

ELECENG 3TP3 – Signals and Systems

Lab 2: Signal Convolution

Instructor: Dr. Kiruba

Prepared and submitted by
Marryam Kamal – kamalm18 – 400446997

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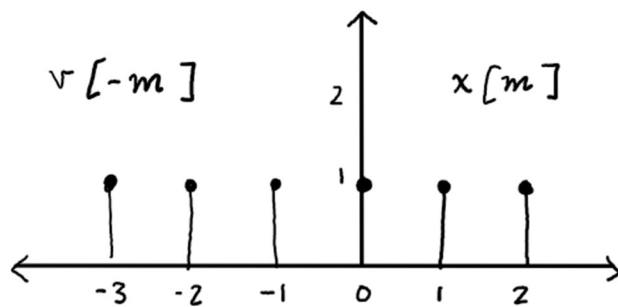
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Part 1 – Textbook Question 2.7

1) Manual Convolution

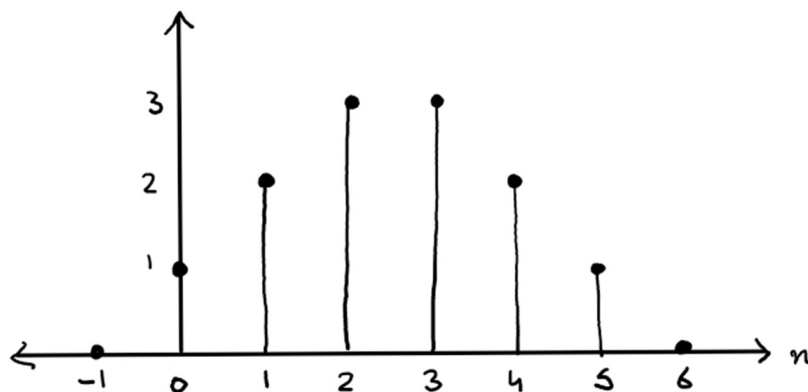
$$2.7 \text{ a)} \quad x[n] * v[n] = \sum_{m=-\infty}^{\infty} x[m] v[n-m]$$



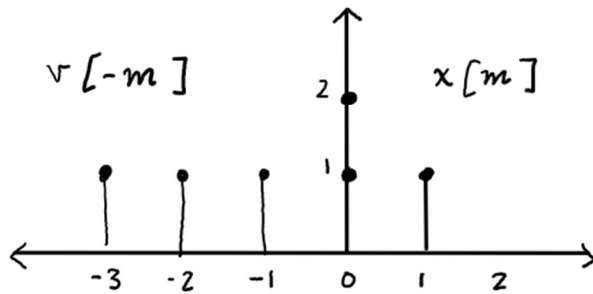
- no overlap for $n < 0$
- no overlap for $n \geq 6$

n	0	1	2	3	4	5	6
$x[n] * v[n]$	1	2	3	3	2	1	0

final
answer



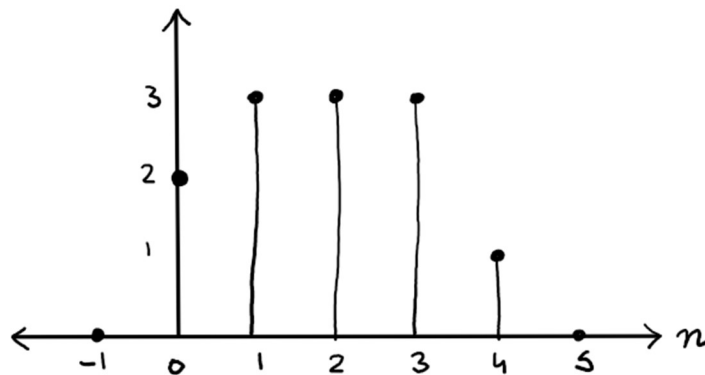
2.7 b) $x[n] * v[n] = \sum_{m=-\infty}^{\infty} x[m] v[n-m]$



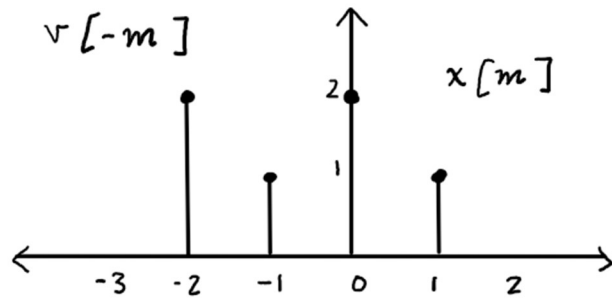
- no overlap for $n < 0$
- no overlap for $n \geq 5$

n	0	1	2	3	4	5
$x[n] * v[n]$	2	3	3	3	1	0

final
answer

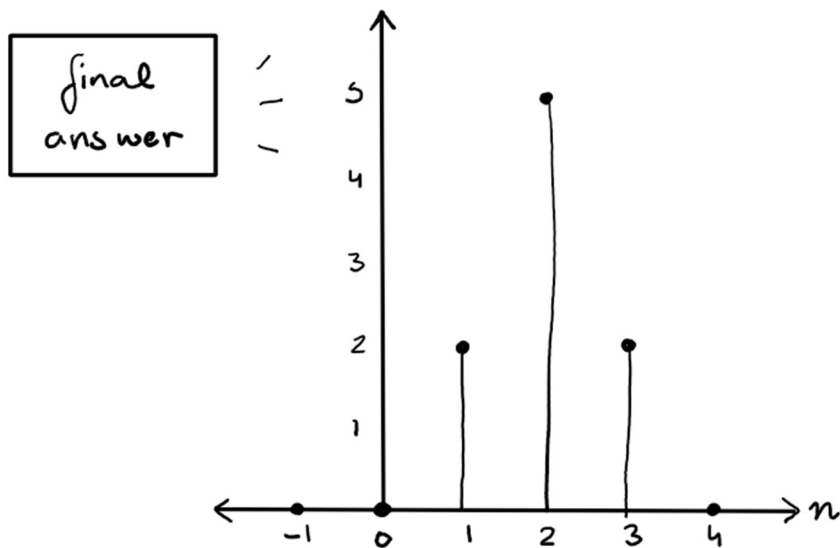


2.7 c) $x[n] * v[n] = \sum_{m=-\infty}^{\infty} x[m] v[n-m]$



- no overlap for $n < 1$
- no overlap for $n \geq 4$

n	1	2	3	4
$x[n] * v[n]$	2	5	2	0



2) Convolution via MATLAB

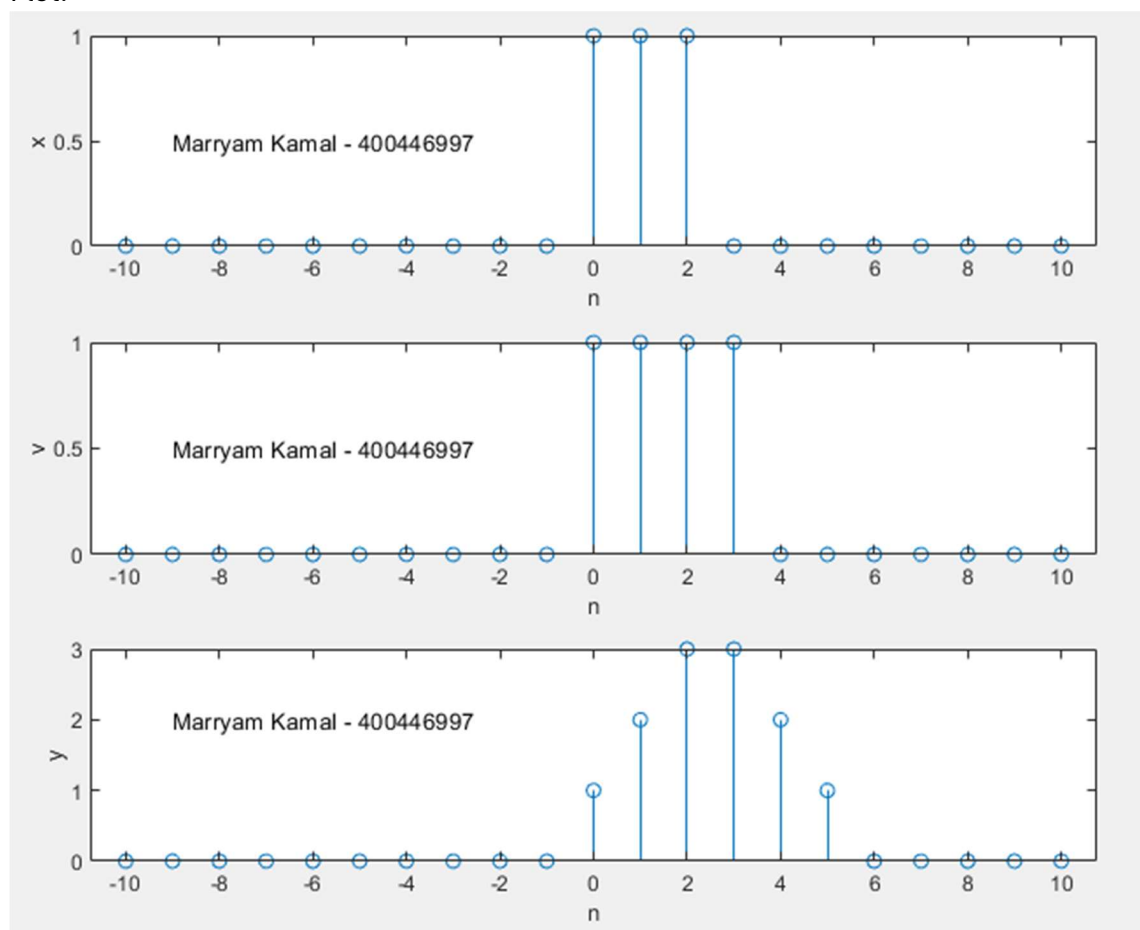
a) Code:

```

% Question 2.7A
n = -10:1:10;
x = f.unitstep(n) - f.unitstep(n-3);
v = f.unitstep(n) - f.unitstep(n-4);
y = conv(x, v, "same");
subplot(3,1,1); stem(n,x);
xlabel('n'); ylabel('x');
text(-9, 0.5, 'Marryam Kamal - 400446997', 'FontSize', 10)
subplot(3,1,2); stem(n,v);
xlabel('n'); ylabel('v');
text(-9, 0.5, 'Marryam Kamal - 400446997', 'FontSize', 10)
subplot(3,1,3); stem(n,y);
xlabel('n'); ylabel('y');
text(-9, 2, 'Marryam Kamal - 400446997', 'FontSize', 10)

```

Plot:



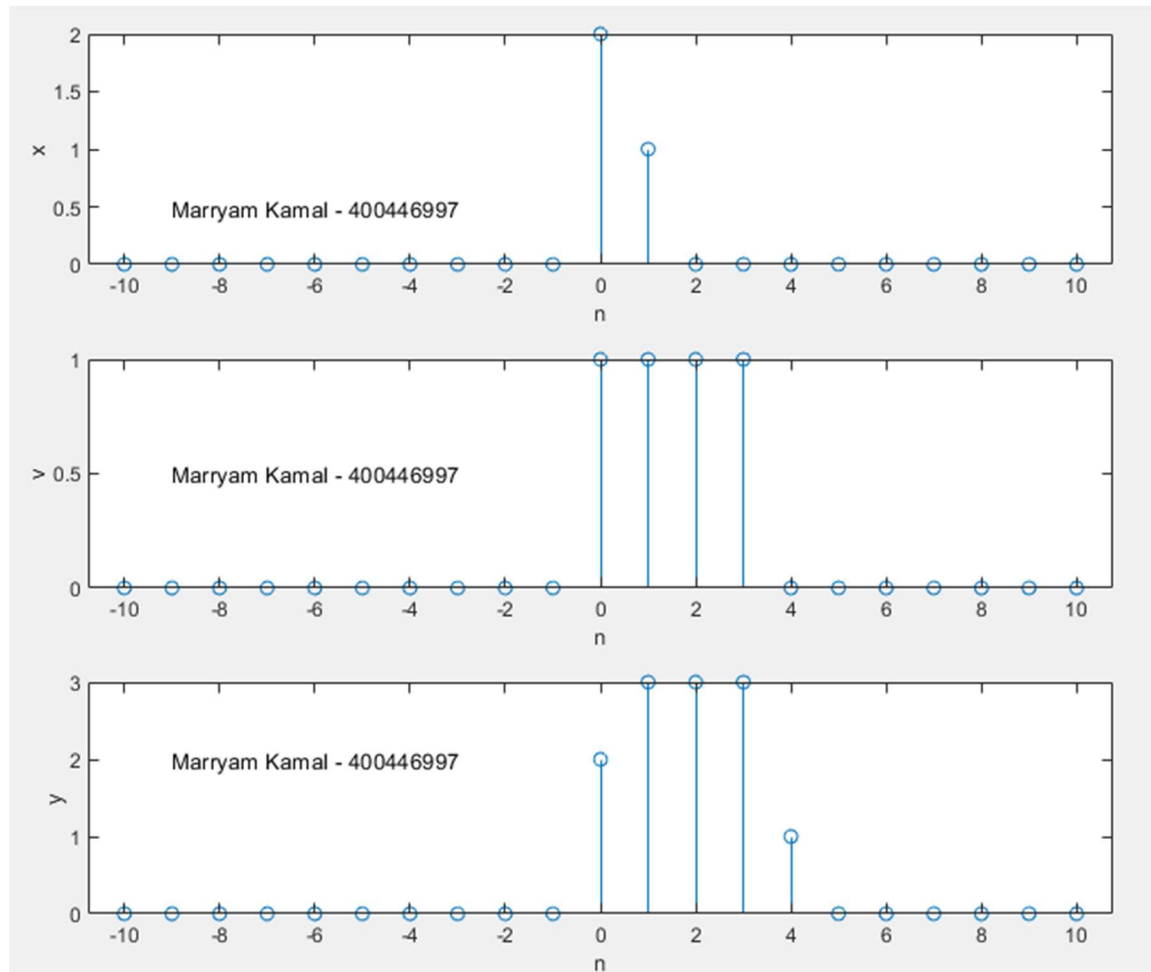
b) Code:

```

% Question 2.7B
n = -10:1:10;
x = 2*f.deltan(n) + f.deltan(n-1);
v = f.unitstep(n) - f.unitstep(n-4);
y = conv(x, v, "same");
subplot(3,1,1); stem(n,x);
xlabel('n'); ylabel('x');
text(-9, 0.5, 'Marryam Kamal - 400446997', 'FontSize', 10)
subplot(3,1,2); stem(n,v);
xlabel('n'); ylabel('v');
text(-9, 0.5, 'Marryam Kamal - 400446997', 'FontSize', 10)
subplot(3,1,3); stem(n,y);
xlabel('n'); ylabel('y');
text(-9, 2, 'Marryam Kamal - 400446997', 'FontSize', 10)

```

Plot:

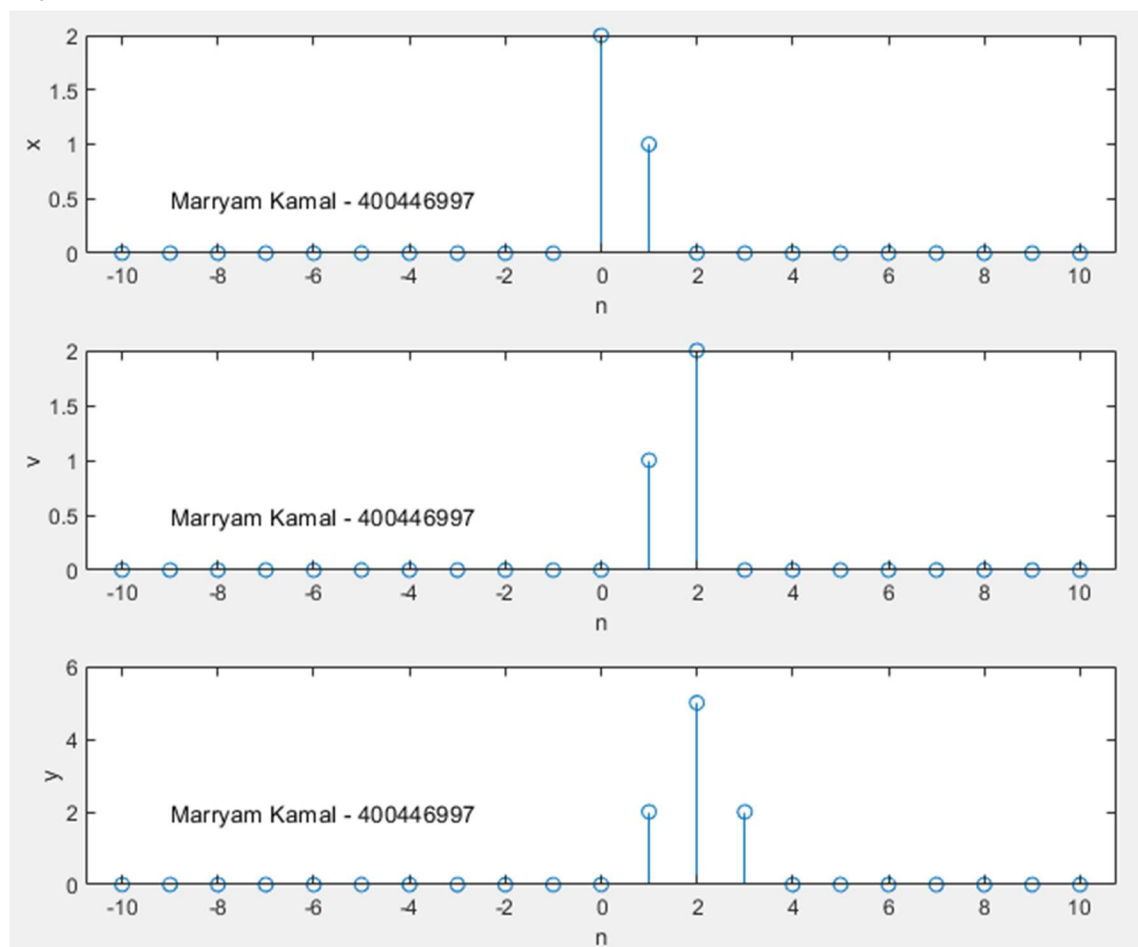


c) Code:

% Question 2.7C

```
n = -10:1:10;  
x = 2*f.delta(n) + f.delta(n-1);  
v = f.delta(n-1) + 2*f.delta(n-2);  
y = conv(x, v, "same");  
subplot(3,1,1); stem(n,x);  
xlabel('n'); ylabel('x');  
text(-9, 0.5, 'Marryam Kamal - 400446997', 'FontSize', 10)  
subplot(3,1,2); stem(n,v);  
xlabel('n'); ylabel('v');  
text(-9, 0.5, 'Marryam Kamal - 400446997', 'FontSize', 10)  
subplot(3,1,3); stem(n,y);  
xlabel('n'); ylabel('y');  
text(-9, 2, 'Marryam Kamal - 400446997', 'FontSize', 10)
```

Plot:



Part 2 – Audio Processing

3) Code:

```
% Audio importing
[signal, Fs] = audioread('my_speech_clip.wav');
L = length(signal);
T = 1/Fs;
t = [0:L-1]*T;
```

4) Code:

```
a = 0.5; % Echo amplitude
Te = 200; % Echo delay in msec
Te_samples = round((Te/1000) * Fs); % Convert echo delay to samples

signal(Te_samples+1:end) = signal(Te_samples+1:end) + a*signal(1:end-Te_samples);
signal = signal / max(abs(signal));

% New audio file
audiowrite('speechwithecho.wav', signal, Fs);
```

5) Choice of Impulse Response:

An echo has 2 components: the original sound and a scaled, delayed version. To achieve this, the impulse response I must convolve the original signal with must overlap in those two locations. This means it must have a delta function at $t = 0$, and then one at $t = T_e$, the delay. Doing so creates the following formula.

$$IR = \delta(t) + a\delta(t - T_e)$$

And so, I created a delta function IR which has a value of $\delta(t) = 1$ at $t = 0$ and $\delta(t) = a$ at $t = T_e$.

Code:

```
a = 0.5; % Echo amplitude
Te = 200; % Echo delay in msec
Te_samples = round((Te/1000) * Fs); % Convert echo delay to samples

IR = zeros(1, Te_samples + 1);
IR(1) = 1; % Delta function at t = 0
IR(Te_samples + 1) = a;

signal = conv(signal, IR, 'full'); % Convolution
signal = signal / max(abs(signal));

% New audio file
audiowrite('echo_convolution.wav', signal, Fs);
```

6) Changing T_e and a :

Keeping $a = 1$, I found that the signal sounded the best on values of $T_e < 10$ ms. The shorter the time between the original speech and echo, the better it will sound. When I decreased a to 0.1, I found that the signal sounded best on values of $T_e < 100$ ms. From this, I can tell that lowering the amplitude, or volume of the audio can further reduce the echo in it.

7) Reverberation:

To achieve reverberation, I first created a vector function impulse response (IR) and then added N_e delta functions to it, each with a decreasing amplitude up to T_e . Convolution with the original resulted in the reservation I needed.

Code:

```
a = 0.5; % Echo amplitude
Te = 200; % Echo delay in msec
Te_samples = round((Te/1000) * Fs); % Convert echo delay to samples
Ne = 5;

IR = zeros(1, Ne*Te_samples + 1); % Create IR vector
for i = 0:Ne-1
    IR(round(i*Te_samples) + 1) = a^i;
end

signal = conv(signal, IR, 'same'); % Convolve
signal = signal / max(abs(signal));

% New audio file
audiowrite('speechwithreverb.wav', signal, Fs);
```