Ain Shams University Faculty of Engineering Discipline Programs



# Signals Project Report Computer Engineering and Software Systems (CESS)

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#### 1.0 Introduction

This project aims to display how the music notes DO, RE, MI, FA sound originally, low pass filtered, and high pass filtered.

## 2.0 Step 1

First off, we calculated the different frequencies of each musical node to be able to find sampling frequency using the equation in the figure below.

```
1
          n1=-9;
 2
          n2 = -7;
 3
          n3 = -5;
          n4=-4;
 4
          F1= 440*(2^(n1/12));
 5
          F2= 440*(2^(n2/12));
 6
          F3= 440*(2^(n3/12));
 7
          F4= 440*(2^(n4/12));
 8
          %QUESTION 1
 9
          Fs = 10* max([F1,F2,F3,F4]);
10
          framesize= round(0.5*Fs);
11
12
          T= 1/Fs;
          t=(0:1:framesize-1)*T;
13
          xt1=cos(2*pi*F1*t);
14
          xt2=cos(2*pi*F2*t);
15
          xt3=cos(2*pi*F3*t);
16
          xt4=cos(2*pi*F4*t);
17
```

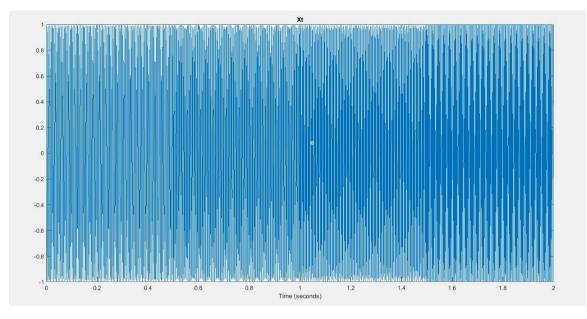
Then, we calculated x(t) by combining the x(t) of each musical node into one vector and generated its sound wave.

```
xt1=cos(2*pi*F1*t);
14
          xt2=cos(2*pi*F2*t);
15
          xt3=cos(2*pi*F3*t);
16
17
          xt4=cos(2*pi*F4*t);
          xt= [xt1 xt2 xt3 xt4];
18
          framesize= framesize *4;
19
          t=(0:1:framesize-1)*T;
20
          %QUESTION2
21
          filename= 'Xt.wav';
22
          audiowrite('Xt.wav',xt, fix(Fs));
23
          sound(xt,Fs);
24
```

#### 4.0 Step 3

Then, we plotted signal x(t) versus time (t) as shown in the figure below

```
%QUESTION3: Plot the Xt:
figure;
plot(t,xt);
xlabel('Time (seconds)');
title('Xt');
```



Then, we computed the energy of the signal x(t) using the equation below.

```
30 %QUESTION4
31 XEnergy= sum(abs(xt.^2))*T;
```

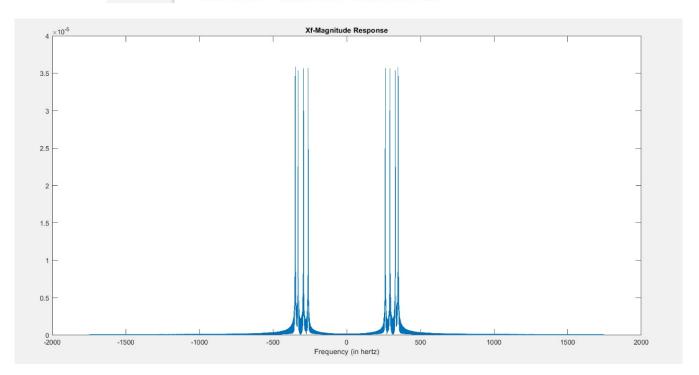
## 6.0 Step 5

Then, we computed the frequency spectrum X(f) of the signal using the equations below.

```
32 %QUESTION5
33 Xf= fftshift(fft(xt,framesize))*T;
34 dF = Fs/framesize;
35 f = -Fs/2:dF:Fs/2-dF;
```

## 7.0 Step 6

Then we plotted the magnitude of X(f) in the frequency range fs/2 <= f <= fs/2, where fs is the sampling frequency as shown in the figure below.



Then we computed the Energy of the signal x(t) from its frequency spectrum X(f), and verified Parseval's Theorem using the equation below.

```
41 %QUESTION7
42 XEnergy2=sum(abs(Xf.^2))*dF;
```

## 9.0 Step 8

Then we designed a Butterworth low-pass filter with filter order 20 such that when the signal x(t) is applied to this filter, the output does NOT contain the MI and FA musical nodes. Moreover, we calculated the cut-off frequency by calculating the average of RE and MI musical nodes together.

```
43 %QUESTION8

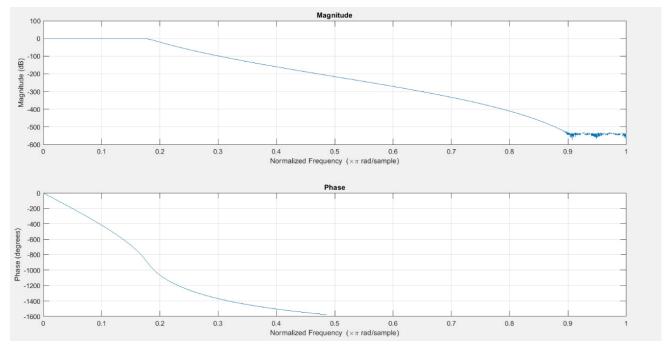
44 Fc= (F2+F3)/2;

45 nF= Fc/(Fs/2);

46 [b,a] = butter(20,nF,"low");
```

#### 10.0 Step 9

Then we plotted the magnitude and phase response of the Butterworth low pass filter as shown in the figure below.



Then, we applied the signal x(t) to this Butterworth LPF and denoted the output signal as y1(t)

```
50 %QUESTION10
51 Y1t= filter(b,a,xt);
```

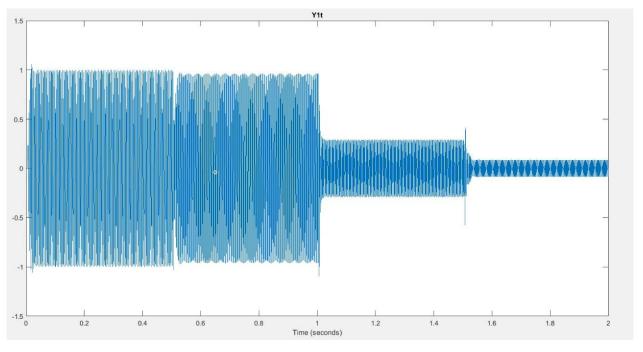
# 12.0 Step 11

Then we stored the generated signal y1(t) as an audio file with extension (\*.wav)

```
%QUESTION11
filename= 'Yt.wav';
audiowrite('Yt.wav',Y1t, fix(Fs));
sound(Y1t,Fs);
```

# 13.0 Step 12

Then we plotted the signal y1(t) versus time (t) as shown in the figure below.



Then we computed the energy of the signal y1(t) using the equation below.

```
61 %QUESTION13
62 Y1Energy= sum(abs(Y1t.^2))*T;
```

## 15.0 Step 14

Then we computed the frequency spectrum y1(f) of this signal.

```
63 %QUESTION14
64 Y1f= fftshift(fft(Y1t,framesize))*T;
```

#### 16.0 Step 15

Then we plotted the magnitude of y1(f) in the frequency range fs/2 <= f <= fs/2 as shown in the figure below

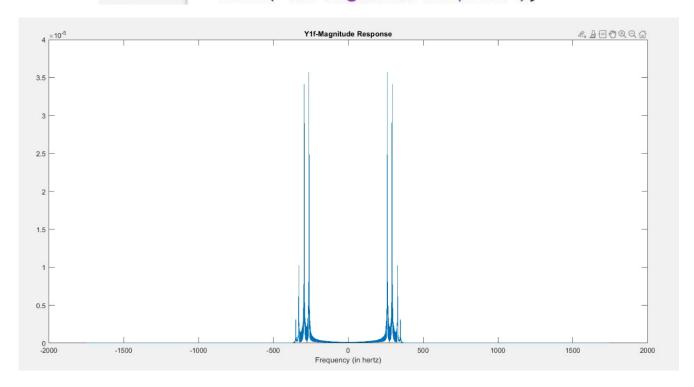
```
65 %QUESTION15

66 figure;

67 plot(f,abs(Y1f)/framesize);

68 xlabel('Frequency (in hertz)');

69 title('Y1f-Magnitude Response');
```



Then we computed the Energy of the signal y1(t) from its frequency spectrum y1(f) and verified Parseval's Theorem using the equation below.

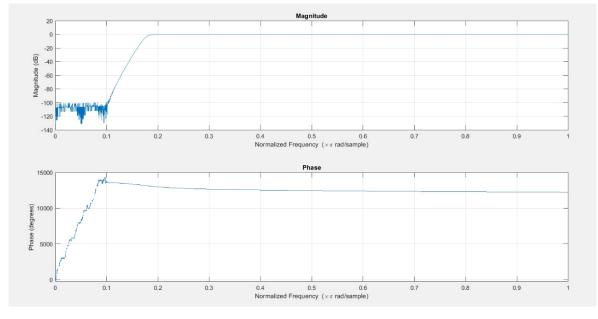
#### 18.0 Step 17

Then we designed a Butterworth high-pass filter with filter order 20 such that when the signal x(t) is applied to this filter, the output does NOT contain the DO and RE musical nodes as shown in the figure below. Moreover, we calculated the cut-off frequency by calculating the average of RE and MI musical nodes together.

#### 19.0 Step 18

Then we plotted the magnitude and phase response of the Butterworth HPF as shown in the figure below.

```
74 %QUESTION18
75 figure;
76 freqz(d, c, framesize);
```



Then we applied the signal x(t) to this Butterworth HPF and denoted the output as y2(t).

```
77 %QUESTION19
78 Y2t= filter(d,c,xt);
```

# 21.0 Step 20

Then we stored the generated signal y2(t) as an audio file with extension (\*.wav).

```
79 %QUESTION20

80 filename= 'Y2t.wav';

81 audiowrite('Y2t.wav',Y2t, fix(Fs));

82 sound(Y2t,Fs);
```

# 22.0 Step 21

Then we plot the signal y2(t) versus time (t) as shown in the figure below.

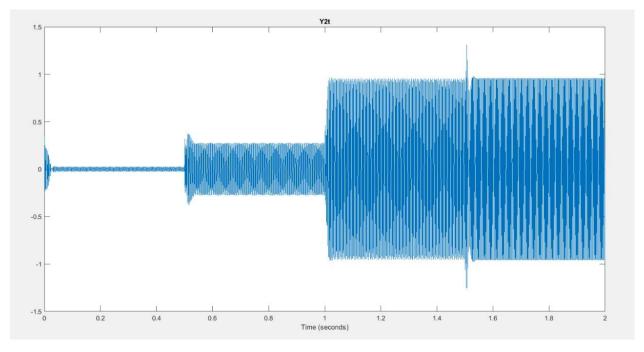
```
83 %QUESTION21

84 figure;

85 plot(t,Y2t);

86 xlabel('Time (seconds)');

87 title('Y2t');
```



Then we computed the energy of the signal y2(t) using the equation below.

88	%QUESTION22
89	Y2Energy= sum(abs(Y2t.^2))*T;

# 24.0 Step 23

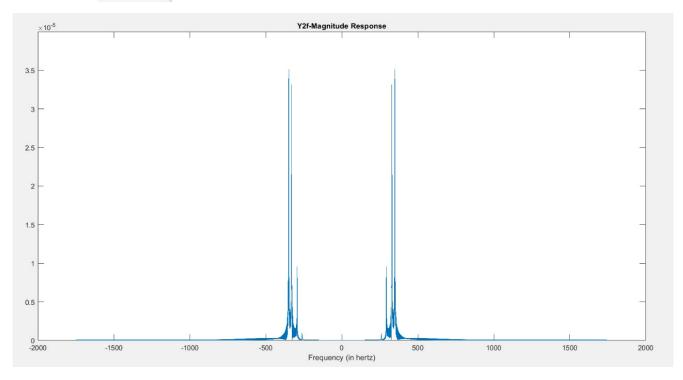
Then we computed the frequency spectrum Y2(f) of this signal.

```
90 %QUESTION23
91 Y2f= fftshift(fft(Y2t,framesize))*T;
```

# 25.0 Step 24

Then we plotted the magnitude of y2(f) in the frequency range fs/2 <= f <= fs/2 as shown in the figure below.

```
92 %QUESTION24
93 figure;
94 plot(f,abs(Y2f)/framesize);
95 xlabel('Frequency (in hertz)');
96 title('Y2f-Magnitude Response');
```



Then we computed the Energy of the signal y2(t) from its frequency spectrum y2(f) and verified Parseval's Theorem using the equation below.

97	%QUESTION25
98	Y2Energy2=sum(abs(Y2f.^2))*dF;

# **27.0 Contributions**

	Habiba	Nadine	Salma	Hamsa	Mahamad
Responsibility: Indicate the specific part of	2,8,9,10 17,18,19	4,5,7,13 16,22,25	1,3,6,12 15,21,24	Shared in the rest	0
preparing the report that each individual was responsible for.					
Cooperation: (10 points) Able to work within team. Willingly performed tasks.	10	10	10	10	0
Punctuality (5 points) On time for team meetings.	5	5	5	Medical Excuse	0
Evaluative (10 points) Offer constructive criticism and helpful evaluation of work.	10	10	10	Medical Excuse	0
	Total: 25	Total:25	Total:25	Total:10 Medical Excuse (Tried to help as best as possible)	Total: 0

