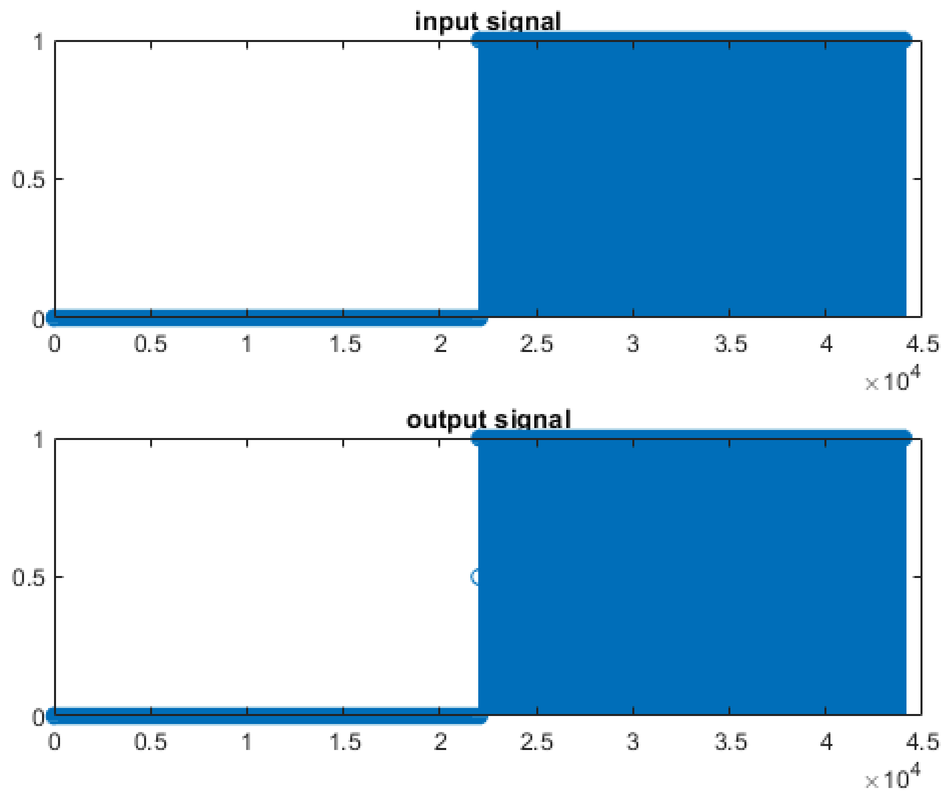
OBSERVATION: Although there is a clear small difference between the input and output of audio x2 as per the system 1 generated graphs, it cannot be sense through bare hearing.

```
% Input 3 to System 1
y1_4 = dt_1(x3);
figure();
subplot 211
stem(1:length(x3),x3); title('input signal');
subplot 212
stem(1:length(y1_4),y1_4); title('output signal');
% soundsc(x3,fs_x3); %x3 original audio
% soundsc(y1_4,fs_x3); %x3 output audio through system 1
```



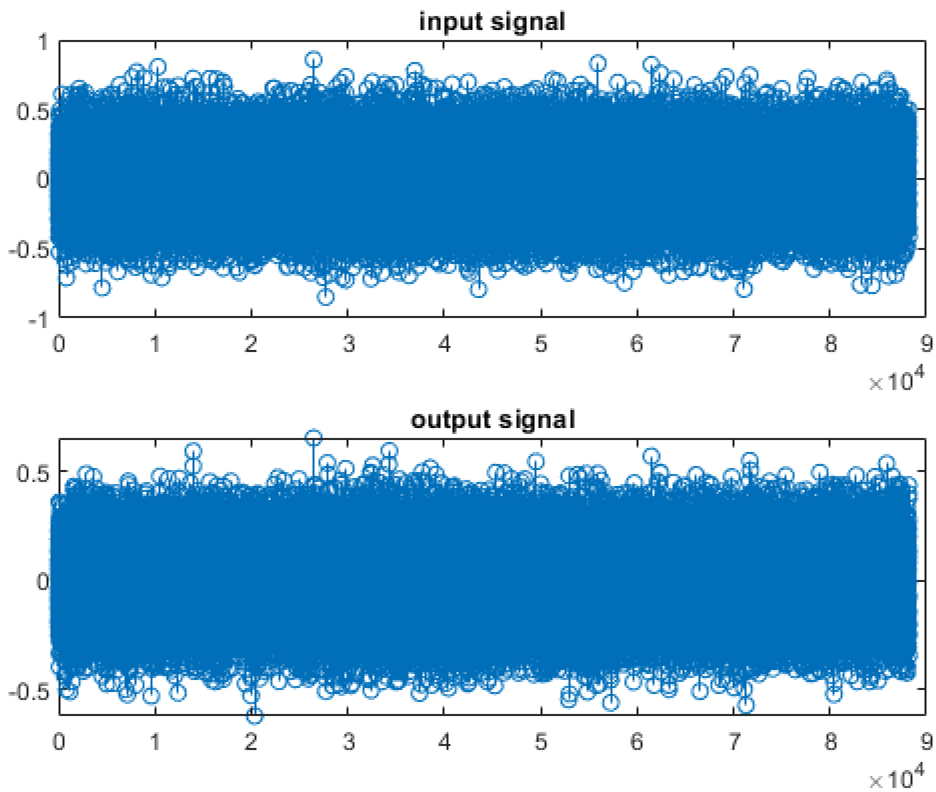
OBSERVATION: Although there is a clear small difference between the input and output of audio x3 as per the system 1 generated graphs, it cannot be sense through bare hearing.

```
% Input 4 to System 1
y1_5 = dt_1(x4);
figure();
subplot 211
stem(1:length(x4),x4); title('input signal');
subplot 212
stem(1:length(y1_5),y1_5); title('output signal');
% soundsc(x4,fs_x4); %x4 original audio
% soundsc(y1_5,fs_x4); %x4 output audio through system 1
```



OBSERVATION: Although there is a clear small difference between the input and output of audio x4 as per the system 1 generated graphs, it cannot be sense through bare hearing.

```
% Input 5 to System 1
y1_6 = dt_1(x5);
figure();
subplot 211
stem(1:length(x5),x5); title('input signal');
subplot 212
stem(1:length(y1_6),y1_6); title('output signal');
% soundsc(x5,fs_x5); %x5 original audio
% soundsc(y1_6,fs_x5); %x5 output audio through system 1
```



OBSERVATION: Although there is a clear small difference between the input and output of audio x5 as per the system 1 generated graphs, it cannot be sense through bare hearing.

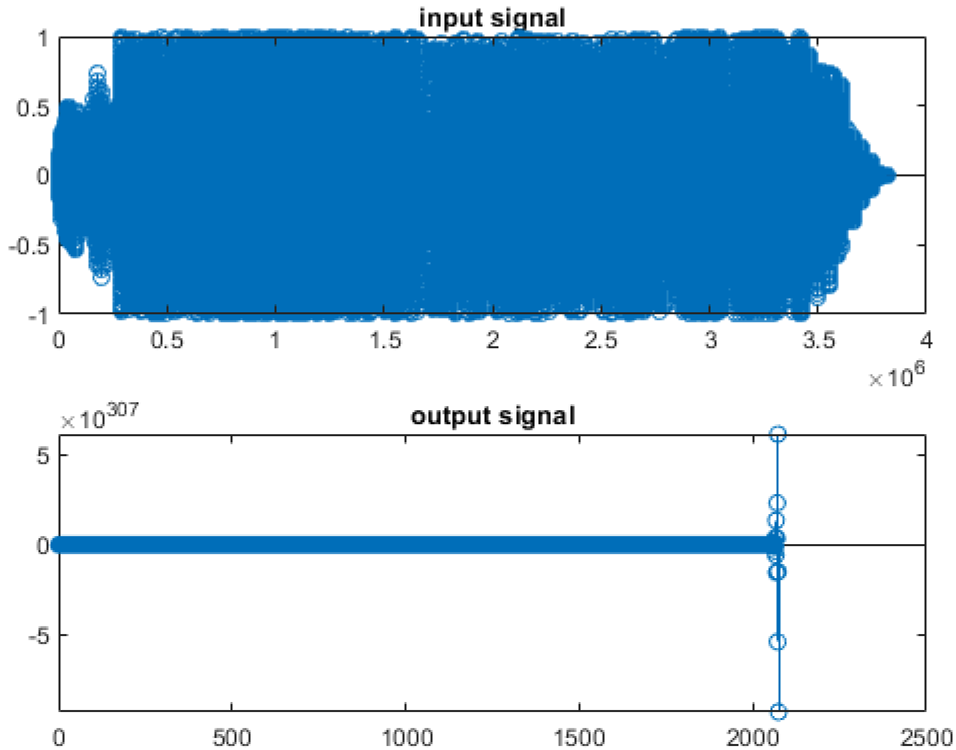
OVERALL, the input and output signals through system 1 greatly oes not differ from each other. The signals are both audible in both ends.

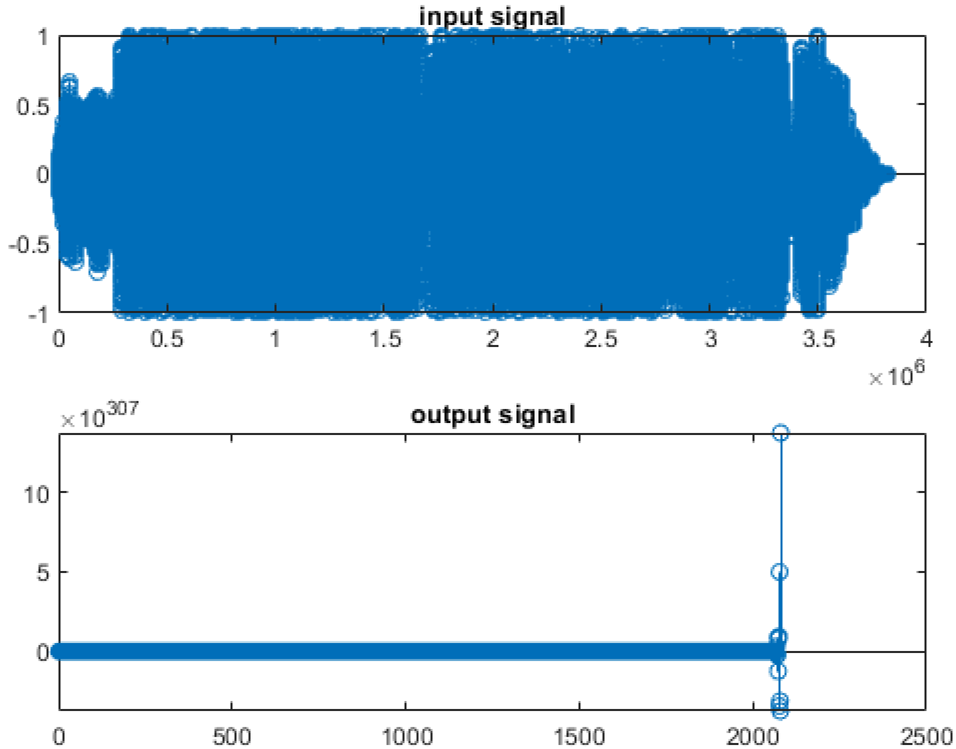
System 2 : $y[n] = x[n] - 2y[n-1] - 2y[n-2]$

Using recursive method.

```
% Input 1 channel 1 to System 2
    y2_1 = dt_2(x1_1);
    figure();
    subplot 211
    stem(1:length(x1_1),x1_1); title('input signal');
    subplot 212
    stem(1:length(y2_1),y2_1); title('output signal');
% Input 1 channel 2 to System 2
    y2_2 = dt_2(x1_2);
    figure();
    subplot 211
    stem(1:length(x1_2),x1_2); title('input signal');
    subplot 212
    stem(1:length(y2_2),y2_2); title('output signal');
y2_1_both = [y2_1(:),y2_2(:)]; %combine two channels.
% soundsc(x1,fs_x1); %x1 original audio
```

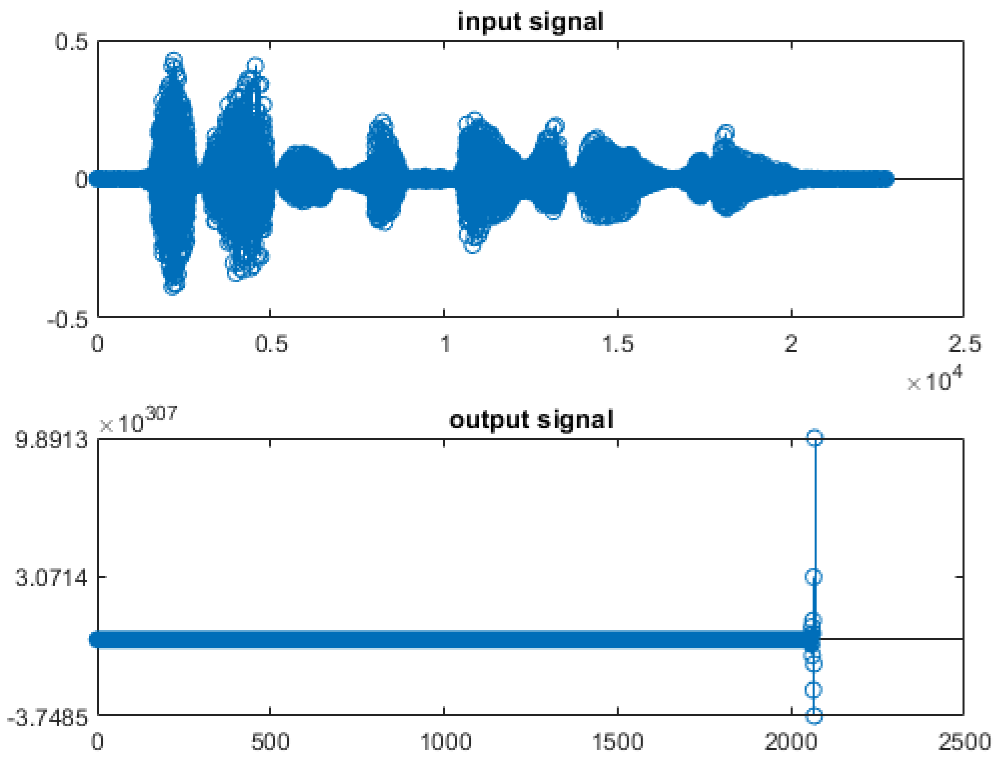
```
% soundsc(y2_1_both,fs_x1); %x1 output audio through system 2
```





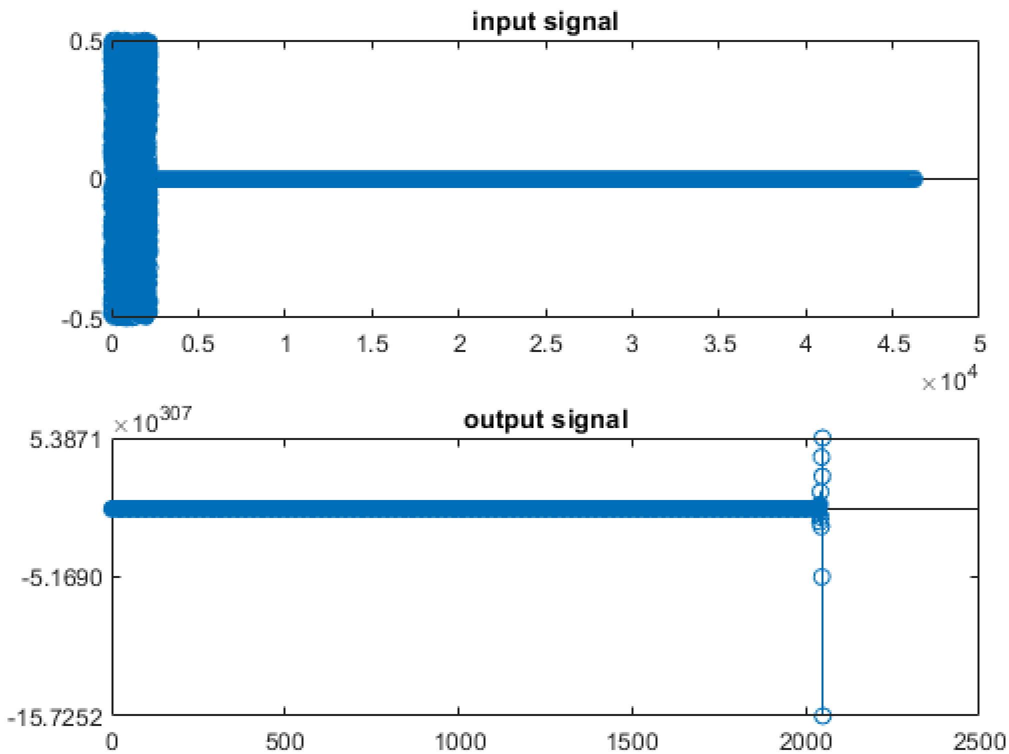
OBSERVATION: As per the system 2 generated graph and auditory comparison of the input and output signals of x1, there exist a great difference between the two as such the amplitude of the output signal was compressed from 0 to near end and was compressed in time-domain at the end.

```
% Input 2 to System 2
y2_3 = dt_2(x2);
figure();
subplot 211
stem(1:length(x2),x2); title('input signal');
subplot 212
stem(1:length(y2_3),y2_3); title('output signal');
% soundsc(x2,fs_x2); %x2 original audio
% soundsc(y2_3,fs_x2); %x2 output audio through system 2
```



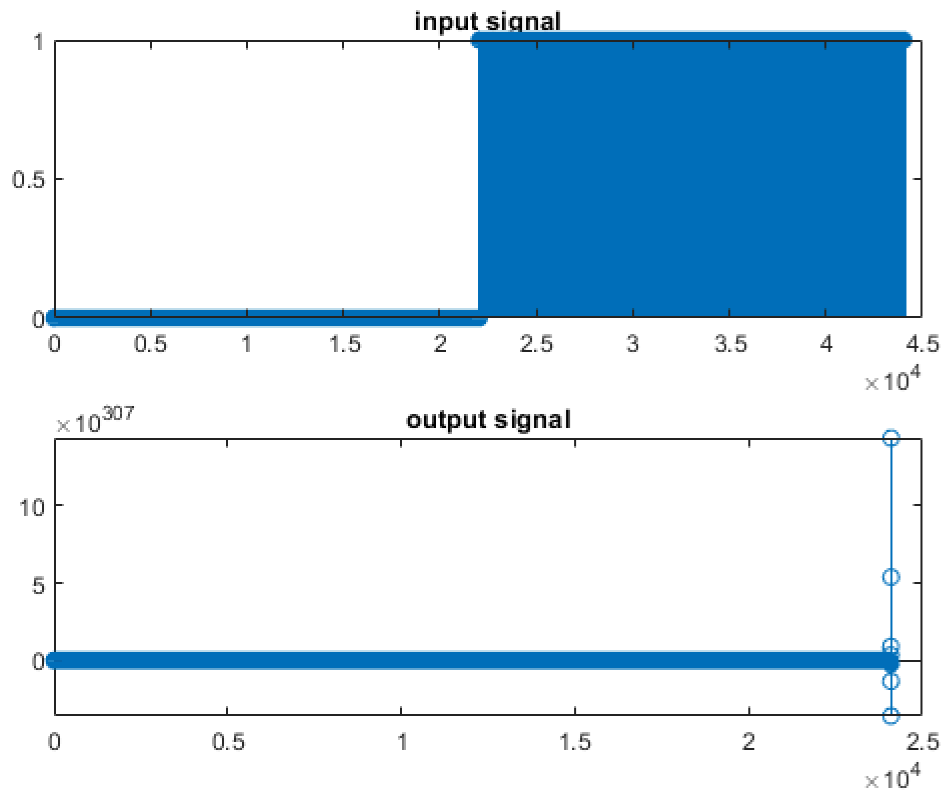
OBSERVATION: As per the system 2 generated graph and auditory comparison of the input and output signals of x2, there exist a great difference between the two as such the amplitude of the output signal was compressed from 0 to near end and was compressed in time-domain at the end.

```
% Input 3 to System 2
y2_4 = dt_2(x3);
figure();
subplot 211
stem(1:length(x3),x3); title('input signal');
subplot 212
stem(1:length(y2_4),y2_4); title('output signal');
% soundsc(x3,fs_x3); %x3 original audio
% soundsc(y2_4,fs_x3); %x3 output audio through system 2
```



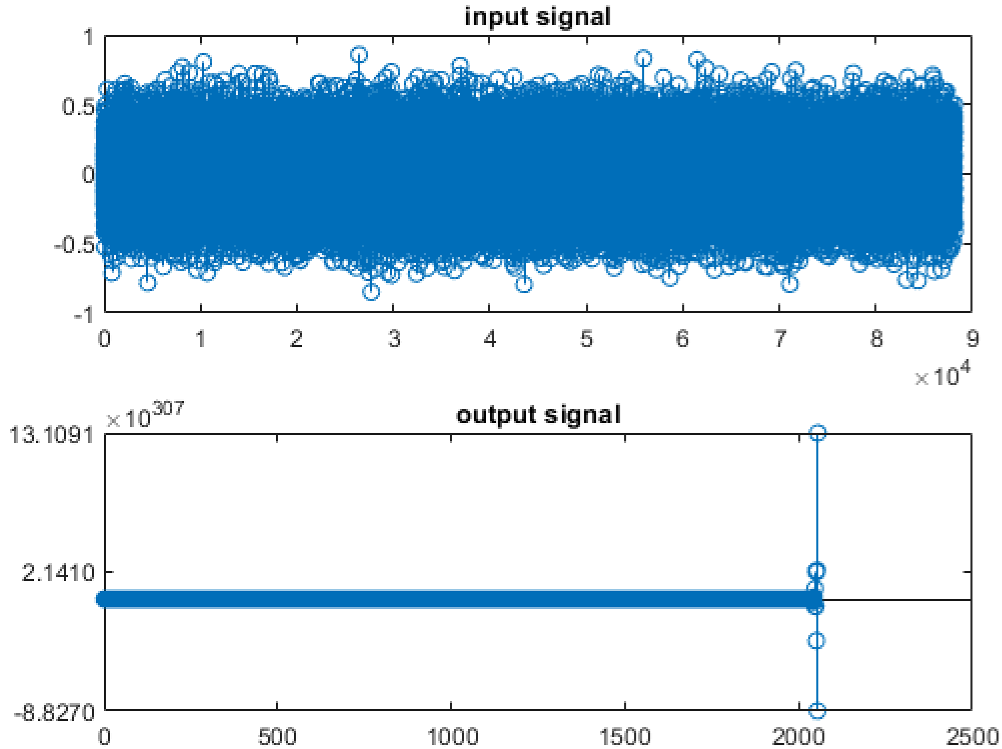
OBSERVATION: As per the system 2 generated graph and auditory comparison of the input and output signals of x3, there exist a great difference between the two as such the amplitude of the output signal was compressed from 0 to near end and was compressed in time-domain at the end.

```
% Input 4 to System 2
y2_5 = dt_2(x4);
figure();
subplot 211
stem(1:length(x4),x4); title('input signal');
subplot 212
stem(1:length(y2_5),y2_5); title('output signal');
% soundsc(x4,fs_x4); %x4 original audio
% soundsc(y2_5,fs_x4); %x4 output audio through system 2
```



OBSERVATION: As per the system 2 generated graph and auditory comparison of the input and output signals of x4, there exist a great difference between the two as such the amplitude of the output signal was compressed from 0 to near end and was compressed in time-domain at the end.

```
% Input 5 to System 2
y2_6 = dt_2(x5);
figure();
subplot 211
stem(1:length(x5),x5); title('input signal');
subplot 212
stem(1:length(y2_6),y2_6); title('output signal');
% soundsc(x5,fs_x5); %x5 original audio
% soundsc(y2_6,fs_x5); %x5 output audio through system 2
```



OBSERVATION: As per the system 2 generated graph and auditory comparison of the input and output signals of x5, there exist a great difference between the two as such the amplitude of the output signal was compressed from 0 to near end and was compressed in time-domain at the end.

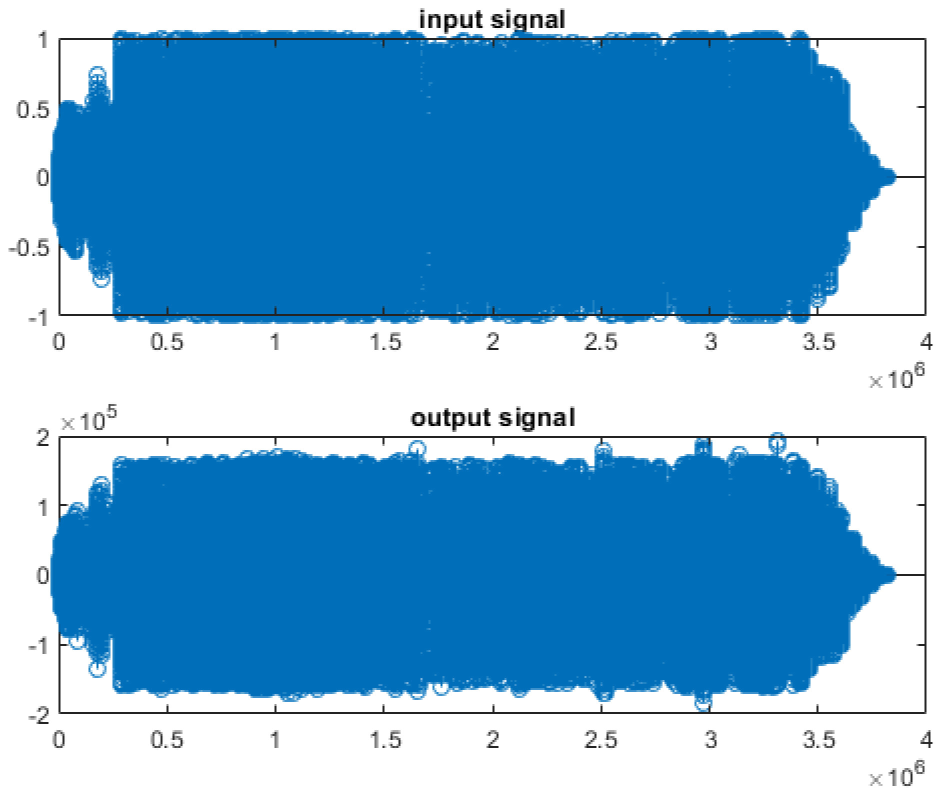
OVERALL, the input signals through system 2 were compressed in terms of space-domain resulting for output signals to expand in time-domain. The signals are no longer audible on human-cognitive level.

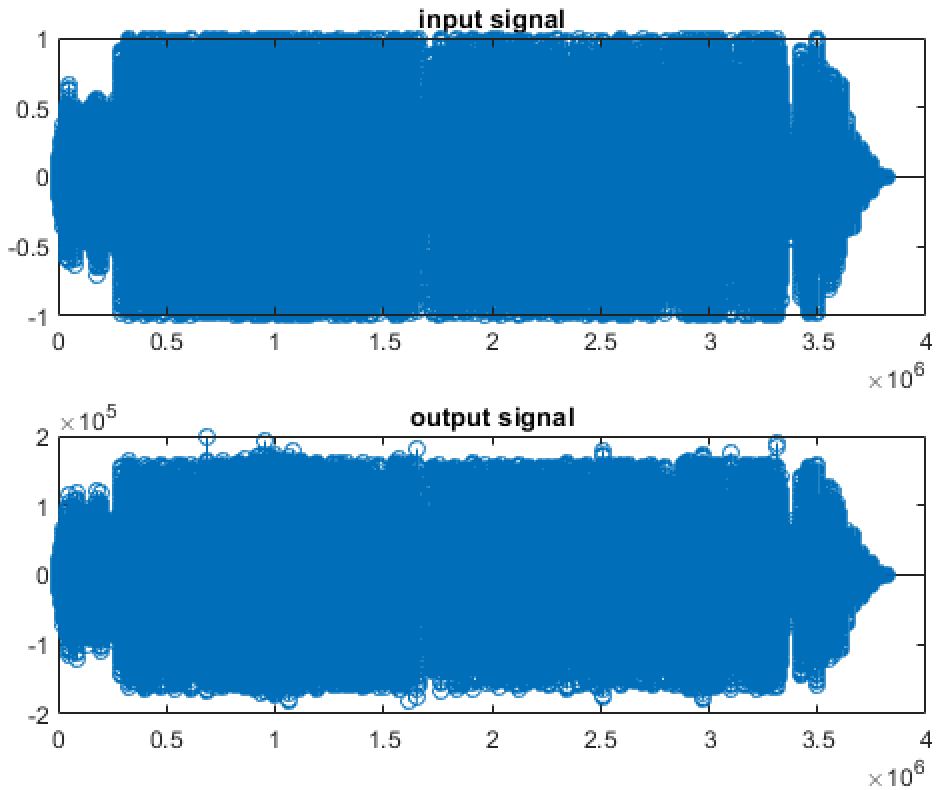
System 3 : $y[n] = 1.5x[n] + 0.5x[n-1] - 2y[n-1] - 2y[n-2]$

Using impulse response and convolution.

```
% Input 1 channel 1 to System 3
    y3_1 = dt_3(x1_1);
    figure();
    subplot 211
    stem(1:length(x1_1),x1_1); title('input signal');
    subplot 212
    stem(1:length(y3_1),y3_1); title('output signal');
% Input 1 channel 2 to System 3
    y3_2 = dt_3(x1_2);
    figure();
    subplot 211
    stem(1:length(x1_2),x1_2); title('input signal');
    subplot 212
    stem(1:length(y3_2),y3_2); title('output signal');
y3_1_both = [y3_1(:),y3_2(:)]; %combine two channels.
```

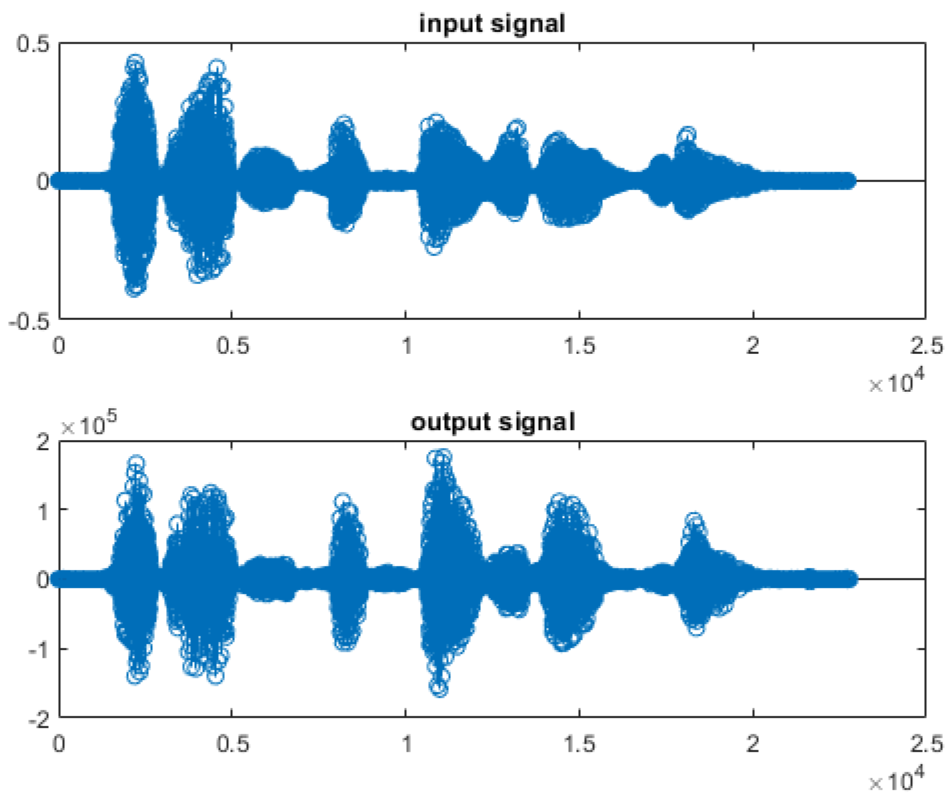
```
% soundsc(x1,fs_x1); %x1 original audio
% soundsc(y3_1_both,fs_x1); %x1 output audio through system 3
```


input signal
1
0.5
0
-0.5
-1
0
0.5
1
1.5
2
2.5
3
3.5
4
$\times 10^6$
output signal
$\times 10^5$
2
1
0
-1
-2
0
0.5
1
1.5
2
2.5
3
3.5
4
$\times 10^6$



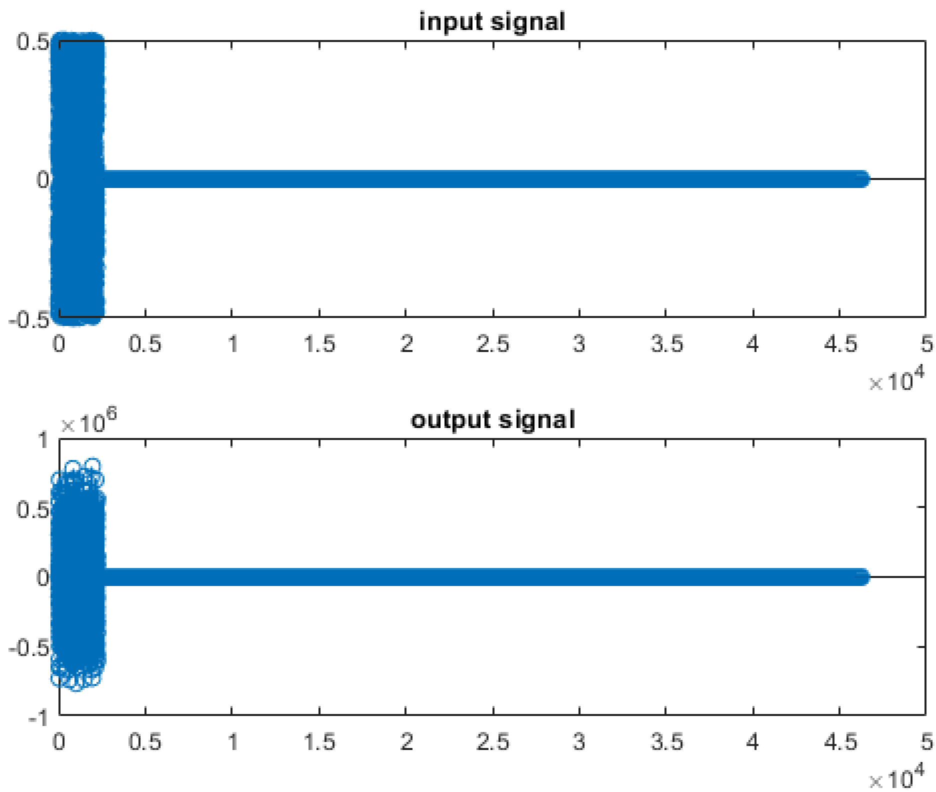
OBSERVATION: The output signal of x1 is smoother than its input after passing system 3.

```
% Input 2 to System 3
y3_3 = dt_3(x2);
figure();
subplot 211
stem(1:length(x2),x2); title('input signal');
subplot 212
stem(1:length(y3_3),y3_3); title('output signal');
% soundsc(x2,fs_x2); %x2 original audio
% soundsc(y3_3,fs_x2); %x2 output audio through system 3
```


input signal
0.5
0
-0.5
0
0.5
1
1.5
2
2.5
×10^4
output signal
×10^5
2
1
0
-1
-2
0
0.5
1
1.5
2
2.5
×10^4

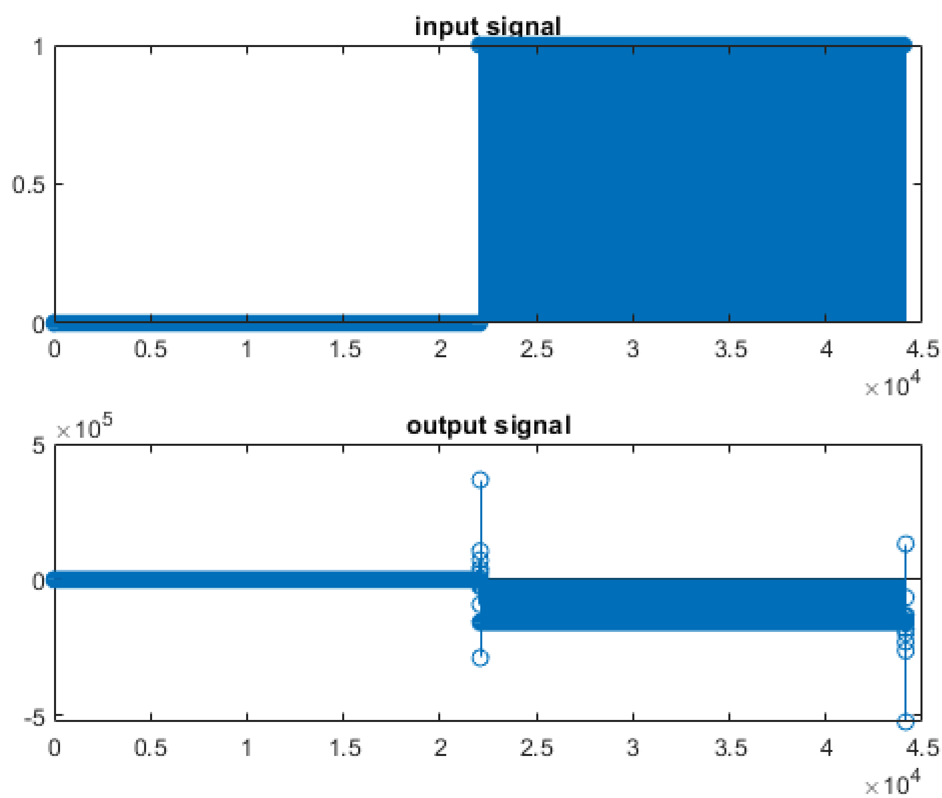
OBSERVATION: The output signal of x2 is smoother than its input after passing system 3.

```
% Input 3 to System 3
y3_4 = dt_3(x3);
figure();
subplot 211
stem(1:length(x3),x3); title('input signal');
subplot 212
stem(1:length(y3_4),y3_4); title('output signal');
% soundsc(x3,fs_x3); %x3 original audio
% soundsc(y3_4,fs_x3); %x3 output audio through system 3
```



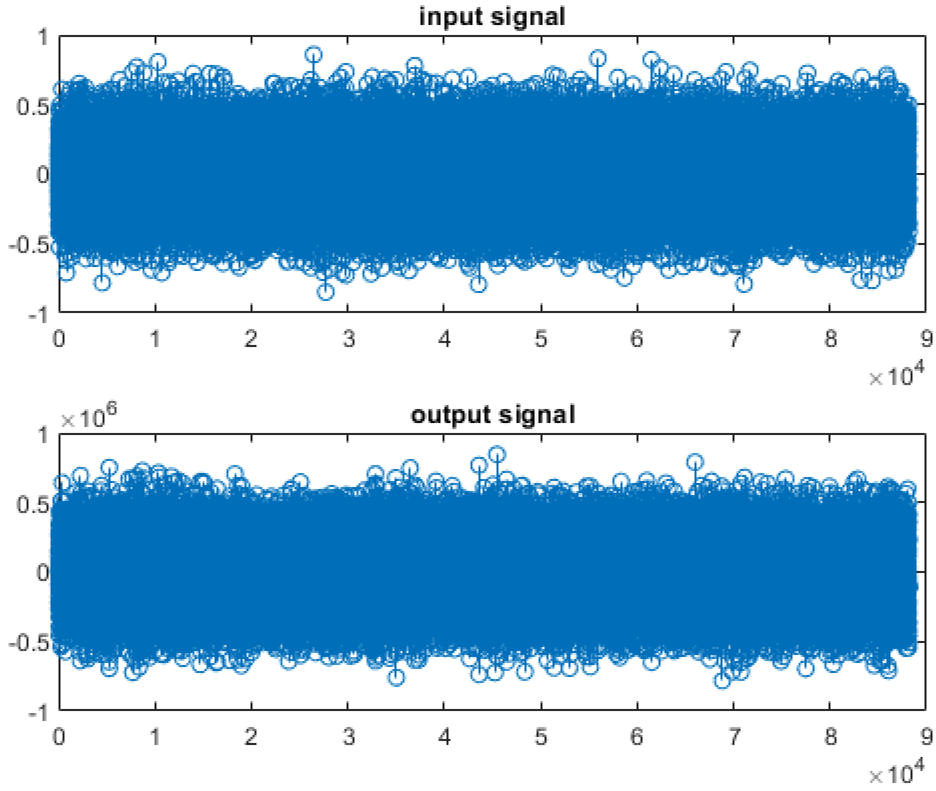
OBSERVATION: The output signal of x3 is smoother than its input after passing system 3.

```
% Input 4 to System 3
y3_5 = dt_3(x4);
figure();
subplot 211
stem(1:length(x4),x4); title('input signal');
subplot 212
stem(1:length(y3_5),y3_5); title('output signal');
% soundsc(x4,fs_x4); %x4 original audio
% soundsc(y3_5,fs_x4); %x4 output audio through system 3
```


input signal
1
0.5
0
0
0.5
1
1.5
2
2.5
3
3.5
4
4.5
$\times 10^4$
output signal
$\times 10^5$
5
0
-5
0
0.5
1
1.5
2
2.5
3
3.5
4
4.5
$\times 10^4$

OBSERVATION: The output signal of x4 is smoother than its input after passing system 3.

```
% Input 5 to System 3
y3_6 = dt_3(x5);
figure();
subplot 211
stem(1:length(x5),x5); title('input signal');
subplot 212
stem(1:length(y3_6),y3_6); title('output signal');
% soundsc(x5,fs_x5); %x5 original audio
% soundsc(y3_6,fs_x5); %x5 output audio through system 3
```



OBSERVATION: The output signal of x5 is smoother than its input after passing system 3.

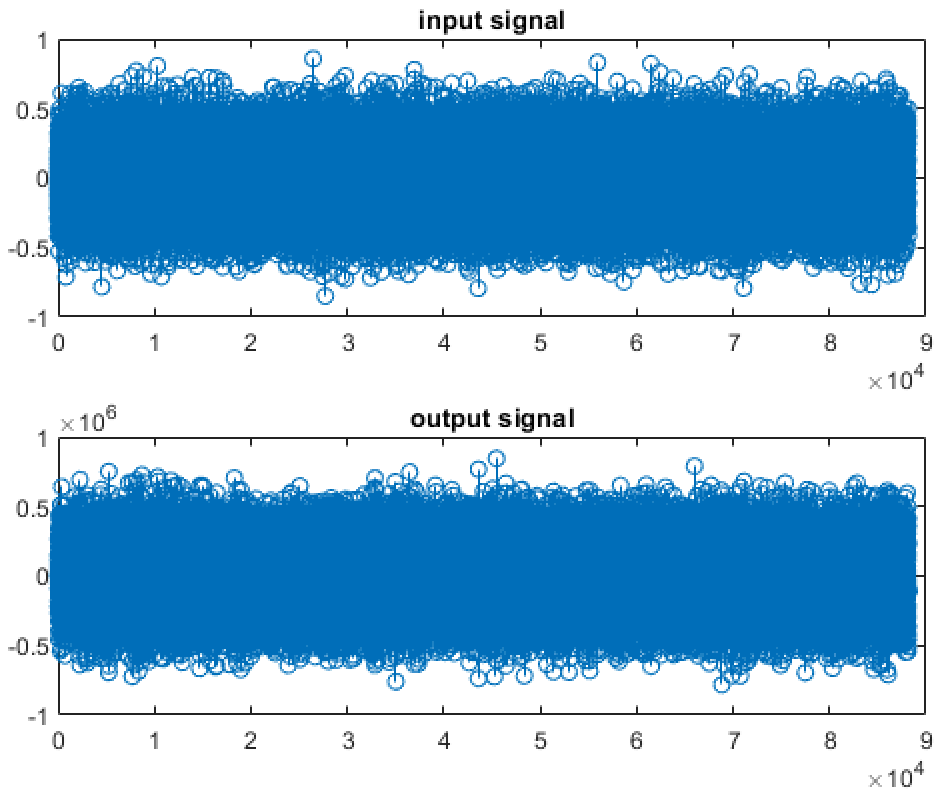
OVERALL, output signals after passing system 3 become smoother in audio as compared with its input counterpart. The output signals amplitude, as observed through charts are smaller by a small percentage._

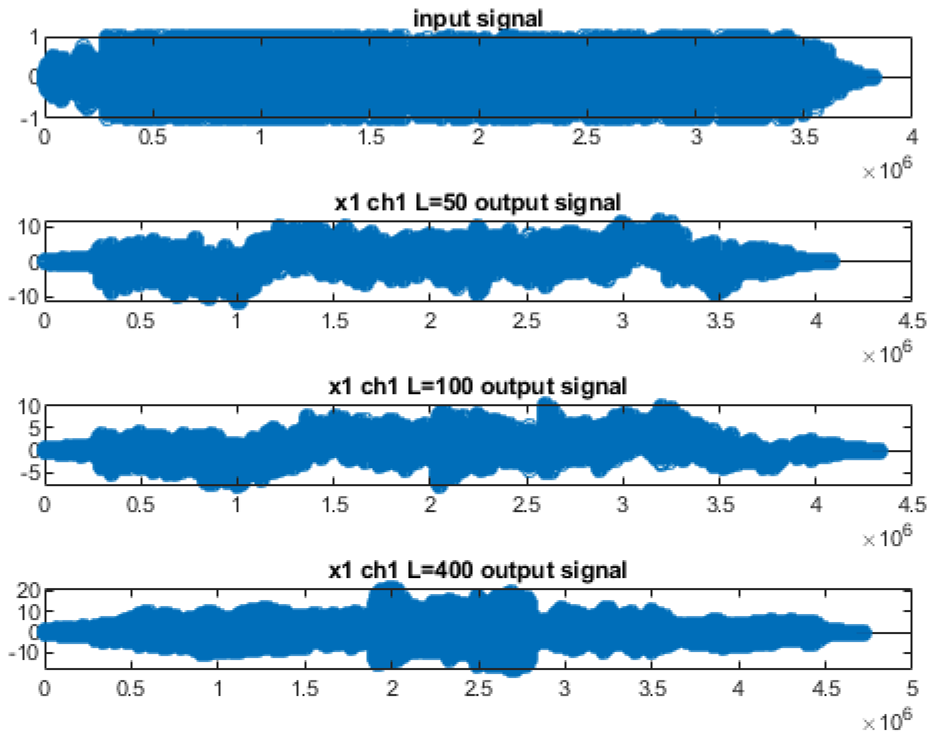
System 4 : $y[n] = x[n] + 0.5y[n-L] + 0.5y[n-L-1]$

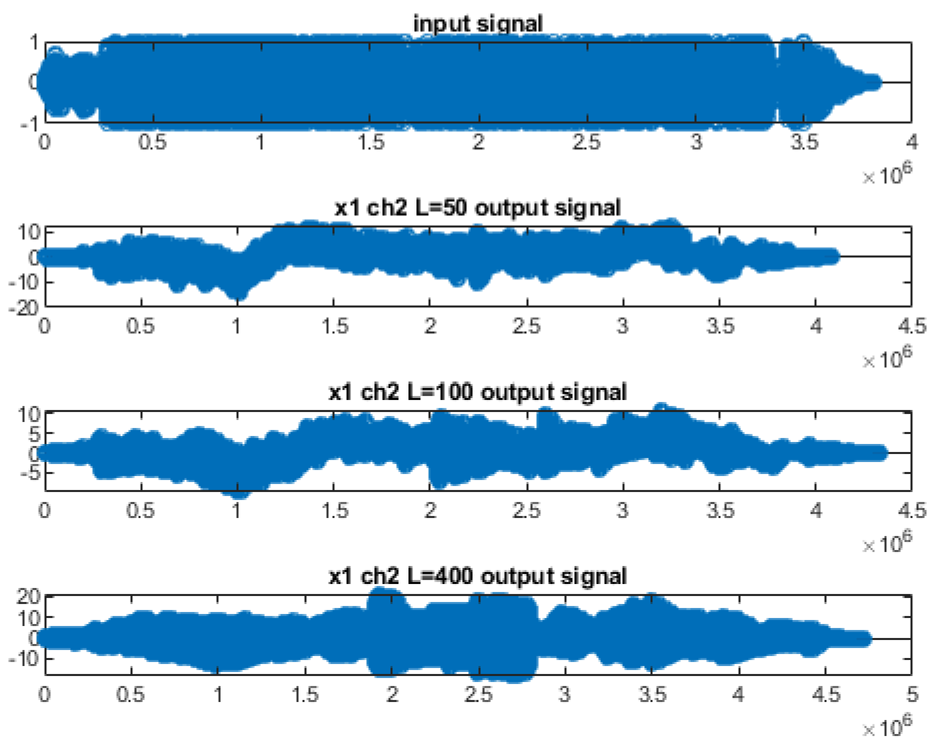
Using impulse response and convolution.

```
% Input 1 channel 1 to System 4
    y4_1_50 = dt_4(x1_1,50); % x1 L=50 ch1 system4
    y4_1_100 = dt_4(x1_1,100); % x1 L=100 ch1 system 4
    y4_1_400 = dt_4(x1_1,400); % x1 L=400 ch1 system 4
    figure();
    subplot 411
    stem(1:length(x1_1),x1_1); title('input signal');
    subplot 412
    stem(1:length(y4_1_50),y4_1_50); title('x1 ch1 L=50 output
 signal');
    subplot 413
    stem(1:length(y4_1_100),y4_1_100); title('x1 ch1 L=100 output
 signal');
    subplot 414
    stem(1:length(y4_1_400),y4_1_400); title('x1 ch1 L=400 output
 signal');
% Input 1 channel 2 to System 4
```

```
    y4_2_50 = dt_4(x1_2,50); % x1 L=50 ch2 system4
    y4_2_100 = dt_4(x1_2,100); % x1 L=100 ch2 system4
    y4_2_400 = dt_4(x1_2,400); % x1 L=400 ch2 system4
    figure();
    subplot 411
    stem(1:length(x1_2),x1_2); title('input signal');
    subplot 412
    stem(1:length(y4_2_50),y4_2_50); title('x1 ch2 L=50 output
 signal');
    subplot 413
    stem(1:length(y4_2_100),y4_2_100); title('x1 ch2 L=100 output
 signal');
    subplot 414
    stem(1:length(y4_2_400),y4_2_400); title('x1 ch2 L=400 output
 signal');
y4_1_both_50 = [y4_1_50(:),y4_2_50(:)]; %combine two channels L=50
y4_1_both_100 = [y4_1_100(:),y4_2_100(:)]; %combine two channels L=100
y4_1_both_400 = [y4_1_400(:),y4_2_400(:)]; %combine two channels L=400
% soundsc(x1,fs_x1); %x1 original audio
% soundsc(y4_1_both_50,fs_x1); %x1 output audio through system 4 L=50
% soundsc(y4_1_both_100,fs_x1); %x1 output audio through system 4
 L=100
% soundsc(y4_1_both_400,fs_x1); %x1 output audio through system 4
 L=400
```

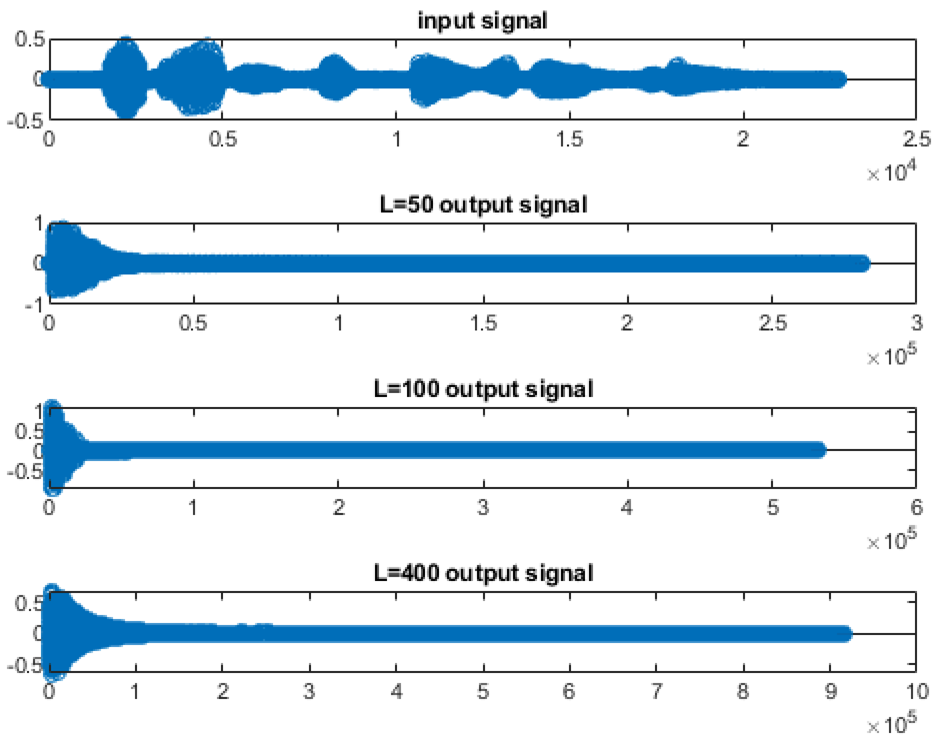



input signal
x1 ch1 L=50 output signal
x1 ch1 L=100 output signal
x1 ch1 L=400 output signal


input signal
x1 ch2 L=50 output signal
x1 ch2 L=100 output signal
x1 ch2 L=400 output signal

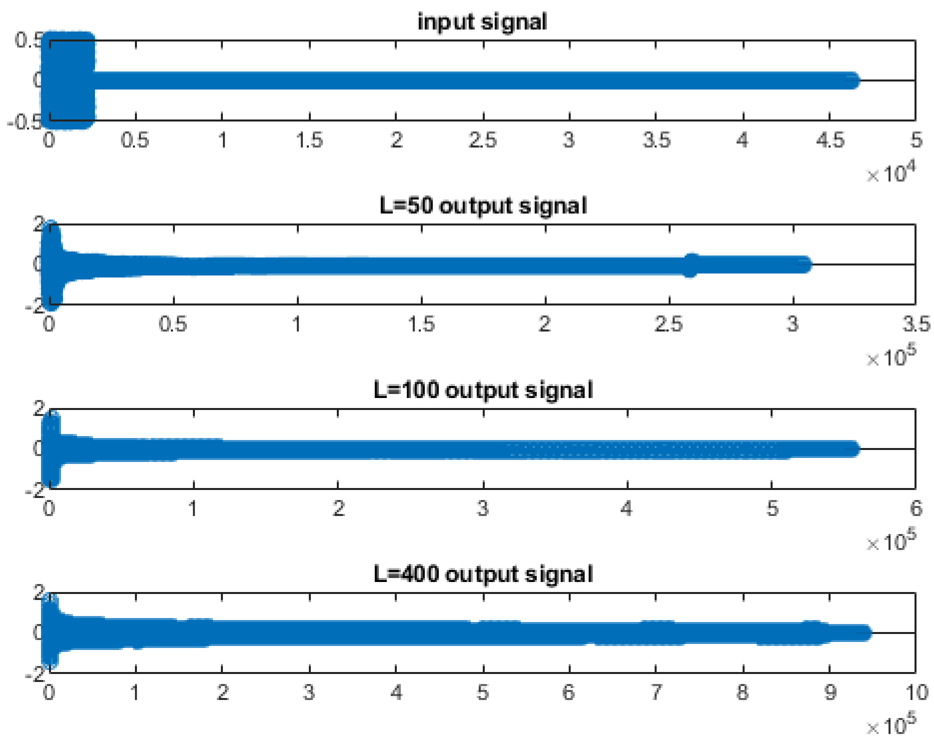
OBSERVATION: Output signals after passing system 4 appear to be downsampled however, the output signal with L=400 has a clearer audio than those x1 with L=100 and L=50. Visually, input and output signals differ greatly with L=50 and L=100 to have a similar plot that are farther with the output signal of L=400.

```
% Input 2 to System 4
y4_3_50 = dt_4(x2,50); % x2 L=50 system4
y4_3_100 = dt_4(x2,100); % x2 L=100 system4
y4_3_400 = dt_4(x2,400); % x2 L=400 system4
figure();
subplot 411
stem(1:length(x2),x2); title('input signal');
subplot 412
stem(1:length(y4_3_50),y4_3_50); title('L=50 output signal');
subplot 413
stem(1:length(y4_3_100),y4_3_100); title('L=100 output signal');
subplot 414
stem(1:length(y4_3_400),y4_3_400); title('L=400 output signal');
% soundsc(x2,fs_x2); %x2 original audio
% soundsc(y4_3_50,fs_x2); %x2 output audio through system 4 L=50
% soundsc(y4_3_100,fs_x2); %x2 output audio through system 4 L=100
% soundsc(y4_3_400,fs_x2); %x2 output audio through system 4 L=400
```



OBSERVATION: Output signals after passing system 4 appear to be downsampled however, the output signal with L=400 has a clearer audio than those x1 with L=100 and L=50. Visually, input and output signals differ greatly with L=50 and L=400 to have a similar plot that are shorter than the output signal of L=100.

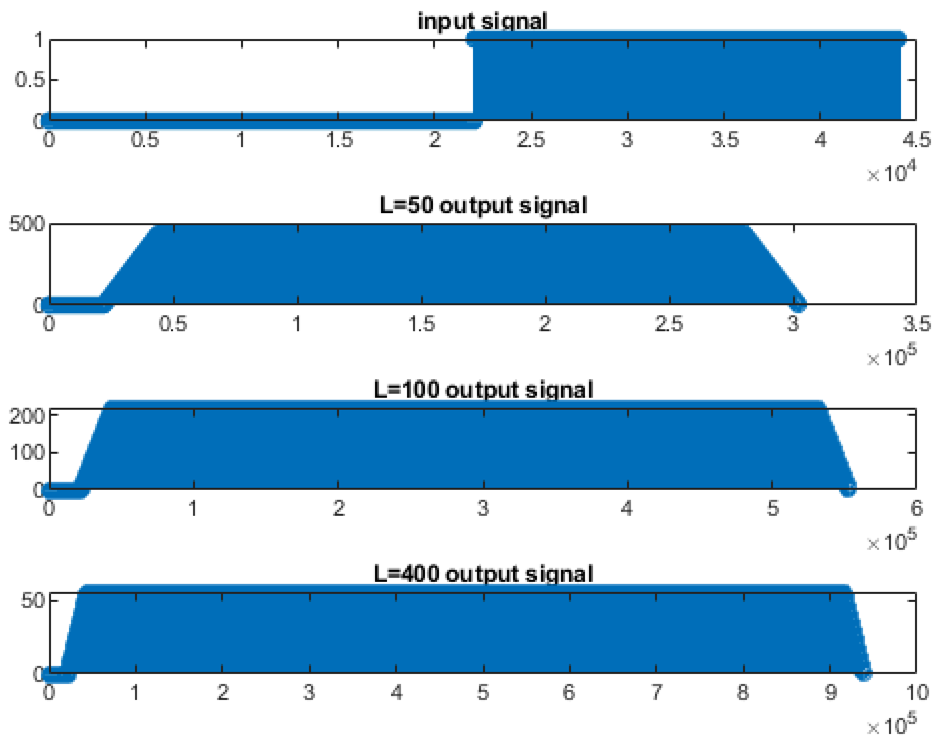
```
% Input 3 to System 4
y4_4_50 = dt_4(x3,50); % x3 L=50 system4
y4_4_100 = dt_4(x3,100); % x3 L=100 system4
y4_4_400 = dt_4(x3,400); % x3 L=400 system4
figure();
subplot 411
stem(1:length(x3),x3); title('input signal');
subplot 412
stem(1:length(y4_4_50),y4_4_50); title('L=50 output signal');
subplot 413
stem(1:length(y4_4_100),y4_4_100); title('L=100 output signal');
subplot 414
stem(1:length(y4_4_400),y4_4_400); title('L=400 output signal');
% soundsc(x3,fs_x3); %x3 original audio
% soundsc(y4_4_50,fs_x3); %x3 output audio through system 4 L=50
% soundsc(y4_4_100,fs_x3); %x3 output audio through system 4 L=100
% soundsc(y4_4_400,fs_x3); %x3 output audio through system 4 L=400
```



OBSERVATION: Output signals after passing system 4 appear to be downsampled. Each output signals being extended in term of time-domain as the value of L increases.

```
% Input 4 to System 3
y4_5_50 = dt_4(x4,50); % x4 L=50 system4
y4_5_100 = dt_4(x4,100); % x4 L=100 system4
y4_5_400 = dt_4(x4,400); % x4 L=400 system4
figure();
subplot 411
```

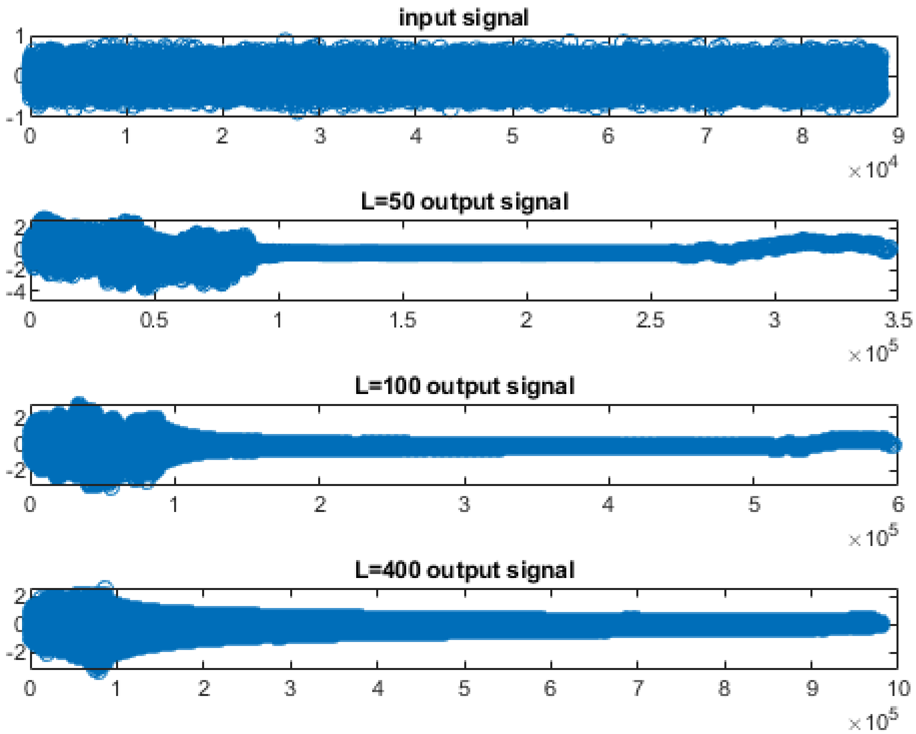
```
stem(1:length(x4),x4); title('input signal');
subplot 412
stem(1:length(y4_5_50),y4_5_50); title('L=50 output signal');
subplot 413
stem(1:length(y4_5_100),y4_5_100); title('L=100 output signal');
subplot 414
stem(1:length(y4_5_400),y4_5_400); title('L=400 output signal');
% soundsc(x4,fs_x4); %x4 original audio
% soundsc(y4_5_50,fs_x4); %x4 output audio through system 4 L=50
% soundsc(y4_5_100,fs_x4); %x4 output audio through system 4 L=100
% soundsc(y4_5_400,fs_x4); %x4 output audio through system 4 L=400
```



OBSERVATION: Output signal mirror the other half of the original signal with each end begin and end on a ramp.

```
% Input 5 to System 4
y4_6 = dt_4(x5,100);
y4_6_50 = dt_4(x5,50); % x5 L=50 system4
y4_6_100 = dt_4(x5,100); % x5 L=100 system4
y4_6_400 = dt_4(x5,400); % x5 L=400 system4
figure();
subplot 411
stem(1:length(x5),x5); title('input signal');
subplot 412
stem(1:length(y4_6_50),y4_6_50); title('L=50 output signal');
subplot 413
stem(1:length(y4_6_100),y4_6_100); title('L=100 output signal');
```

```
subplot 414
stem(1:length(y4_6_400),y4_6_400); title('L=400 output signal');
% soundsc(x5,fs_x5); %x5 original audio
% soundsc(y4_6_50,fs_x5); %x5 output audio through system 4 L=50
% soundsc(y4_6_100,fs_x5); %x5 output audio through system 4 L=100
% soundsc(y4_6_400,fs_x5); %x5 output audio through system 4 L=400
```



OBSERVATION: Output signals appear to being shrank at 1x10^n with L=50 in a gradual manner and L=100 and 400 on a fading manner.

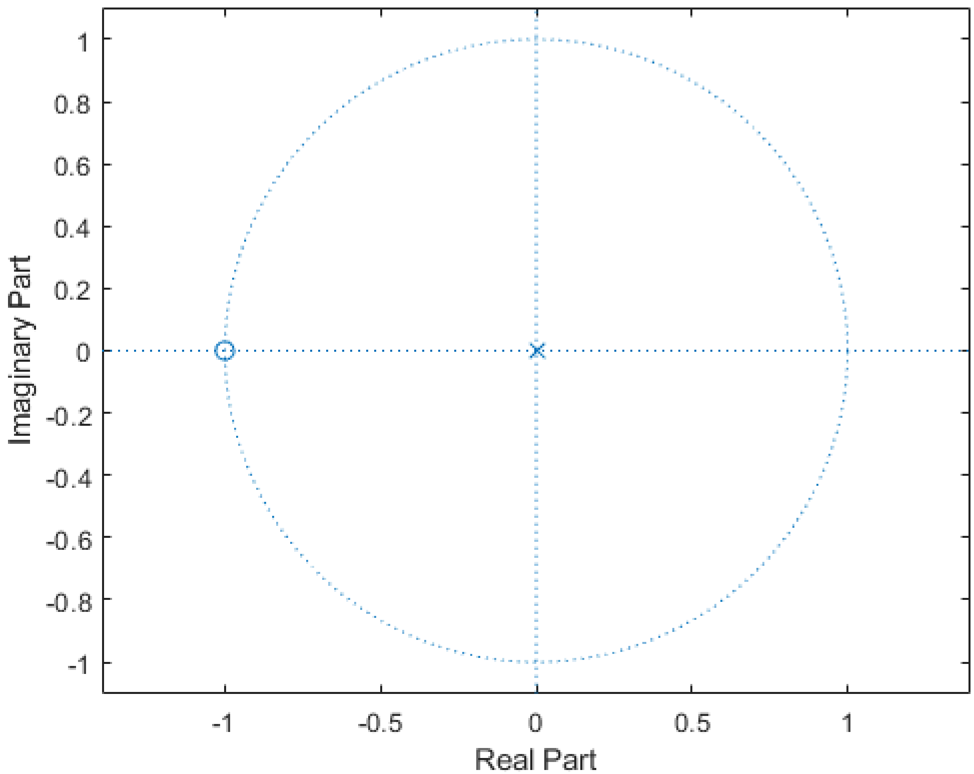
OVERALL, output signals after system 4 appeared to be downsampled and played on a lower tone as compared to the original. Among the three L-values, output signals that are generated with L=400 are more audible compared to the other signals generated from L=50 and L=100. Furthermore, certain portion og the output siganls are being squeezed in a fading manner depending on the value of L.

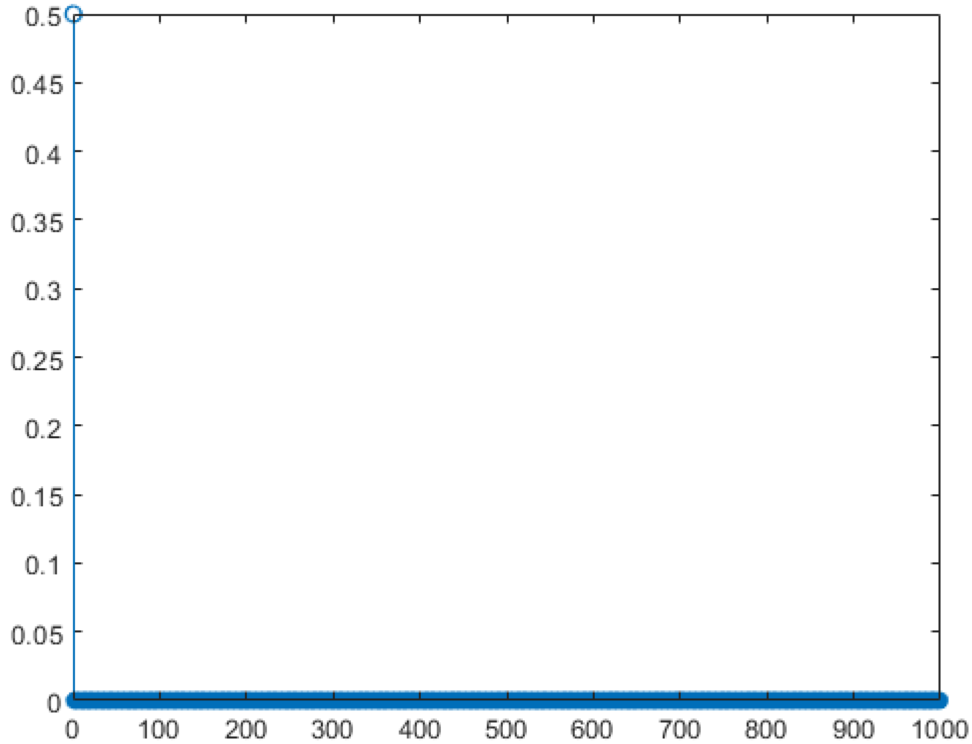
Is the system BIBO stable?

System 1 is a BIBO stable system as poles are inside the boundary of the system as shown in the zplane. Also, impulse response graph of system 1 shows that it is finite thus, the values are bounded and does not change in value.

```
as1 = [1]; % system 1 output coef
bs1 = [0.5 0.5]; % system 1 input coef
figure(); zplane(bs1,as1); % generate z-pole of system 1;
s1N=1000;
s1n=0:s1N-1;
```

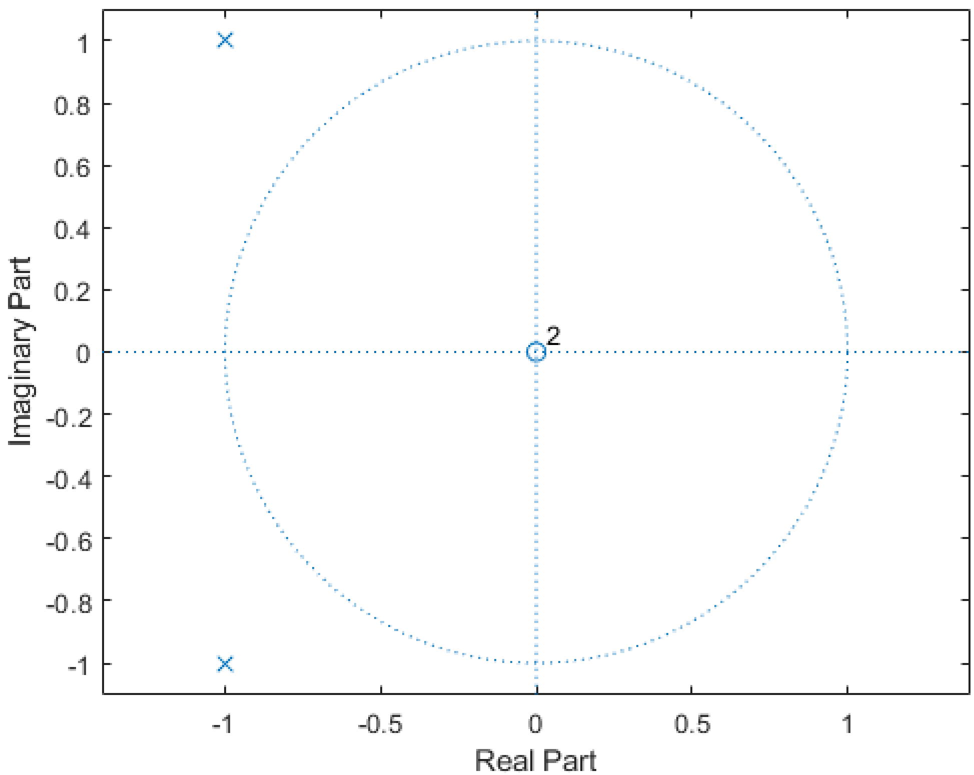
```
s1x = (s1n==0);
s1y=filter(bs1,as1,s1x);
figure(); stem(s1n,s1y);
```

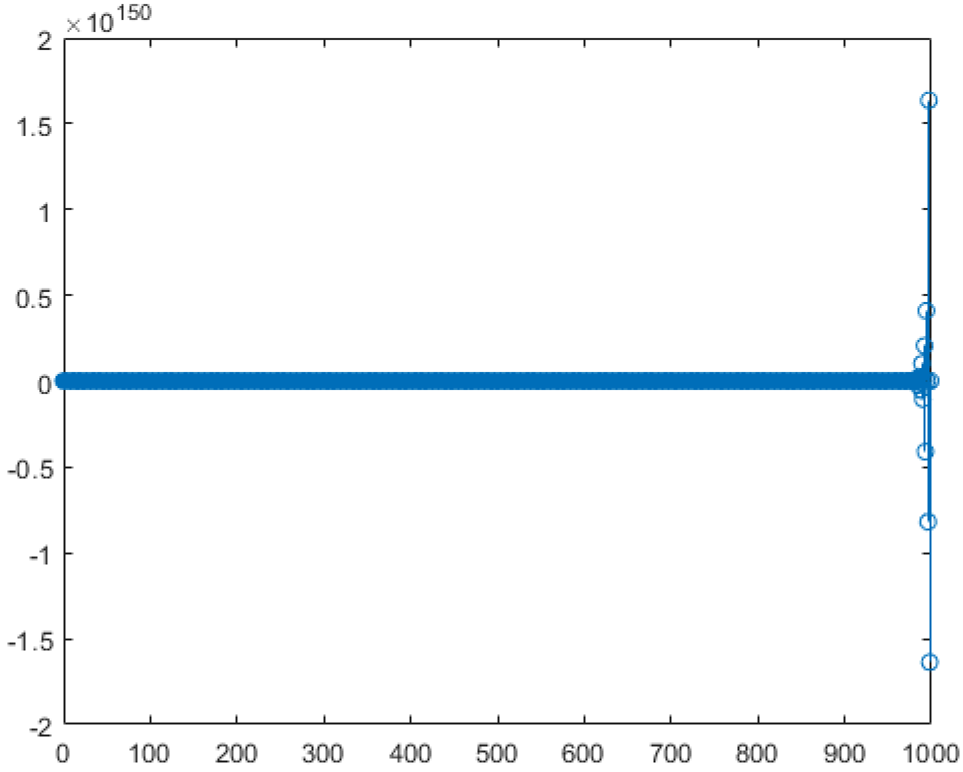




System 2 is NOT a BIBO stable system with its poles are outside of the system's circle as shown in the zplane. This is supported by the impulse response graph with its amplitude changes by near end of time.

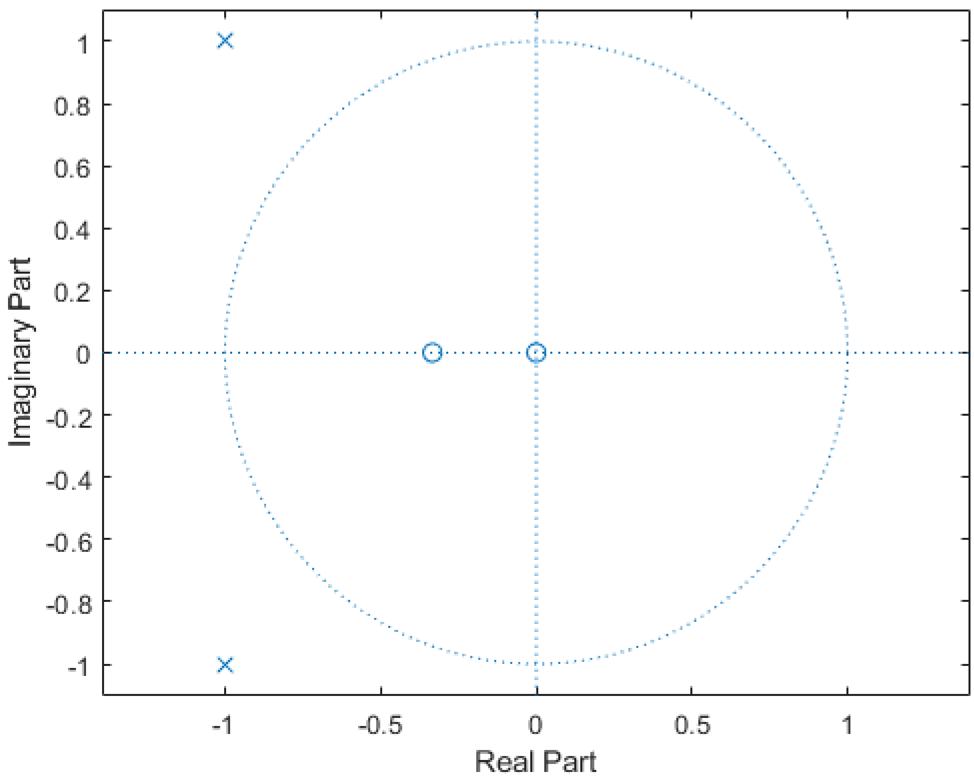
```
as2 = [1 2 2]; % system 2 output coef
bs2 = [1]; % system 2 input coef
figure(); zplane(bs2,as2); % generate z-pole of system 2;
s2N=1000;
s2n=0:s2N-1;
s2x = (s2n==0);
s2y=filter(bs2,as2,s2x);
figure(); stem(s2n,s2y);
```

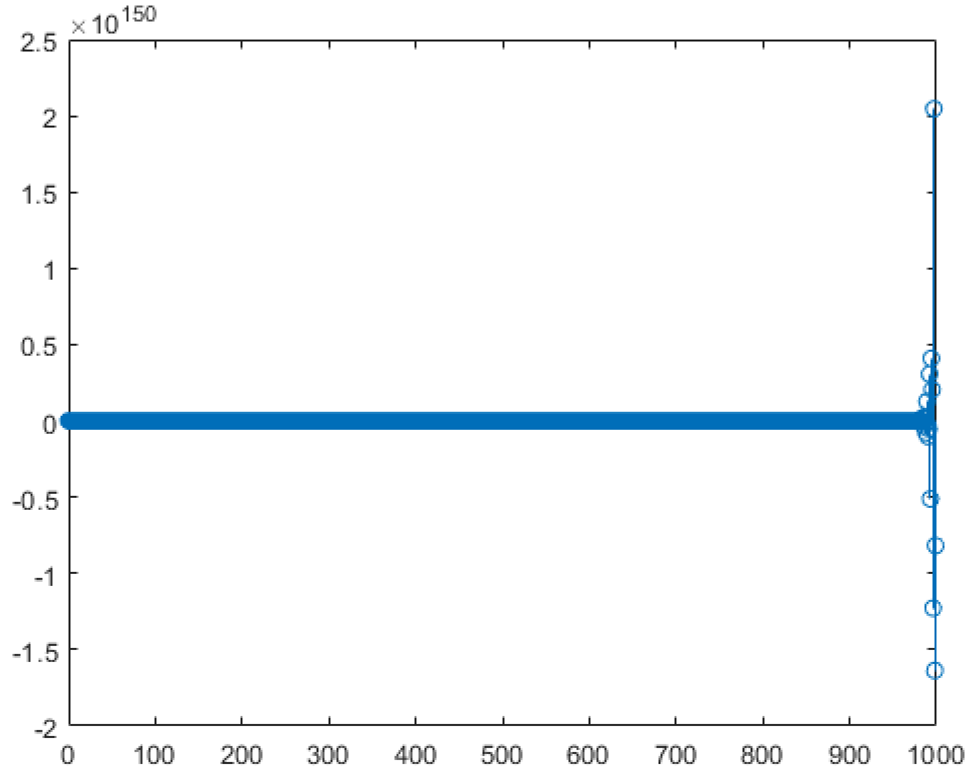

Imaginary Part
Real Part
2
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
-1
-0.5
0
0.5
1


×10^150
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
100
200
300
400
500
600
700
800
900
1000

System 3 is NOT a BIBO stable system as its poles are outside of the system's circle as shown in the zplane. Similar to system 2, its impulse reponse graph shows that its amplitude chanegs by near end of time.

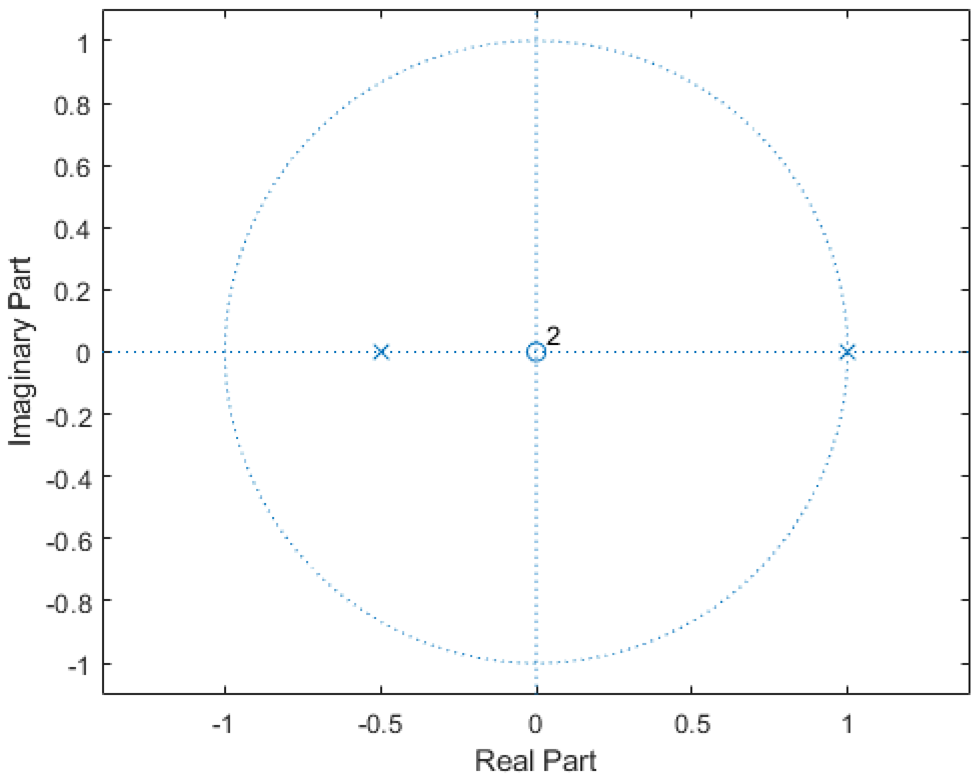
```
as3 = [1 2 2]; % system 3 output coef
bs3 = [1.5 0.5]; % system 3 input coef
figure(); zplane(bs3,as3); % generate z-pole of system 3;
s3N=1000;
s3n=0:s3N-1;
s3x = (s3n==0);
s3y=filter(bs3,as3,s3x);
figure(); stem(s3n,s3y);
```

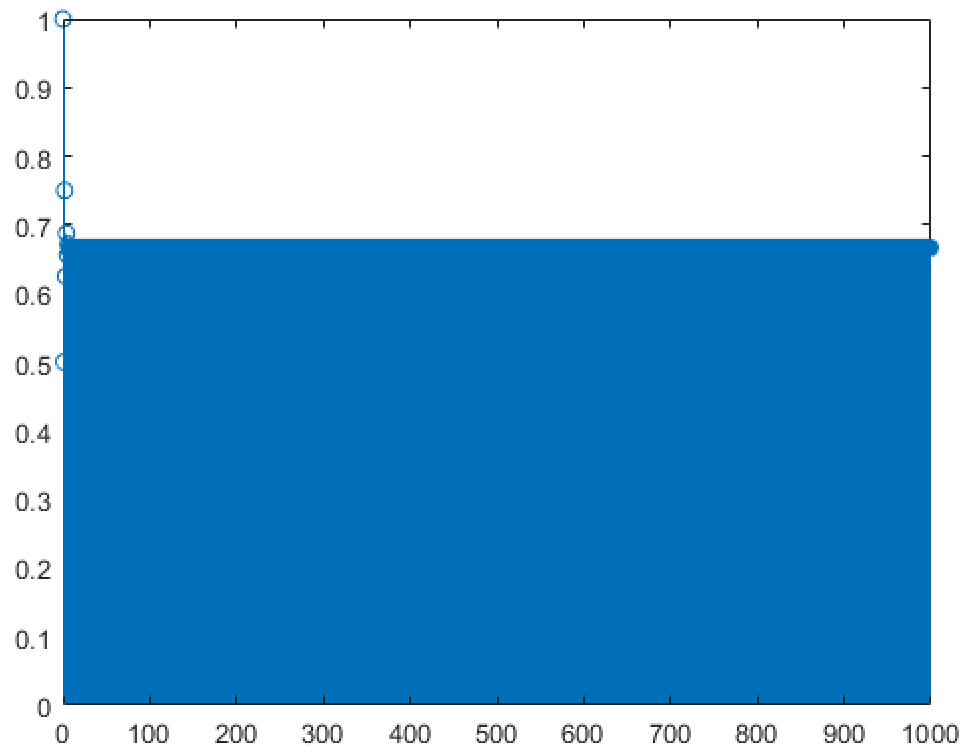




System 4 is a BIBO stable system as poles are present inside the circle of the system as shown in the zplane below. Additionally, its impulse response grah shows that its amplitude remains constant until the end.

```
as4 = [1 -0.5 -0.5]; % system 3 output coef
bs4 = [1]; % system 3 input coef
figure();
zplane(bs4,as4); % generate z-pole of system 4;
s4N=1000;
s4n=0:s4N-1;
s4x = (s4n==0);
s4y=filter(bs4,as4,s4x);
figure(); stem(s4n,s4y);
```


1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
Imaginary Part
2
-1
-0.5
0
0.5
1
Real Part


1
0.9
0.8
0.7
0.6
0.5
0.4
0.3
0.2
0.1
0
0
100
200
300
400
500
600
700
800
900
1000

Is the system causal?

System 1 is a causal system as its ouputs depend on the current and previous inputs. For reference, consider the provided function which was implemented in matlab below. The values of y(n) depends on the terminal values of 1st term and 2nd term. Likewise, the 2nd term depends on the previous output of the 1st term in the inputs side.

```
% function y = dt_1(x)
%     % x must be defined in the main function
%     y = zeros(1,length(x)); %initialize output signal y
%     for n = 1:length(x) %5 time indices
%         if n<2
%             y(n) = x(n);
%         else
%             y(n) = 0.5*x(n) + 0.5*x(n-1);
%         end
%     end
% end
```

System 2 is a causal system as its output depend only on the past, and present values of input and previous outputs. For reference, consider the function which was implemented in matlab below. function y = dt_2(x)

```
%     % x must be defined in the main function
%     y = zeros(1,length(x)); %initialize output signal y
%     for n = 1:length(x) %5 time indices
%          if n==1
%             y(n) = x(n);
%          elseif n==2
%             y(n) = x(n) - 2*y(n-1);
%          else
%             y(n) = x(n) - 2*y(n-1)- 2*y(n-2);
%          end
%     end
% end
```

System 3 is a causal system as its output depend only on the past, and present inputs and past outputs. For reference, consider system 3 function with n = 1. present output = present inpt + past inp - previous outputs

```
% y[1] = 1.5*x[1] + 0.5*x[1-1] - 2*y[1-1] - 2*y[1-2]
% y[1] = 1.58x[1] + 0.5*x[0] - 2*y[0] - 2*y[-1]
```

System 4 is also a causal system with its input is dependent on current and past inputs and past outputs. Consider substituting L=100, n=1 to the function below.

```
% y[1] = x[1]+0.5*y[1-100]+0.5*y[1-100-1]
% y[1] = x[1]+0.5*y[-99]+0.5*y[-100]
```

Is the system FIR or IIR?

System 1: Since system 1 is causal and its pole is located at the origin, system 1 is an FIR. See the Figure output of line 418 in the attached file below for evidence. As a rule of thumb, if the system is causal and its poles are located at the origin, then it is FIR.

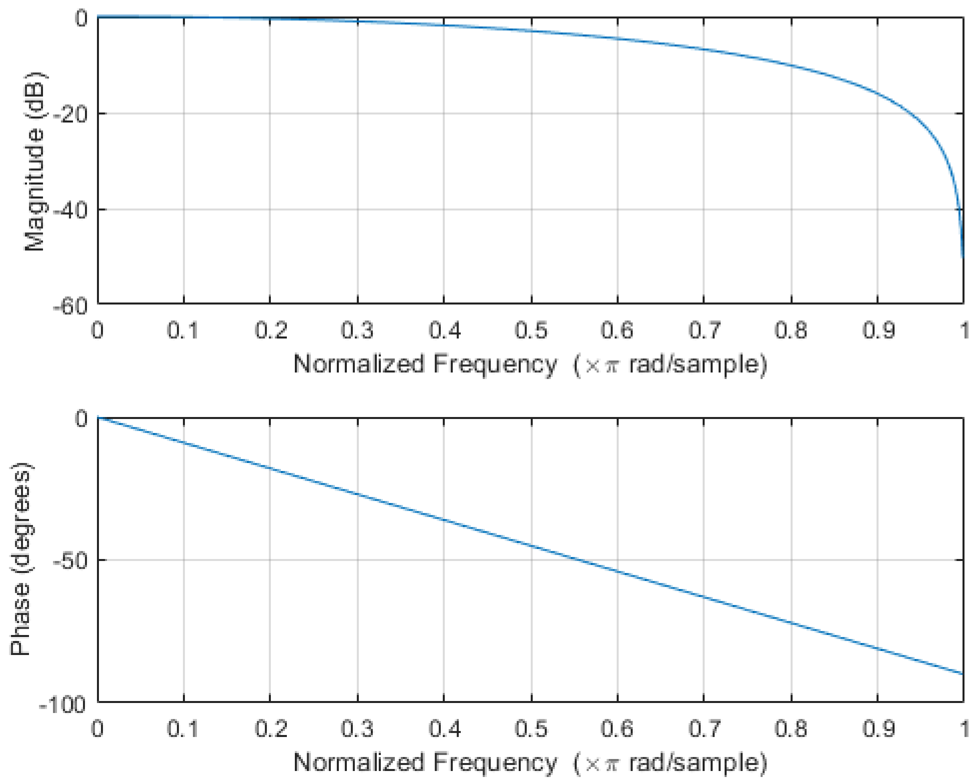
System 2: Since system 2 is causal and its poles are beyond the boundary of the system nor the origin, it is an IIR. See the figure output of line 430 in the ttached file below for evidence.

System 3: Since system 3 is causal and its poles are outside the boundary of the system nor the origin, it is an IIR. See the figure output of line 442 in the attached file below.

System 4: Since system 4 is causal and its poles are not in the origin but within the boundary of the system, it is an IIR. See the figure output of line 452 in the attached file below.

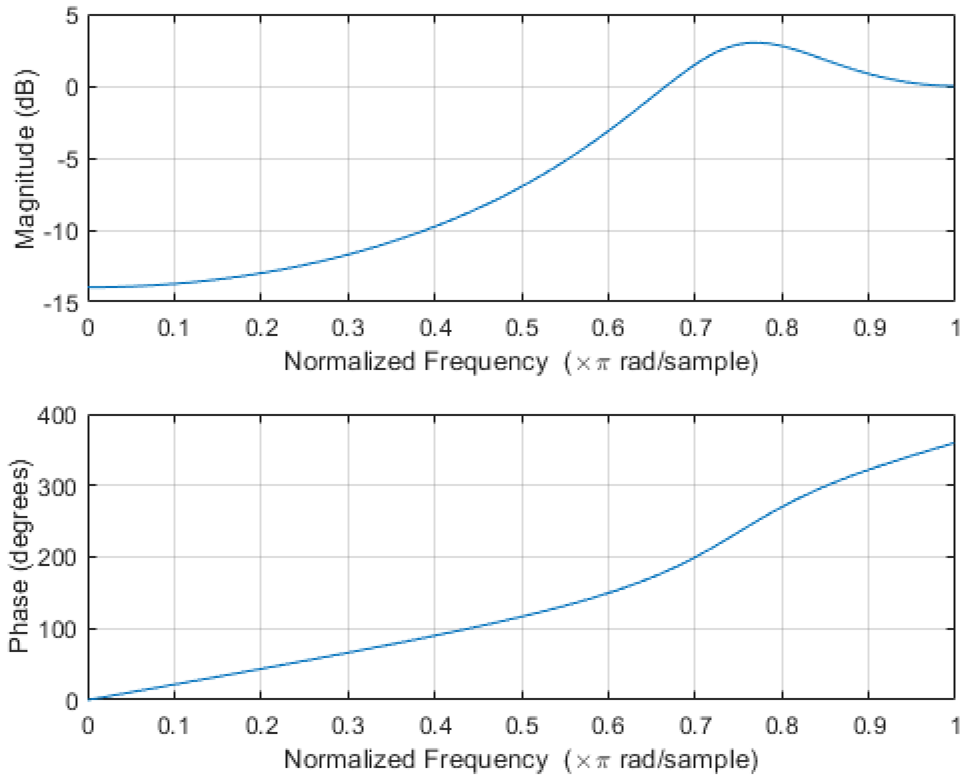
What does the system do?

```
% System 1
figure(); freqz(bs1,as1);
```



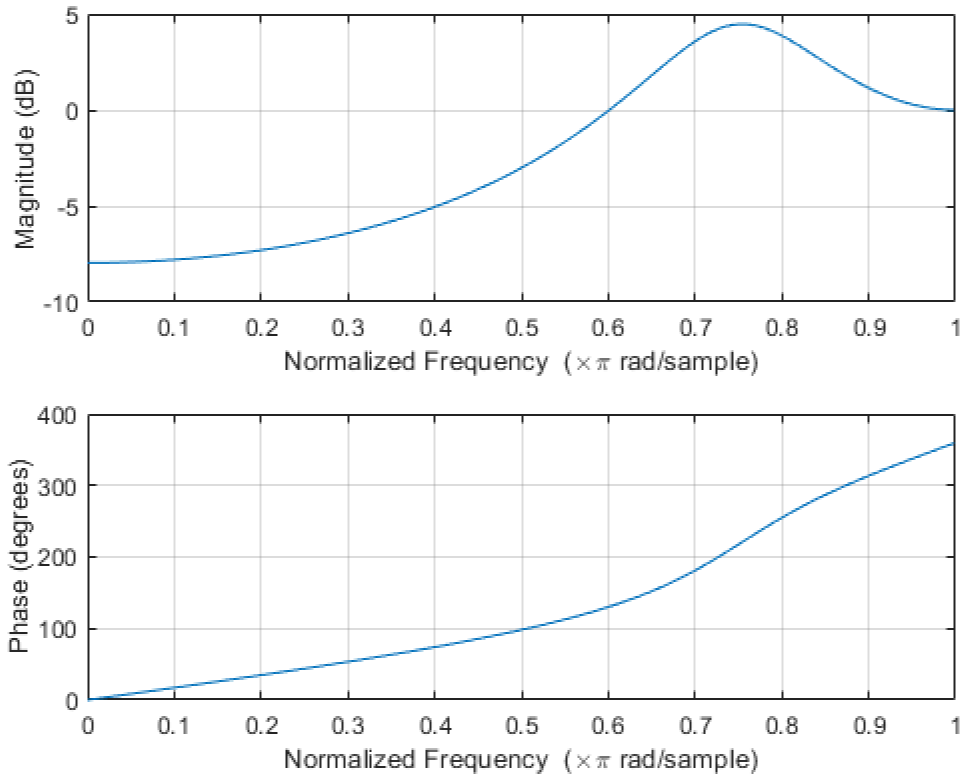
OBSERVATION: System 1 shifted the input signal from 0 to -50dB with a phase of 0 to -90 degrees per cycle. This results for the output signal to be audible and be sense to have been no difference with the original audio signal.

```
% System 2
figure(); freqz(bs2,as2);
```



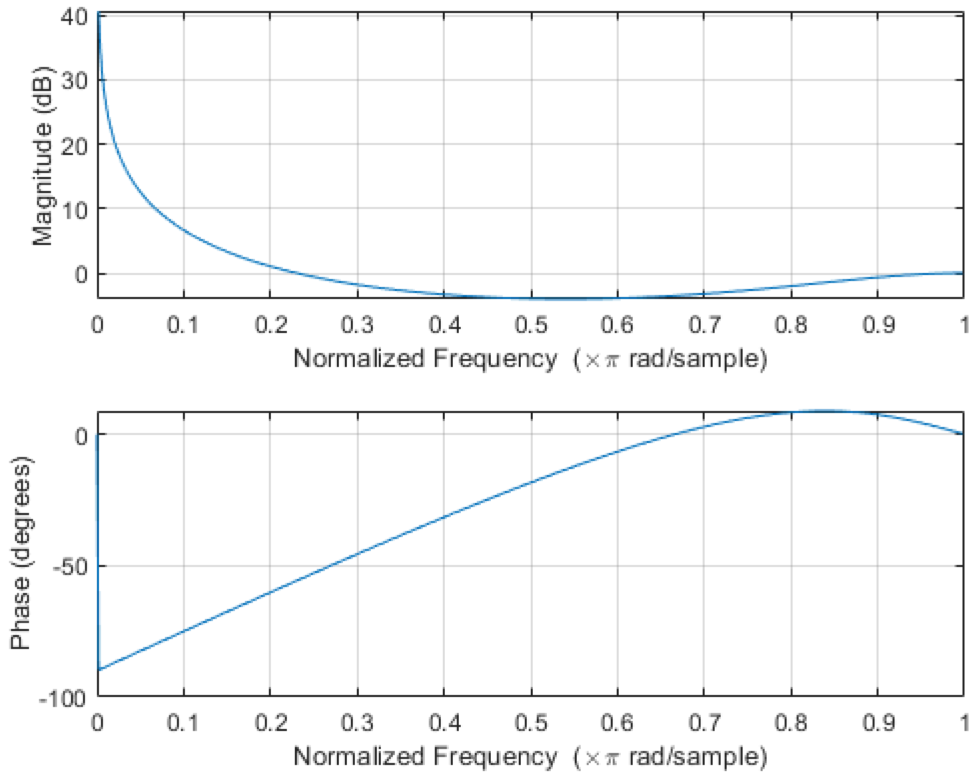
OBSERVATION: Sysem 2 shifted the magnitude of the input signal from -14 to 3dB with a phase of 0 to 350 degrees per cycle. This makes the output signal of the audio to be inaudible on human-level.

```
% System 3
figure(); freqz(bs3,as3);
```



OBSERVATION: System 3 shifted the magnitude of the input signal from -8dB %to 5dB with a phase of 0 to 360 degrees per cycle. This results for the %audio output signal to be audible just like system 1 and be sensed like %there is no difference with the original audio due to small difference.

```
% System 4
figure(); freqz(bs4,as4);
```



OBSERVATION: System 4 shifted the magnitude of the input signal logarithmically from 40 to -3dB with a phase of -90 to 0 degrees per cycle. This makes the output signal to be audible on human level but with a mixed of a white-noise-like-audio or downsampled signal.

Overall

System 1 is an FIR, a causal, and BIBO stable system that shifted the input audio from 0 to -50dB with a phase of 0 to -90 degrees per cycle and resulted to an output audio signal that is audible to human-level and having a little to no difference quality with the original audio signals.

System 2 is an IIR, causal, and BIBO unstable system that shifted the input audio signals from -14 to 3dB with a phase of 0-350 degrees per cycle and makes the resulting output signal inaudible on human-level.

System 3 is an IIR, causal, and BIBO unstable system that shifted the input audio signals from -8 to 5dB with a phase of 0-360 degrees per cycle and outcome an output signal that is audible to the human-level with quality as system 1 and the original audio signal.

System 4 on the other hand is an IIR, causal, and BIBO stable system that log-shifted input audio signal from 40 to -3dB with a phase angle of -90 to 0 degrees making the output signal to be audible on human-level but with white-noise-sound-included or of audible lower quality as the original audio signal.

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