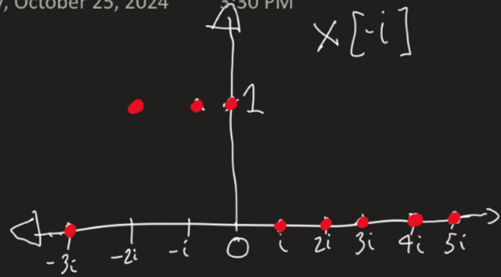
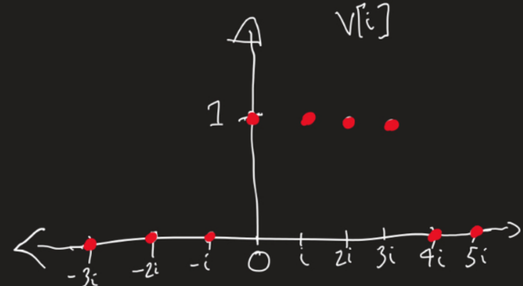
1. a) Manual computations of the convolutions of x[n] and v[n].

Friday, October 25, 2024 3:30 PM

a)





$y[i] = 0,\ i<0$

$y[0] = x[0] \cdot v[0] = 1 \cdot 1 = 1$

$y[i] = x[0] \cdot v[0] + x[i] \cdot v[i] = 2$

$\quad 1 \cdot 1 + 1 \cdot 1$

$y[2i] = x[0] \cdot v[0] + x[i] \cdot v[i] + x[2i] \cdot v[2i] = 3$

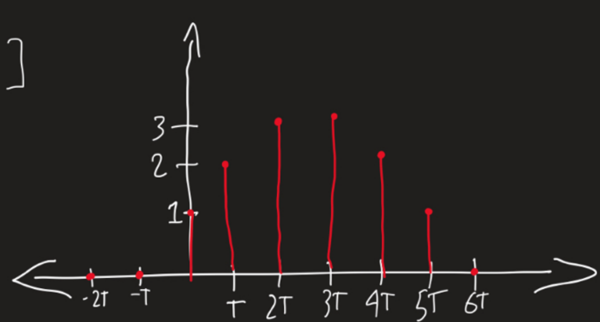
$y[3i] = x[i] \cdot v[i] + x[2i] \cdot v[2i] + x[3i] + v[3i] = 3$

$y[4i] = x[2i] \cdot v[2i] + x[3i] \cdot v[3i] = 2$

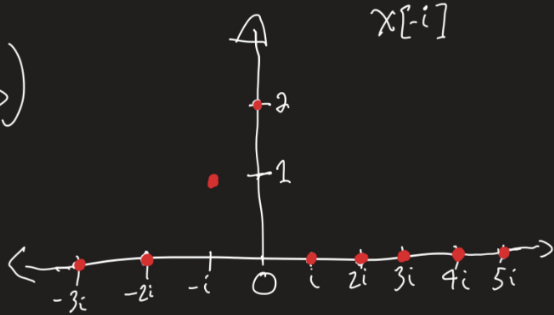
$y[5i] = x[3i] = 1$

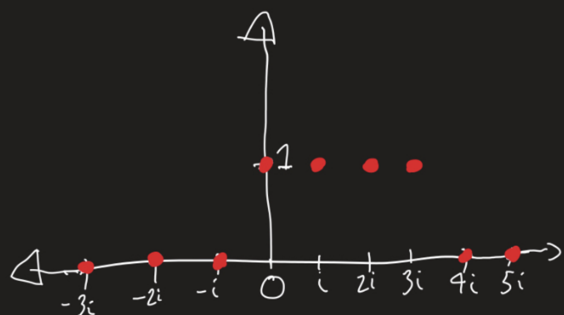
$y[i] = 0,\ i > 5i$

y[T]



b)





$y[i]=0,\ i<0$

$y[0]=x[0]\cdot v[0]=1\cdot 2=2$

$y[i]=\underset{1}{x[0]}\cdot \underset{1}{v[0]}+\underset{2}{x[i]}\cdot \underset{2}{v[i]}=3$

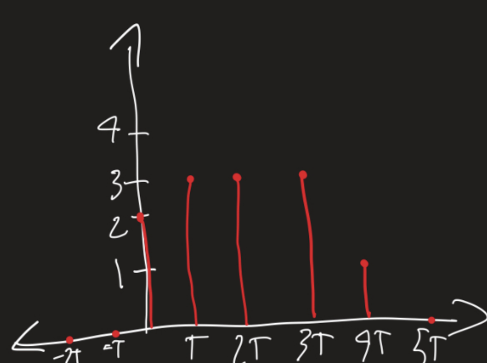
$y[2i]=x[i]\cdot v[i]+x[2i]\cdot v[2i]=3$

$y[3i]=x[2i]\cdot v[2i]+x[3i]\cdot v[3i]=3$

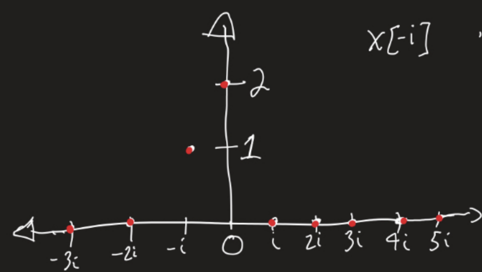
$y[4i]=x[3i]\cdot v[3i]+x[4i]\cdot \underset{0}{v[3i]}=1$

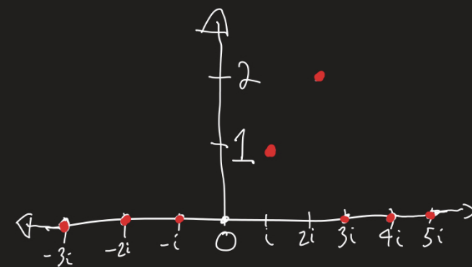
$y[i]=0,\ i>4i$

$y[T]$



c)





$y[i]=0,\ i<i$

$y[0]=x[0]\cdot \underset{0}{v[0]}=0$

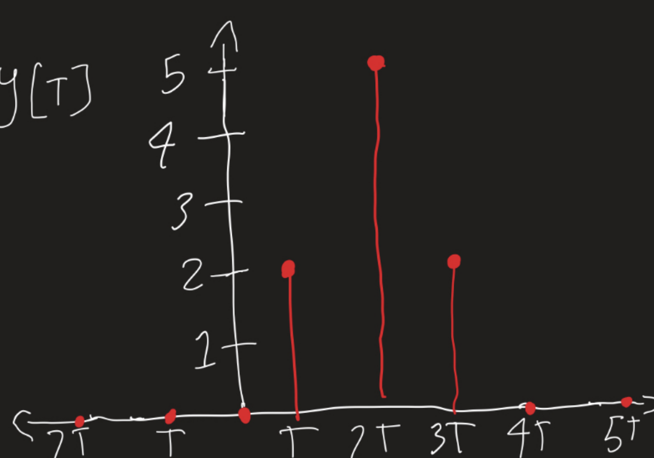
$y[i]=x[0]\cdot \underset{0}{v[0]}+x[i]\cdot v[i]=2$

$y[2i]=x[i]\cdot v[i]+x[2i]\cdot v[3i]=5$

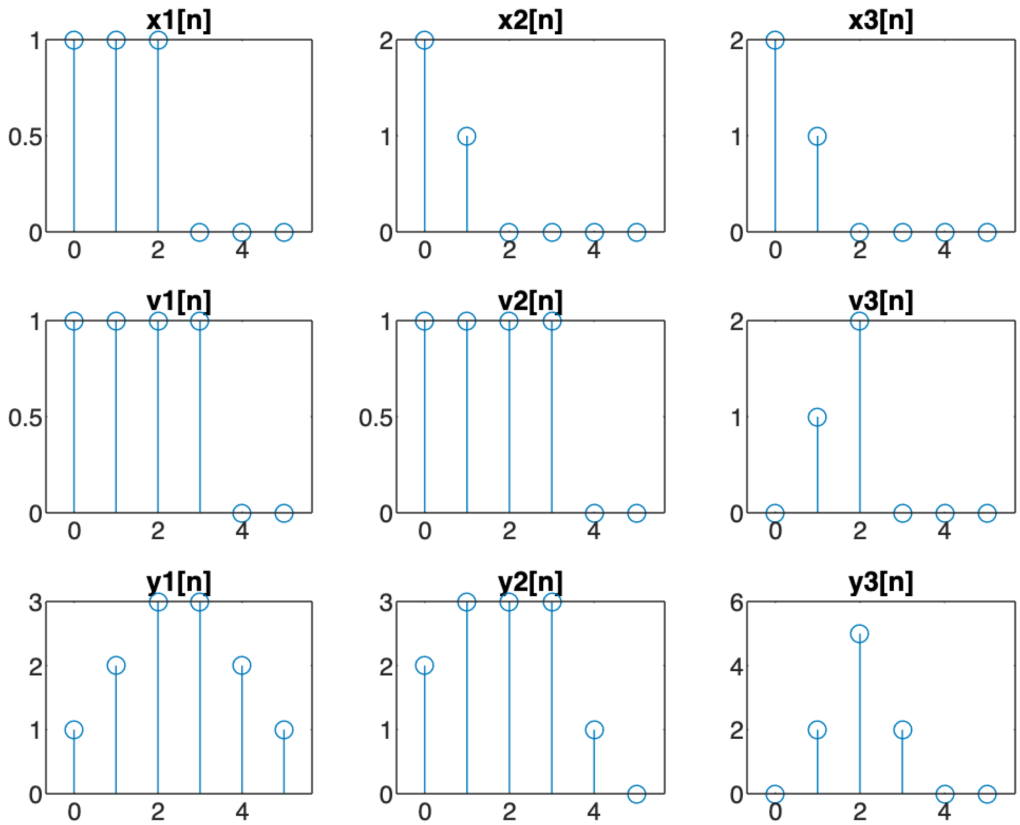
$y[3i]=x[2i]\cdot v[2i]=2$

$y[i]=0,\ i>3i$

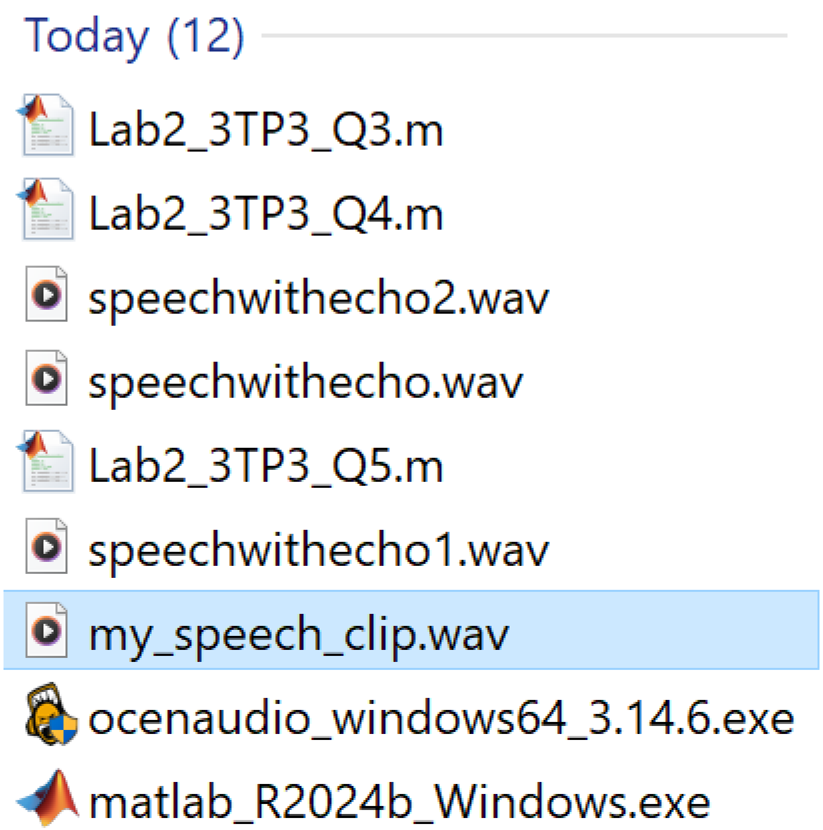
$y[T]$



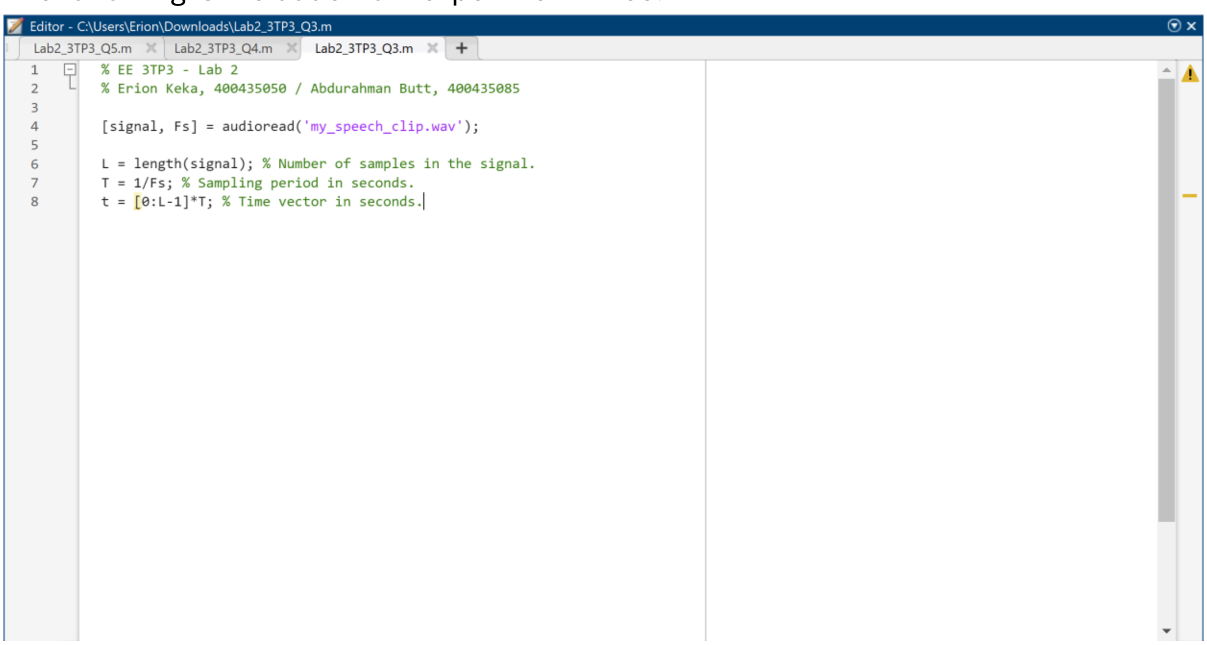
b) Graphs of x[n], v[n], and y[n]



2. The highlighted document is our team's audio clip.



3. The following is the code from experiment three.



4. The following is the code from experiment four.

Editor - C:\Users\Erion\Downloads\Lab2_3TP3_Q4.m

Lab2_3TP3_Q5.m | Lab2_3TP3_Q4.m

```
% EE 3TP3 - Lab 2
% Erion Keka, 400435050 / Abdurahman Butt, 400435085

[signal, Fs] = audioread('my_speech_clip.wav');

L = length(signal); % Number of samples in the signal.
T = 1/Fs; % Sampling period in seconds.
t = [0:L-1]*T; % Time vector in seconds.

Te = 500; % Defining Te -> The echo delay in msec.

alpha = 0.5; % Defining Alpha -> Amplitude Factor

sample = round((Te/1000)/T); % Converting to seconds then, determing samples.
echo = [zeros(sample,1);signal.*alpha]; % Creating the echo signal.
new_signal = [signal;zeros(sample, 1)]; % Matching the original signal length to that of the echo.
signalplusecho = new_signal + echo; % Adding the echo signal and the updated original signal.

% Rescaling and outputting the signal.
signalplusecho = signalplusecho/max(abs(signalplusecho));
audiowrite('speechwithecho2.wav', signalplusecho, Fs);

```

5. The following is the code from experiment five. Note: Alpha was set to 1 for Q6.

Editor - C:\Users\Erion\Downloads\Lab2_3TP3_Q5.m

Lab2_3TP3_Q5.m | Lab2_3TP3_Q4.m

```
% EE 3TP3 - Lab 2
% Erion Keka, 400435050 / Abdurahman Butt, 400435085

[signal, Fs] = audioread('my_speech_clip.wav');

L = length(signal); % Number of samples in the signal.
T = 1/Fs; % Sampling period in seconds.
t = [0:L-1]*T; % Time vector in seconds.

Te = 500; % Defining Te -> The echo delay in msec.

alpha = 1; % Defining Alpha -> Amplitude Factor

sample = round((Te/1000)/T); % Converting to seconds then, determing samples.
IR = [1;zeros(sample-1,1); alpha];


signalplusecho = conv(signal, IR); % Generating the output signal.

% Rescaling and outputting the signal.
signalplusecho = signalplusecho/max(abs(signalplusecho));
audiowrite('speechwithecho1.wav', signalplusecho, Fs);
```

An impulse response with a value of 1 was chosen initially with zeros filling the rest until the sample is reached with the echo delay. Upon reaching the sample, there is an alpha value which is the amplitude of the waveform. This convolution method allows for the original sample to come first then, the second sample with the applied amplitude shifted onwards.

6. Upon fixing the value of alpha to 1 and experimenting with a variety of values for Te, we noticed that when Te becomes approximately 18, the quality of audio becomes

acceptable. The audio becomes acceptable at a Te value of approximately 18 because this is the value in which the echo effect has been eliminated. The audio sounds acceptable as it is two waveforms placed on top of each other rather than one after the other. When the value of alpha is decreased, the value of Te does increase for which the audio can be deemed acceptable. This is a result of the value of alpha being the amplitude of the waveform, thus by decreasing the value of alpha, we are decreasing the volume of the echo which allows for more headspace when changing the value of Te.

7. The following is the code from experiment seven.

```
% EE 3TP3 - Lab 2
% Erion Keka, 400435050 / Abdurahman Butt, 400435085

% Load the original speech signal
[speech, fs] = audioread('my_speech_clip.wav');

Ne = 5;        % Number of echoes
alpha = 0.5;  % Decay factor for each echo
Te = 500;    % Echo delay interval in seconds (50 ms)

% Create impulse response for reverberation
impulse_response = zeros(1, round(Ne * Te * fs));  % Initialize with zeros
impulse_response(1) = 1;  % Original signal

for i = 1:Ne
    impulse_response(round(i * Te * fs) + 1) = alpha^i; % Place decayed echo at each delay
end

% Apply reverberation using convolution
reverberated_speech = conv(speech, impulse_response);

% Normalize to prevent clipping
reverberated_speech = reverberated_speech / max(abs(reverberated_speech));

% Save and play the reverberated speech
audiowrite('reverberated_speech.wav', reverberated_speech, fs);
sound(reverberated_speech, fs); % Play the reverberated signal
```

Upon repeating the step from experiment six with the reverberated speech, we notice that a Te value of approximately 8 was the base point to allow for an acceptable audio clip.