Lab1 Report

Students :

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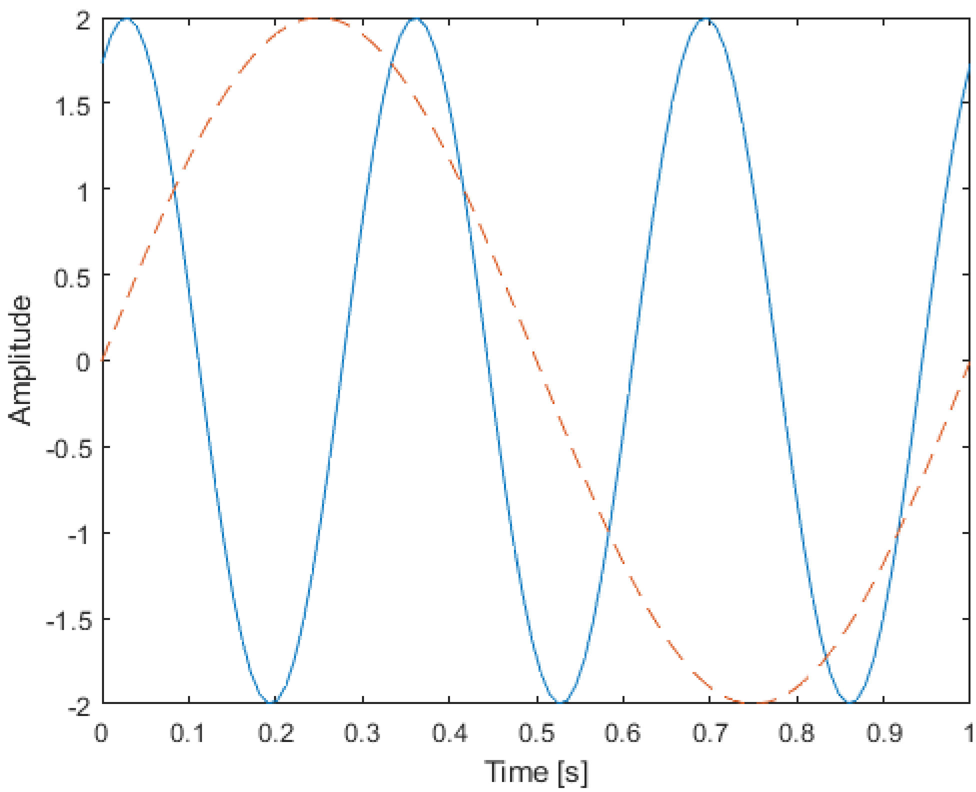
Task 1

Generate three periods of a continuous-time sinusoid with amplitude A = 2, frequency f = 3 Hz ($\omega = 2\pi f$), and phase $\varphi = \pi/3$. In addition, create a second sinusoid of equal length (in time). The second sinusoid should have amplitude A = 2, frequency f = 1 Hz, and phase $\varphi = 0$. Plot these in the same figure, one of the signals should be dashed while the other should be solid, see `help plot`, and `help hold`. Remember to choose the time vector t, so that the figures look good

```
task1 = figure('Name','Task 1');
figure(task1)
% Define a general sinus
sinusoid = @(A,Omega,phase,t) A*sin(Omega.*t+phase);
t = linspace(0,1,100);
sig_1 = sinusoid(2,3.*2.*pi,pi./3,t);
sig_2 = sinusoid(2,1.*2.*pi,0,t);

plot(t,sig_1);
hold on
fp = plot(t,sig_2,'--');
ylabel("Amplitude");    % Since there is no better lable
xlabel("Time [s]")
ylabel("Amplitude")
hold off
```



Task 2

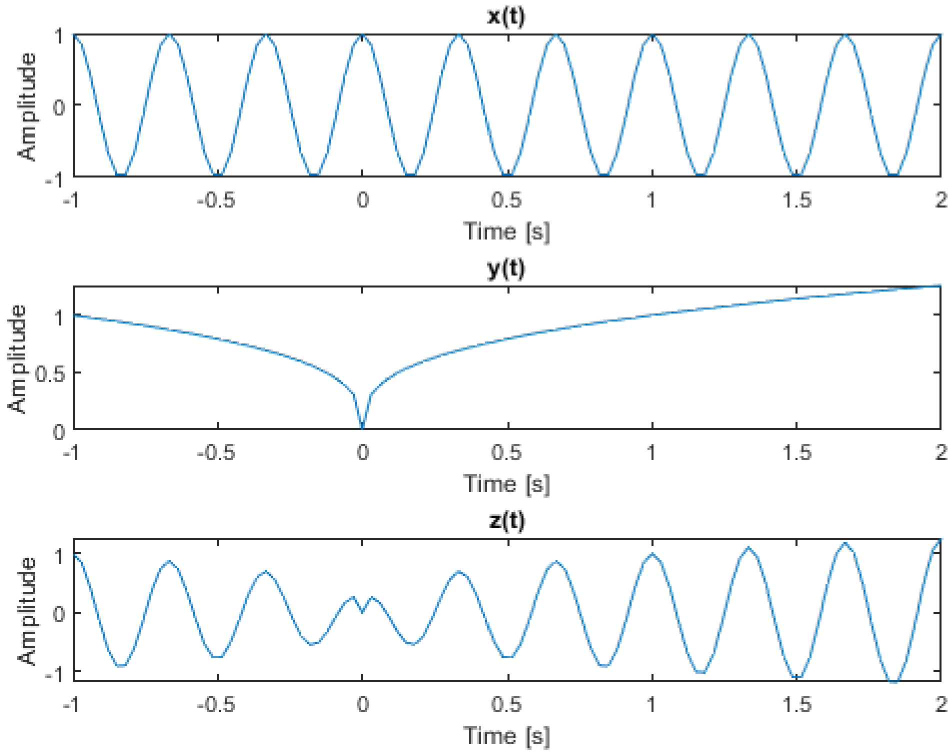
Generate the following three continuous-time signals for the time interval $-1 \le t \le 2$

- $x(t) = \cos(6\pi t)$
- $y(t) = |t|$^1/3
- $z(t) = x(t)y(t)$

Plot the three signals in the same window, but in three different figures, see `subplot`

```
task2 = figure('Name',"Task 2");
figure(task2);
time = linspace(-1,2,100);
x = @(t) cos(6.*pi.*t);
y = @(t) abs(t).^(1/3);
z = @(t) x(t).*y(t);
t = [-1,2];
title("Task 2");
subplot(3,1,1);
plot(time,x(time));
ylabel("Amplitude");    % Since there is no better lable
title("x(t)");
xlabel("Time [s]");
subplot(3,1,2);
plot(time,y(time));
ylabel("Amplitude");    % Since there is no better lable
title("y(t)");
```

```
xlabel("Time [s]");
subplot(3,1,3);
plot(time,z(time));
ylabel("Amplitude");    % Since there is no better lable
title("z(t)");
xlabel("Time [s]");
```



Task 3

Repeat Exercise 2 for the three discrete-time signals obtained by sampling x(t), y(t),

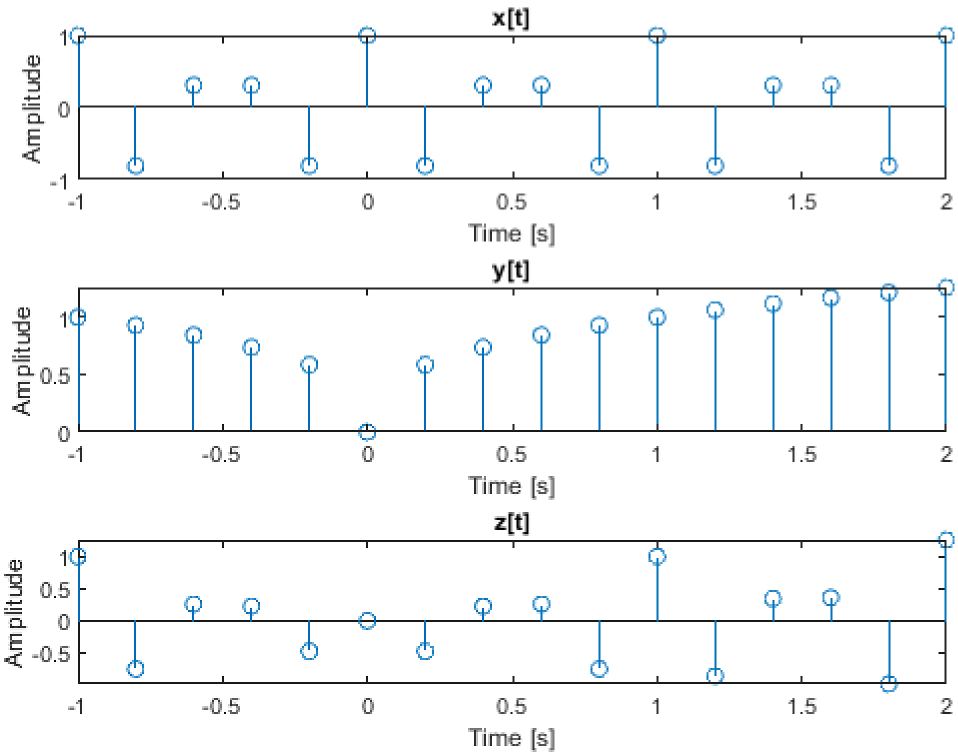
and z(t) using Ts = 1/5 s, i.e., illustrate

- x[n] = x(nTs)
- y[n] = y(nTs)
- z[n] = z(nTs)

Remember to label the axes carefully

```
task3 = figure("Name","Task 3");

times = -1:1/5:2;
disc_x = x(times);
disc_y = y(times);
disc_z = z(times);
subplot(3,1,1);
stem(times,disc_x);
ylabel("Amplitude");    % Since there is no better lable
```

```
xlabel("Time [s]");
title("x[t]");
subplot(3,1,2);
stem(times,disc_y);
ylabel("Amplitude");    % Since there is no better lable
xlabel("Time [s]");
title("y[t]");
subplot(3,1,3);
stem(times,disc_z);
ylabel("Amplitude");    % Since there is no better lable
xlabel("Time [s]");
title("z[t]");
```



We can see that some information is lost in the sampeling process, this is logical since we don't sample at the propper sample rate.

Task 4

A particular LTI system, call it System 1, is described through the input-output

relation

y[n] =1/8(x[n] + x[n −1] + x[n −2] + x[n −3] + x[n −4] + x[n −5] + x[n −6] + x[n −7])

Find the impulse response, h1[n], of System 1 (analytically).

Simple,

$$Y[n] = h[n] * x[n] \Rightarrow h[n] = \frac{1}{8}(\delta[n] + \delta[n-1] + \delta[n-2] + \delta[n-3] + \delta[n-4] + \delta[n-5] + \delta[n-6] + \delta[n-7])$$

And expressed in code it is

```
h = (1./8)*ones([1,8]);
h_handle = @(n) (n<=7 & n>= 0).*(1./8);
differential_signal = (1./8)*ones([1,8]);
```

Helpers

```
unit_sig = @(start_p,end_p,delay) start_p:1:end_p == delay;
step_sig = @(start_p,end_p,plat_start,plat_end) (start_p:1:end_p <= plat_end) - (start_p:1:end_
convolve = @(x,h,n,k_start,k_end) h(k_start:1:k_end).*(x(n-(k_end-k_start):1:n));
```

Task 5

Use Matlab to find the output obtained by feeding the input signal

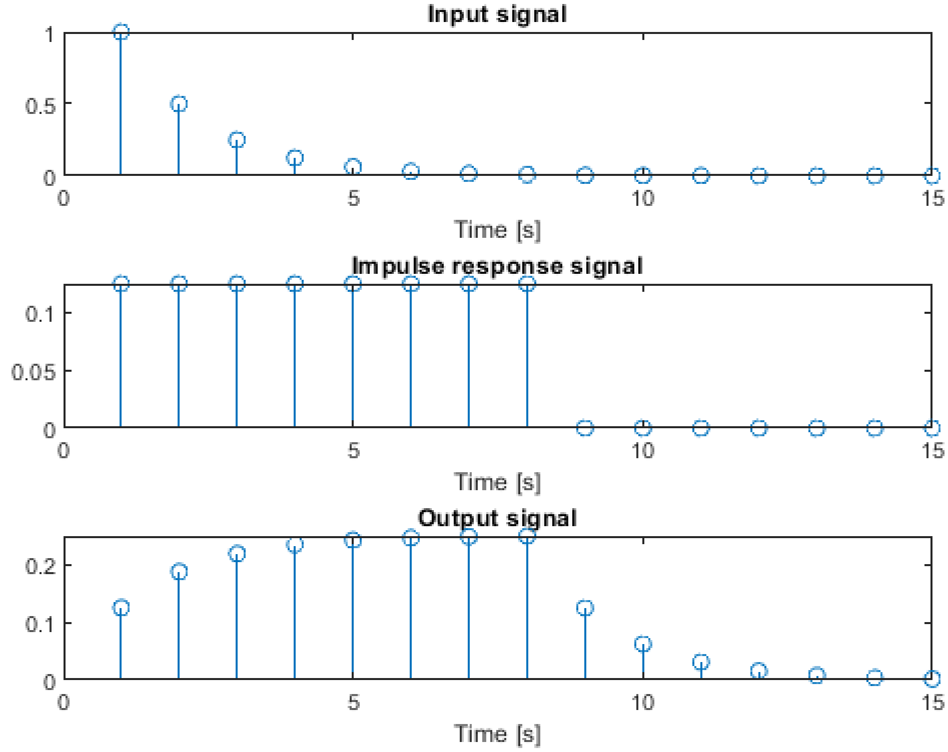
x[n] =.5^n

for 0 ≤n ≤10

0 otherwise

through System 1. Illustrate the input and output signals in an appropriate way.

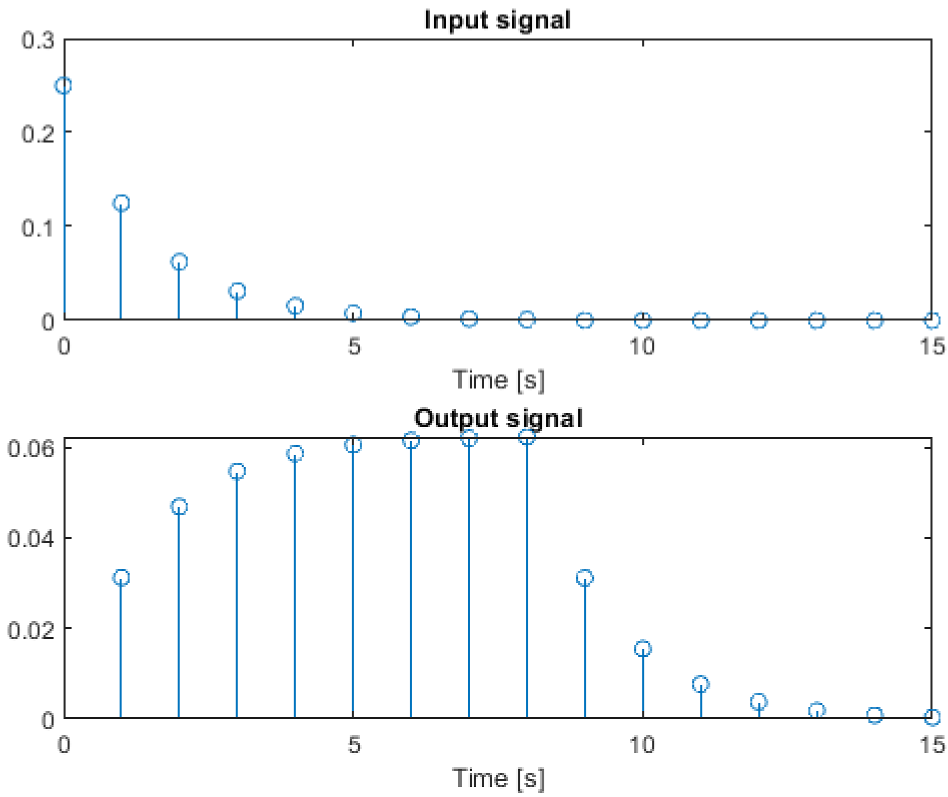
```
time_steps = 15;
time = 0:1:time_steps;
task5 = figure("Name","task 5");
% Define a new input signal
sig_in = @(x) (x <= 10 ).*( x >= 0).*(0.5.^x);

out = conv(sig_in(time),h_handle(0:1:8),"full");
% Plot things
figure(task5);
subplot(3,1,1);
stem(sig_in(time));
xlim([0,time_steps])
xlabel("Time [s]");
title("Input signal");
subplot(3,1,2);
stem(h_handle(time));
xlim([0,time_steps])
xlabel("Time [s]");
title("Impulse response signal");
subplot(3,1,3);
stem(out);
xlim([0,time_steps])
xlabel("Time [s]");
title("Output signal");
```



Task 6

Repeat Exercise 5, but this time by using the input signal x2[n] = x[n + 2].

```
time = 0:1:time_steps;
% Define the new input signal
sig_in_2 = @(n)sig_in(n+2);
out = conv(sig_in_2(time),h_handle(0:1:8));
task6 = figure("Name","task 6");


% Plot things
figure(task6);
subplot(2,1,1);
stem(time,sig_in_2(time));
xlim([0,time_steps])
xlabel("Time [s]");
title("Input signal");
subplot(2,1,2);
plt = stem(out);
xlim([0,time_steps])

xlabel("Time [s]");
title("Output signal");
```



We can se that the output has been shifted over by 2, this is because the input was shifted over by 2.

Task 7

Repeat Exercise 5, but this time consider a system with impulse response

h2[n] = h1[−n]

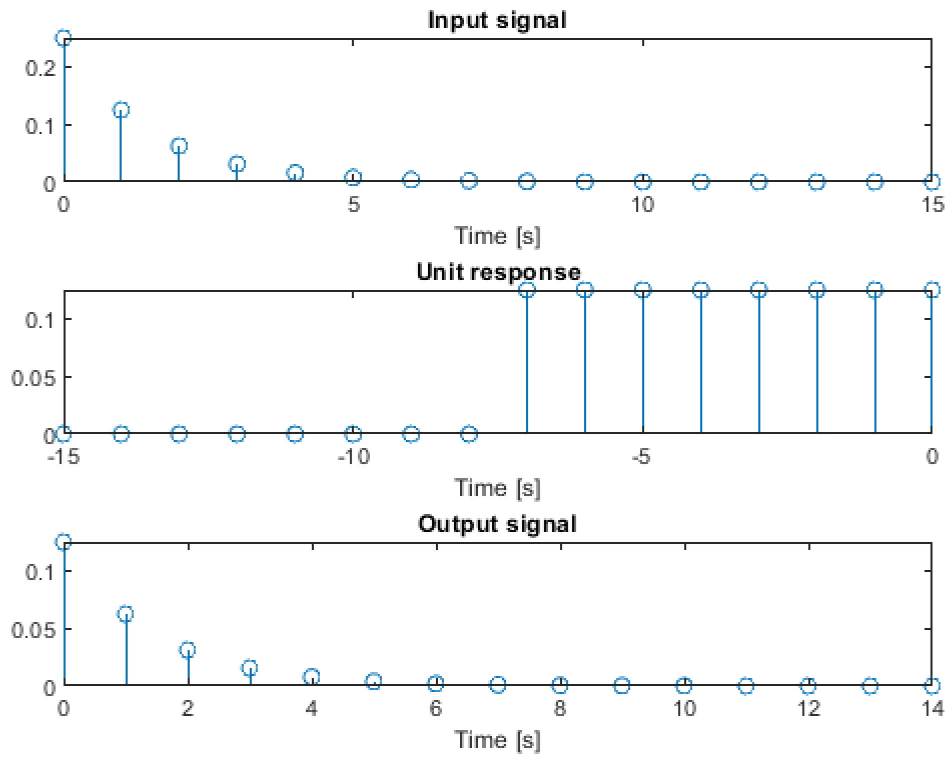
```
h2_handle = @(n) h_handle(-n);
```

Now the impulse response is

$$h[n] = \frac{1}{8}(\delta[-n] + \delta[-n-1] + \delta[-n-2] + \delta[-n-3] + \delta[-n-4] + \delta[-n-5] + \delta[-n-6] + \delta[-n-7])$$

```
time = 0:1:time_steps;
out = conv(sig_in(time),h2_handle(time));



% Plot things
task7 = figure("Name","task 6");
figure(task7);
subplot(3,1,1);
stem(time,sig_in_2(time));
xlabel("Time [s]");
```

```
title("Input signal");
xlim([0,time_steps])
subplot(3,1,2);
stem(-time,h2_handle(-time));
xlabel("Time [s]");
title("Unit response");
subplot(3,1,3);
stem(time(1:time_steps),out(1,1:time_steps));
xlabel("Time [s]");
title("Output signal");
```



Now it only scales down n since it can't be a rolling average filter anny more. this is because -x where x is some offset never equals zero for k >= 0 except for when x = 0 so it only scales the current value.

Task 8

Repeat Exercise 5 but this time by using the input signal

x[n] = cos (π/8 n) + cos(π/4 n) for 0 ≤n ≤127

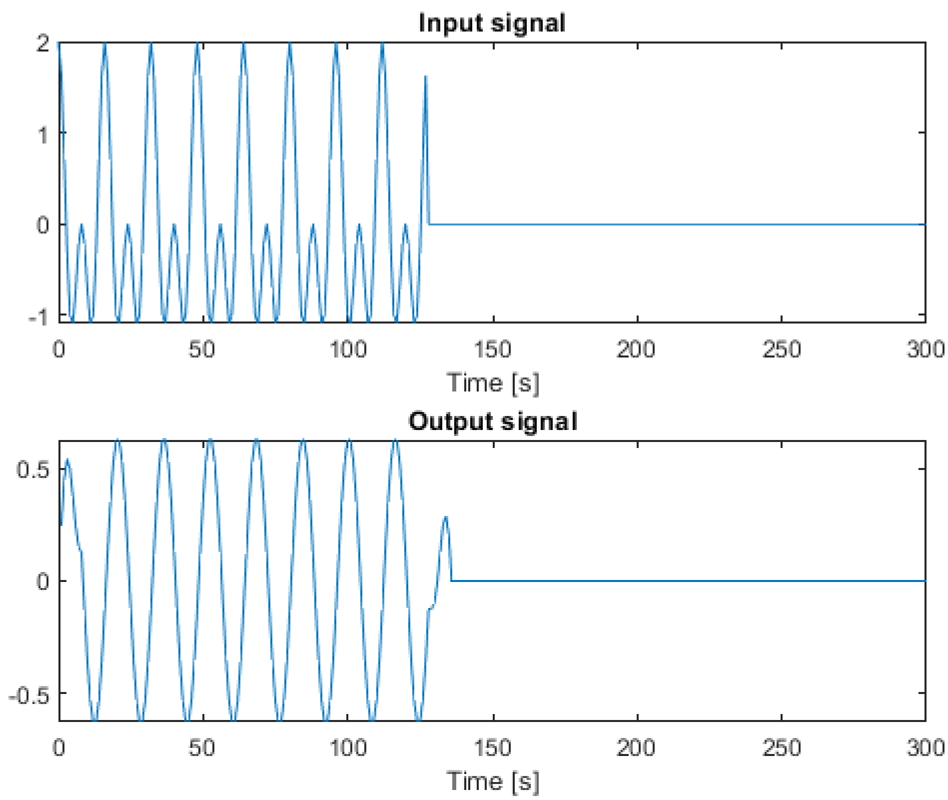
0 otherwise

Comment on the results, especially regarding the frequency content of the input and

output signals. Can you explain this behavior?

```
time_steps = 300;
time = 0:1:time_steps;
sig = @(n)(cos((pi/8).*n) + cos((pi/4).*n)).*(n <=127 & n >= 0);
```

```
out = conv(sig(time),h_handle(time),"full");


task8 = figure("Name","task 8");
% Plot things
figure(task8);
subplot(2,1,1);
plot(time,sig(time));
xlabel("Time [s]");
title("Input signal");
subplot(2,1,2);
plot(out);
xlim([0,time_steps])
xlabel("Time [s]");
title("Output signal");
```



It does not change the frequency since it's just a filter, this is one of the fundemental properties of FIR filters.

It does however change the amplitude and introduces a phase shift. this is due the fact that the rate of change is lower than the input

Task 9

In the file `eva.mat`, two Matlab variables `xl` and `xr` can be found. You can load these

into the workspace through the load command, see `help load`. `xl` and `xr` contains

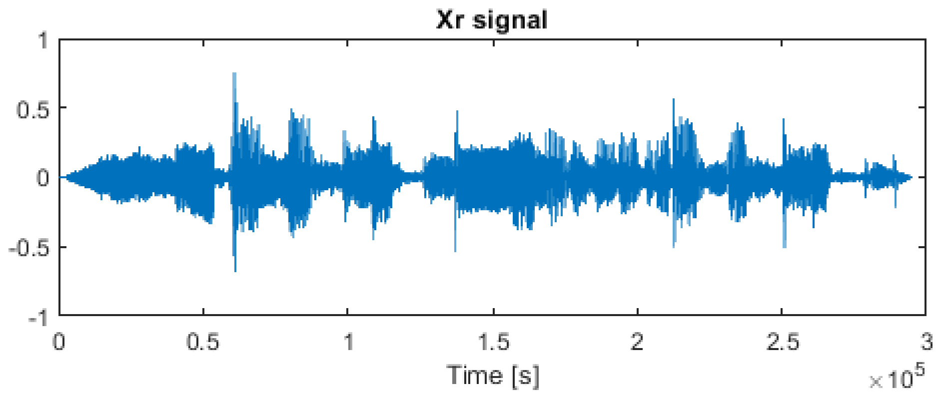
the left and right channel of an audio segment sampled at 44,1 kHz. You can listen to it in Matlab using the `sound` command,

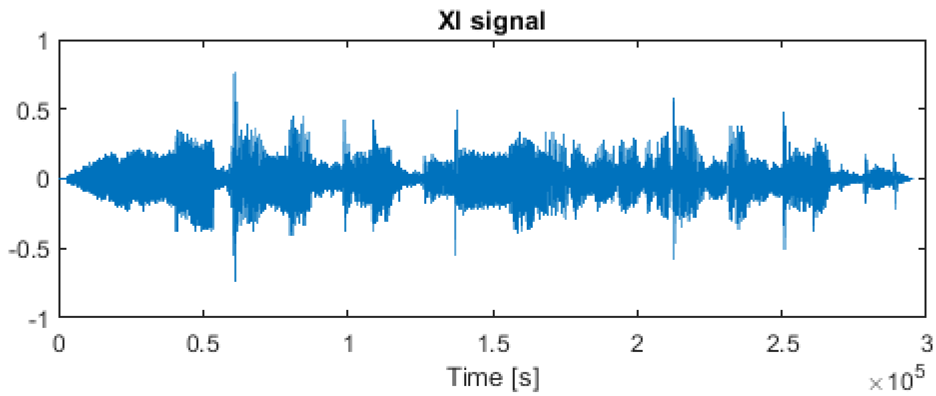
`sound([xl,xr],44100)`

Feed both channels through System 1 and listen to the result. Comment briefly

on how you experience the processed audio signal.

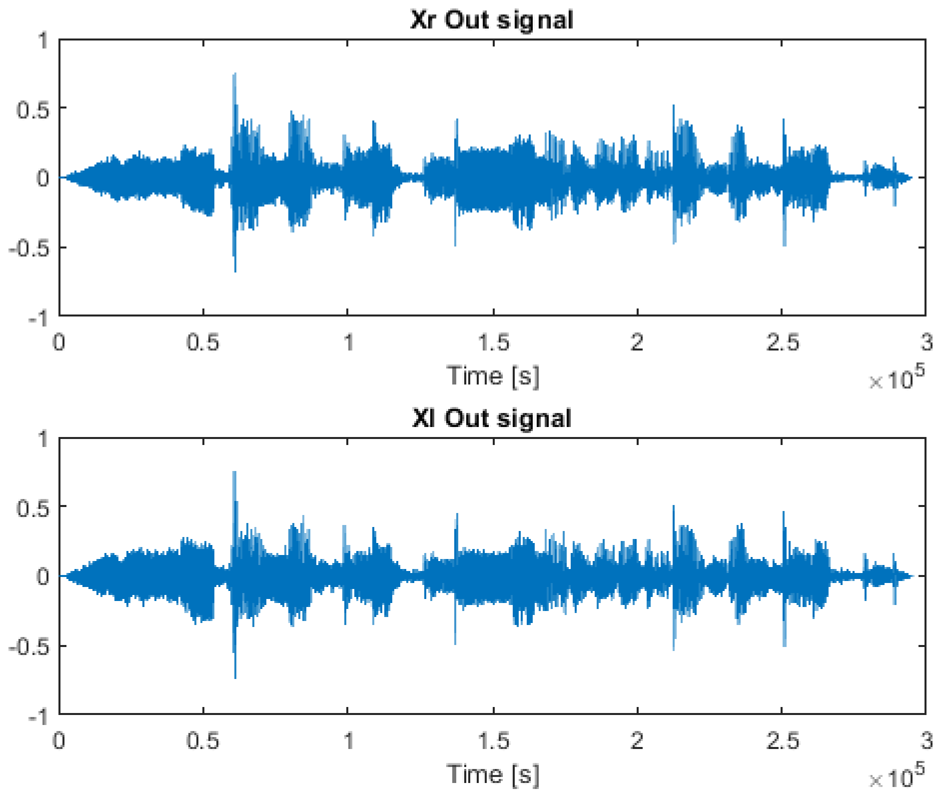
```
% Load the data
data = load("eva.mat",'xr','xl');
% Split the field
xr = data.xr;
xl = data.xl;
% Create output vectors for the system
out_xr = conv(xr,h_handle(0:1:10),"full");
out_xl = conv(xl,h_handle(0:1:10),"full");
% Convolve over the input data



% Plot original signal
task9 = figure("Name","task 9");
figure(task9);
subplot(2,1,1);
plot(1:length(xr),xr);
xlabel("Time [s]");
title("Xr signal");
subplot(2,1,2);
plot(1:length(xr),xl);
xlabel("Time [s]");
title("Xl signal");
```





And after the system we get

```
% Plot the output signals
task9_2 = figure("Name","task 9 2");
figure(task9_2);
subplot(2,1,1);
plot(out_xr);
xlabel("Time [s]");
title("Xr Out signal");
subplot(2,1,2);
plot(out_xl);
xlabel("Time [s]");
title("Xl Out signal");
```




```
% Play the sound
%sound([xl,xr],44100);
%sound([out_xl,out_xr],44100);
```

The processed audio seems more quiet, it also seems a bit more "muddy" as in it has been smoothed out a bit, removing some of the audio detail.