

3TP3: Signals & Systems

Lab #2

Instructor: Dr. Kirubarajan

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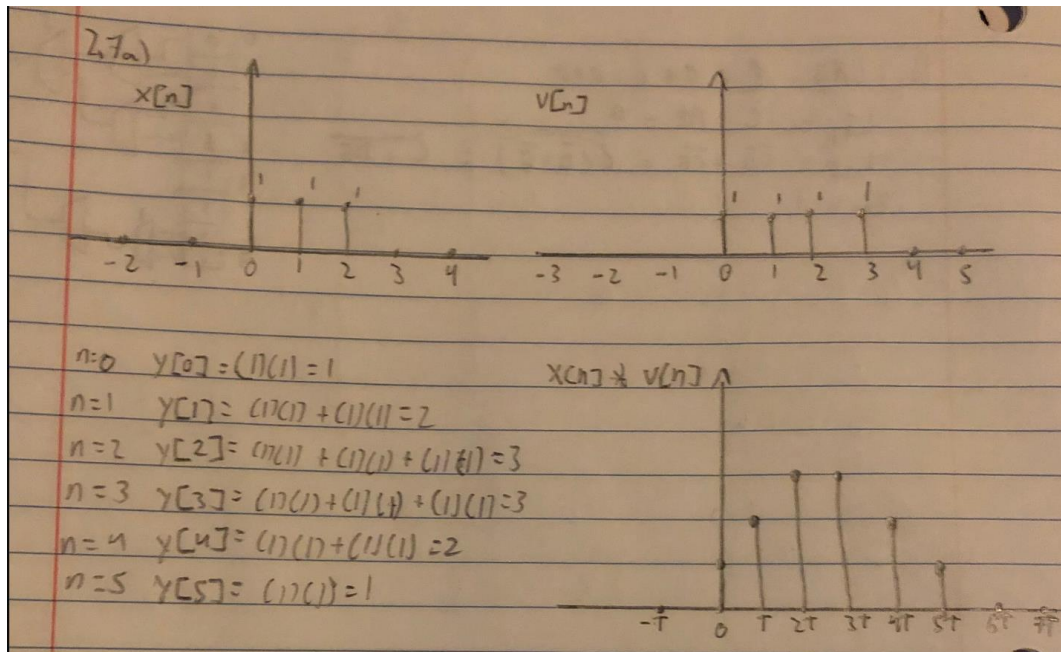
Samer Rafidi – L02 – rafidis- 400333524

As a future member of the engineering profession, the student is responsible for performing the required work in an honest manner, without plagiarism and cheating. Submitting this work with my name and student number is a statement and understanding that this work is my own and adheres to the Academic Integrity Policy of McMaster University and the Code of Conduct of the Professional Engineers of Ontario. Submitted by [**Tamer Rafidi, rafidit, 400333527**]

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Question 1

1a)



$$\begin{aligned}
 n=0 \quad y[0] &= (1)(1) = 1 \\
 n=1 \quad y[1] &= (1)(1) + (1)(1) = 2 \\
 n=2 \quad y[2] &= (1)(1) + (1)(1) + (1)(1) = 3 \\
 n=3 \quad y[3] &= (1)(1) + (1)(1) + (1)(1) = 3 \\
 n=4 \quad y[4] &= (1)(1) + (1)(1) = 2 \\
 n=5 \quad y[5] &= (1)(1) = 1
 \end{aligned}$$

```

%Set time ranges
t = -5:5;
t_2 = -10:10;

%Set two variables to SimpleFunctions() object
x = SimpleFunctions();
v = SimpleFunctions();

%Created x[n] and v[n]
x = x.unitstep(t)-x.unitstep(t-3);
v = v.unitstep(t)-v.unitstep(t-4);

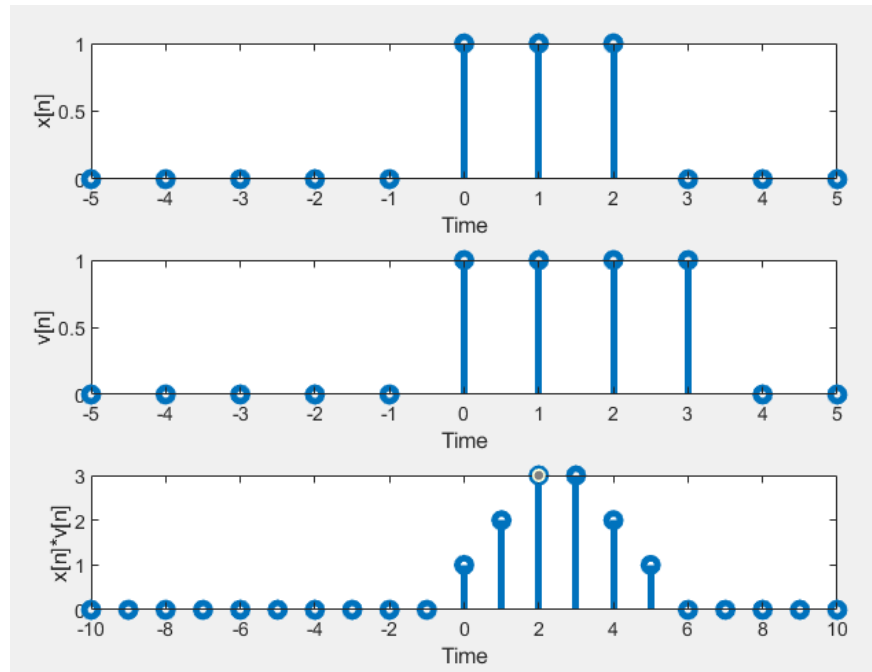
%Computing convolution of x[n] and v[n]
con = conv(x, v);

%Creating three subplots x[n], v[n], and x[n]*v[n]
subplot(3,1,1);
stem(t, x, "LineWidth",3);
xlabel('Time')
ylabel('x[n]')

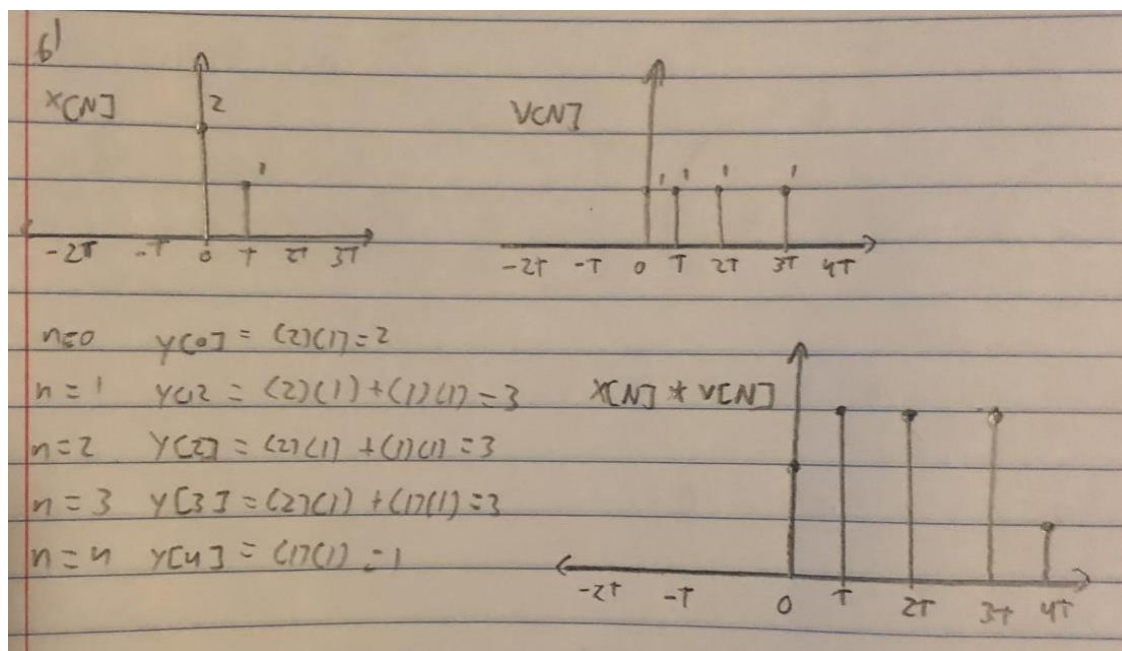
subplot(3,1,2);
stem(t, v, "LineWidth",3);
xlabel('Time')
ylabel('v[n]')

subplot(3,1,3);
stem(t_2, con, "LineWidth",3);
xlabel('Time')
ylabel('x[n]*v[n]')

```



1b)



```

%Set time ranges
t = -5:5;
t_2 = -10:10;

%Set two variables to SimpleFunctions() object
x = SimpleFunctions();
v = SimpleFunctions();

%Created x[n] and v[n]
x = 2*x.unitstep(t) - x.unitstep(t-1) - x.unitstep(t-2);
v = v.unitstep(t)-v.unitstep(t-4);

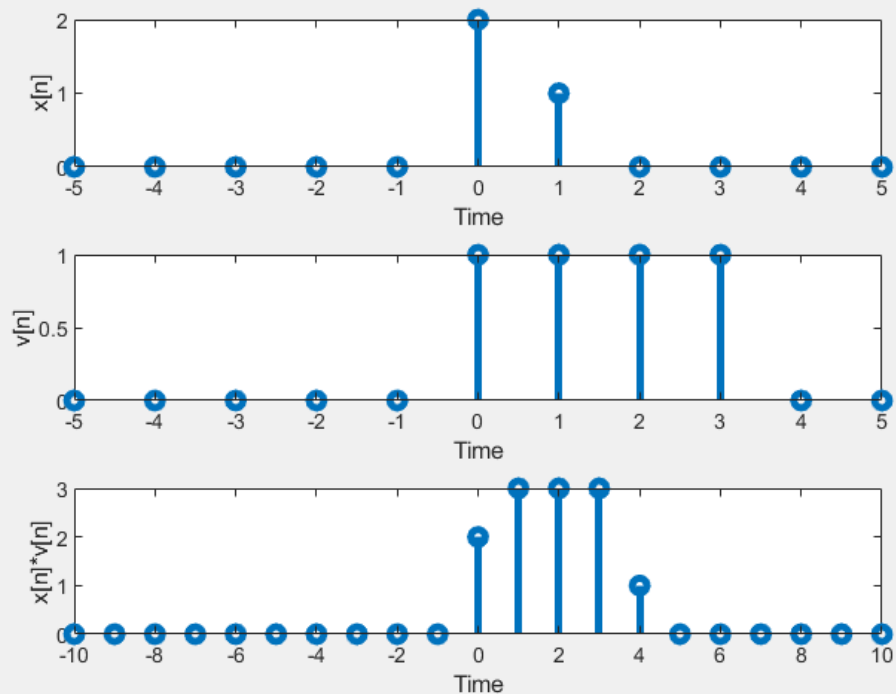
%Computing convolution of x[n] and v[n]
con = conv(x, v);

%Creating three subplots x[n], v[n], and x[n]*v[n]
subplot(3,1,1);
stem(t, x, "LineWidth",3);
xlabel('Time')
ylabel('x[n]')

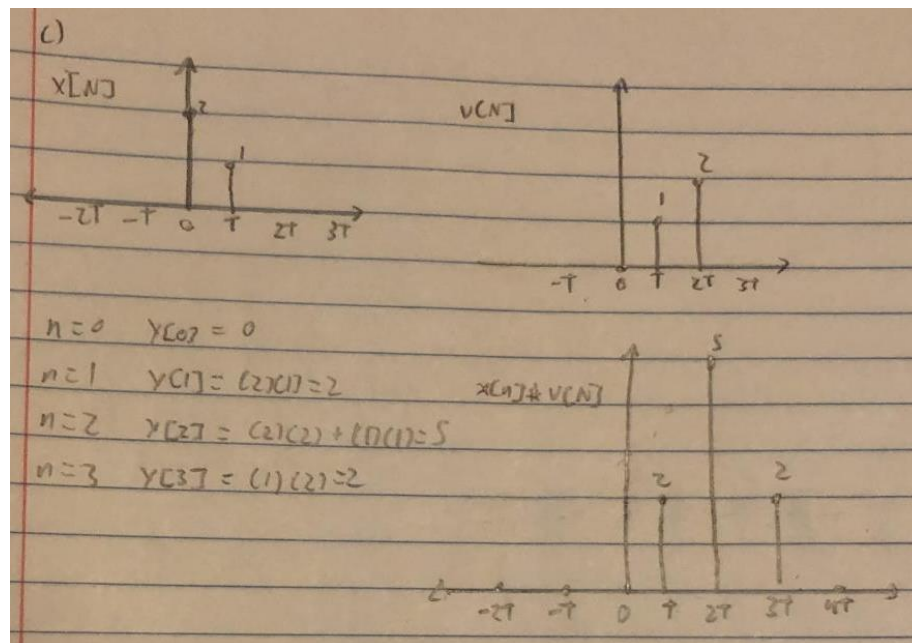
subplot(3,1,2);
stem(t, v, "LineWidth",3);
xlabel('Time')
ylabel('v[n]')

subplot(3,1,3);
stem(t_2, con, "LineWidth",3);
xlabel('Time')
ylabel('x[n]*v[n]')

```



1c)



```
%Set time ranges
```

```
t = -5:5;
```

```
t_2 = -10:10;
```

```
%Set two variables to SimpleFunctions() object
```

```
x = SimpleFunctions();
```

```
v = SimpleFunctions();
```

```
%Created x[n] and v[n]
```

```
x = 2*x.unitstep(t) - x.unitstep(t-1) - x.unitstep(t-2);
```

```
v = v.unitstep(t-1) + v.unitstep(t-2) - 2*v.unitstep(t-3);
```

```
%Computing convolution of x[n] and v[n]
```

```
con = conv(x, v);
```

```
%Creating three subplots x[n], v[n], and x[n]*v[n]
```

```
subplot(3,1,1);
```

```
stem(t, x, "LineWidth",3);
```

```
xlabel('Time')
```

```
ylabel('x[n]')
```

```
subplot(3,1,2);
```

```
stem(t, v, "LineWidth",3);
```

```
xlabel('Time')
```

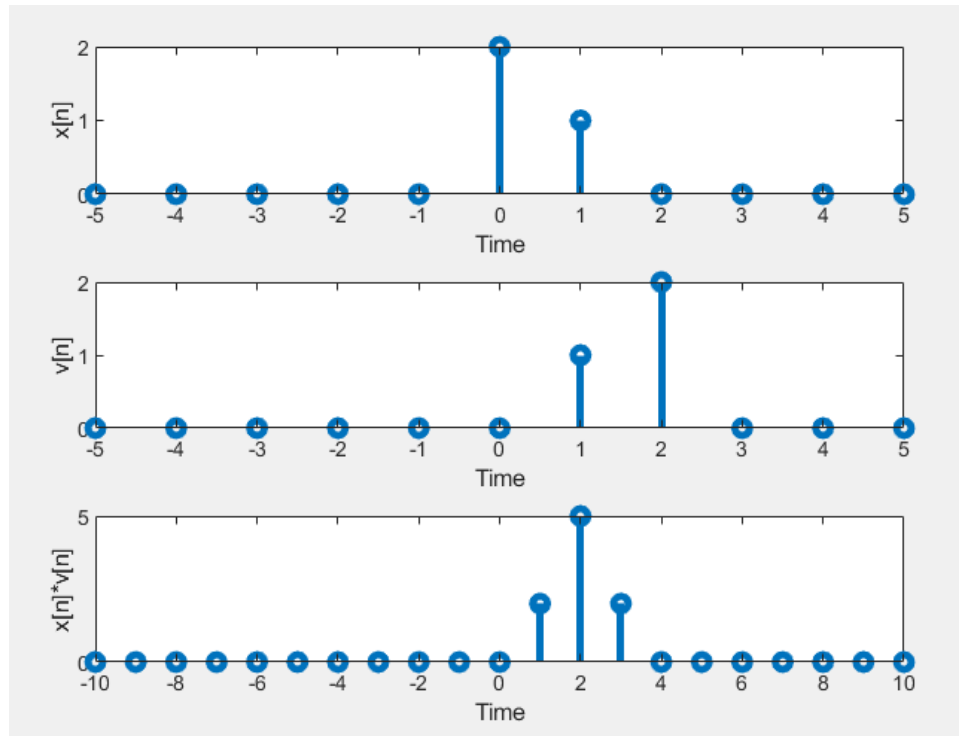
```
ylabel('v[n]')
```

```
subplot(3,1,3);
```

```
stem(t_2, con, "LineWidth",3);
```

```
xlabel('Time')
```

```
ylabel('x[n]*v[n]')
```



Question 3

```

1  [signal, Fs] = audioread('my_speech_clip.wav');
2
3  L = length(signal); % Number of samples in the signal.
4  T = 1/Fs; % Sampling period in seconds.
5  t = (0:L-1)*T; % Time vector in seconds.
6

```

Question 4

```
1 [signal, Fs] = audioread('my_speech_clip.wav');
2
3 L = length(signal); % Number of samples in the signal.
4 T = 1/Fs; % Sampling period in seconds.
5 t = (0:L-1)*T; % Time vector in seconds.
6
7 % Creates variable for alpha and echo delay(msec).
8 Te = 1000;
9 alpha = 0.3;
10
11 %Creating a variable that keeps track of the number of samples delayed.
12 S_delay = round((Te/1000) / T);
13
14 %Vector created to store echos
15 signalplusecho = zeros(L+S_delay, 1);
16 signalplusecho(1:L) = signal;
17
18 %Echo is added
19 signalplusecho(S_delay+1:end) = signalplusecho(S_delay+1:end) + alpha*signal;
20 %Rescales to ensure that the magntiude doesn't exceed 1 (avoids clipping)
21 signalplusecho = signalplusecho/max(abs(signalplusecho));
22
23 % Used to check the magntiude to make sure that no sample exceeds 1
24 magnitude = sqrt(signalplusecho);
25 magnitude;
26
27 audiowrite('speechwithecho.wav', signalplusecho, Fs);
```

Question 5

```
1 [signal, Fs] = audioread('my_speech_clip.wav');
2
3 L = length(signal); % Number of samples in the signal.
4 T = 1/Fs; % Sampling period in seconds.
5 t = (0:L-1)*T; % Time vector in seconds.
6
7 % Creates variable for alpha and echo delay(msec).
8 Te = 250;
9 alpha = 0.3;
10
11 %Creating a variable that keeps track of the number of samples delayed.
12 S_delay = round((Te/1000) / T);
13
14 %Vector created to store impulse response
15 IR = zeros(S_delay+1,1);
16 %Value is set to 1 so we don't lose original signal
17 IR(1) = 1;
18 %The impulse response at the delayed time is set to alpha
19 IR(S_delay) = alpha;
20
21
22 %Echo is added after convolution
23 signalplusecho = conv(signal, IR);
24 %Rescales to ensure that the magntiude doesn't exceed 1 (avoids clipping)
25 signalplusecho = signalplusecho/max(abs(signalplusecho));
26
27 % Used to check the magntiude to make sure that no sample exceeds 1
28 magnitude = sqrt(signalplusecho);
29 magnitude;
30
31 audiowrite('speechwithconvo.wav', signalplusecho, Fs);
```

Question 7

```
1 [signal, Fs] = audioread('my_speech_clip.wav');
2
3 L = length(signal); % Number of samples in the signal.
4 T = 1/Fs; % Sampling period in seconds.
5 t = (0:L-1)*T; % Time vector in seconds.
6
7 % Creates variable for alpha and echo delay(msec).
8 Te = 250;
9 Ne = 2; % Number of echos
10 alpha = 0.3;
11
12 %Creating a variable that keeps track of the number of samples delayed.
13 S_delay = round((Te/1000) / T);
14
15 %Vector created to store impulse response
16 IR = zeros(Ne*S_delay+1,1);
17 %Value is set to 1 so we don't lose original signal
18 IR(1) = 1;
19 %The impulse response at the delayed time is set to alpha
20 IR(S_delay) = alpha*Ne;
21
22 %For loop that runs from 0 to Ne
23 for j = 0: Ne
24     IR(j * S_delay+1) = alpha^(j); %IR is set to alpha ^ j
25 end
26
27 %Echo is added after convolution
28 signalplusecho = conv(signal, IR);
29 %Rescales to ensure that the magntiude doesn't exceed 1 (avoids clipping)
30 signalplusecho = signalplusecho/max(abs(signalplusecho));
31
32 % Used to check the magntiude to make sure that no sample exceeds 1
33 magnitude = sqrt(signalplusecho);
34 magnitude;
35
36 audiowrite('speechwithreverb.wav', signalplusecho, Fs);
```


4) Report

Question #1 asks us to manually compute the convolutions of $x[n]$ and $v[n]$ in question 2.7 from the textbook, then verify our results using MATLAB. On MATLAB, we use the `SimpleFunctions()` function from lab 1 to create $x[n]$ and $v[n]$ from the questions. Then we use the built-in function `conv` to compute the convolution for $x[n]$ and $v[n]$. We then set the length of the x-axis to range from -5 to 5 for $x[n]$ and $v[n]$, as well as set another time variable from -10 to 10 for $x[n]*v[n]$. These variables are used in the stem function when creating the subplots for $x[n]$, $v[n]$, and the convolution of $x[n]*v[n]$. After plotting the graph, we use `xlabel` and `ylabel` to complete our plot.

Question #4 asks us to create an echo and put that echo into our voice recording. First, I start by creating T_e which is the time when the echo will start. α is created and this is used to scale the amplitude of the signal. I use S_delay to keep track of how many signals are going to be delayed. `Signalplusecho` is the audio that will be distorted.

Question #5 asks us to create an echo using convolution. First, we create a vector (IR) which stores the impulse response. We set it to 1 so we can store the original value of signal and not lose that. Then, we set α to the delayed time. Lastly, we use the convolution function to create the new distorted sound and then we scale it to ensure that no sound is lost in the process.

Question #6 asks us to make $\alpha = 1$ and see at which value of T_e we get an acceptable sound. I tested values from 500 msec down to 10 msec. In my testing, I heard the echo throughout all the values, however at $T_e = 20$ is the maximum I can go before I hear the echo again. $T_e = 30$ was close however the echo was still noticeable. When decreasing the α , I noticed that I could have a higher value of T_e without the echo being noticeable.

Question #7 asks us to create an echo using reverberation. This was done by using a for loop which created N_e amount of impulse responses. Repeating #6, I observed that the same pattern was very close in this code. I heard the echo throughout all the values, however it was very low at $T_e = 30$. At $T_e = 40$, the echo was noticeable. When I decreased the α value, I did not hear a change.