```
diary on
format compact
%Johnny Li
%EEL3135 Fall 2018
%Lab 1 Part 2
type sinusoid
%SINUSOIDS Function created for lab1.2.
%Script based on given instruction 2.1.
%Time vector t that is two cycles of the 5000 Hz sinusoid defined.
T=1/5000;
%The period.
tt = -T:1/100000:T;
%Time vector.
%2.1.2
%Constant values.
A1=21; %A1 is my age.
A2=1.2*A1; %A2 values are define in the lab.
          %M is my birth month.
M=9;
          %D is my birthday.
D=11;
%Phase constant.
tm1=(37.2/M)*T;
                  %tml values are define in the lab.
tm2=(-41.3/D)*T; %tm2 values are define in the lab.
%Generate two 5000 Hz sinusoids: (equations are defined in the lab)
x1=A1*cos(2*pi*(5000)*(tt-tm1));
x2=A2*cos(2*pi*(5000)*(tt-tm2));
%2.1.3
%Create a third sinusoid that is the sum:
x3=x1+x2;
%2.1.4
%Plot all three sinusoids.
%x1 plot
subplot(3,1,1)
                  %Define axis.
plot(tt,x1)
                  %Period vs sinusoid.
title('Johnny, the Author of the lab (x1)')
                                                %My name.
%x2 plot, repeat process.
subplot(3,1,2)
plot(tt,x2)
title('Amumu, the Sad Mummy (x2)')
%x3 plot, repeat process.
subplot(3,1,3)
plot(tt,x3)
title('Fiora, the Grand Duelist (x3)')
```

sinusoid

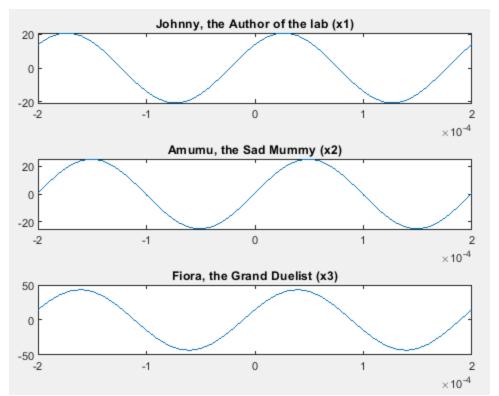


Figure 1: 2.1.4

Plot defined sinusoids.

```
type one cos
function [x,tt] = one_cos(A,W,P,D)
%one_cos Function for lab1.2.
%The function creates a sinusoid based on given instruction 2.1.5.
%Uses four inputs: amplitude (A), frequency (W), phase (P), duration (D)
%and two outputs: the values of the signal (x) and the corresponding times
%The function should generate exactly 25 values of the sinusoid per period.
%2.1.5
%Find the period from frequency given.
T=(2*pi)/W;
tt = 0:T/25:D;
%Function
x=A*cos(W*tt+P);
%Taken from 1.3 of part 1
plot(tt,x)
grid on
title('PLOT OF A SINUSOID')
xlabel('TIME (sec)')
end
```

one cos(95,200*pi,5,0.025)

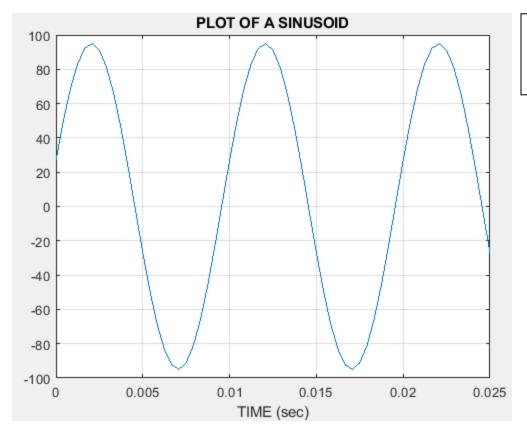


Figure 2: 2.1.5

Plot sinusoid from given values.

 $\$ Question: What is the expected period in milliseconds? $\$ The expected period is T=(2*pi)/W, where W is 200pi rad/s, therefore $\$ T=10ms.

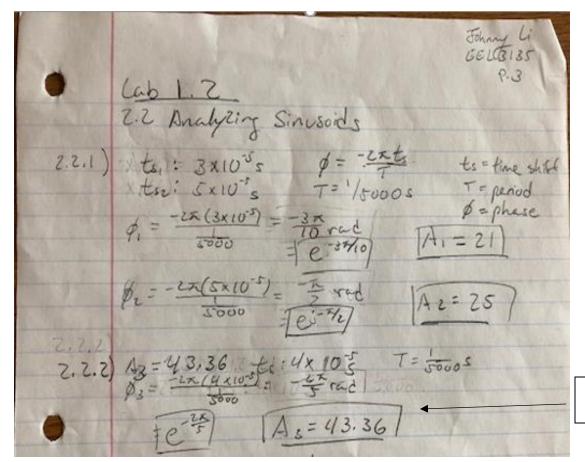
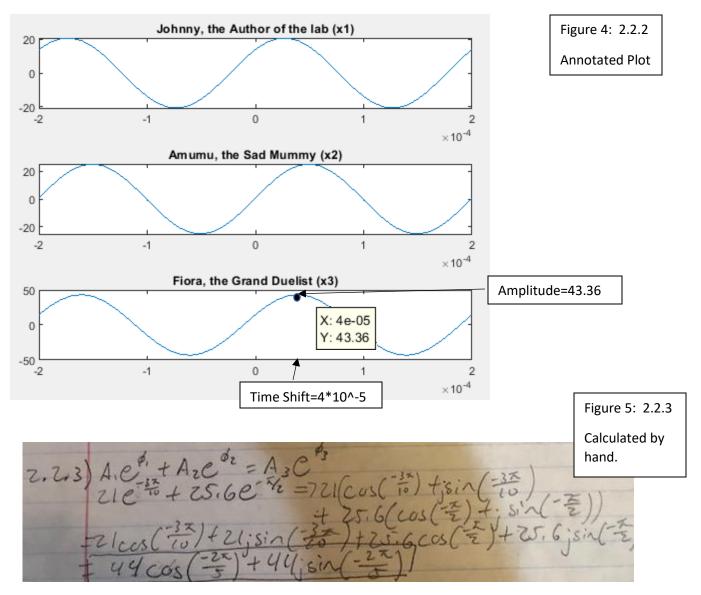


Figure 3: 2.2.1 Calculated by hand.

Phasor3 calculated



%Phase percent error is calculated with the |(e-t)/t|*100, where e is the %experimental result and t is the theoretical result. Therefore, %e=-2pi/5 and t=43.36cos-2pi/5, the result is 0%. Amplitude percent error is %calculated to be 1.36%, where e=43.36 and t=44. These error is %most likely the outcome of rounding on the calculations done by hand.

```
%2.3 %Time vector t that is two cycles of the 5000 Hz sinusoid defined. Setup. %The period. tt = -T:1/100000:T; A1=21; %A1 is my age. M=9; %M is my birth month. %Phase constant. tm1=(37.2/M)*T; %tm1 values are define in the lab.
```

%x1=A1*cos(2*pi*(5000)*(tt-tm1)); original function.

%Code that will generate x1 by using the complex-amplitude representation. x1 = real(A1*exp(j*2*pi*5000*(tt-tm1))); plot(tt,x1)

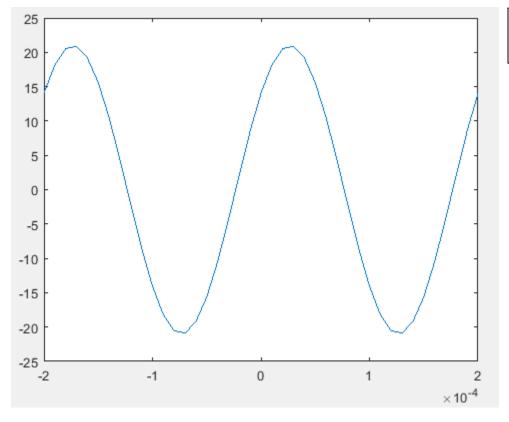


Figure 6: 2.3 Plot of x1

\$2.4.1 \$Using fs=11025 samples/sec, instantiate a time vector of 0.5 second <math>\$duration. Vector x1 of samples of a sinusoid with amplitude A=100, \$angular frequency w=2pi(800), and phase = -pi/3.

```
%Frequency sample
fs=11025;
%Amplitude
A=100;
%Duration
D=0.5;
%Angular frequency
W=2*pi*(800);
%Period
T=(2*pi)/W;
%Time
tt = 0:1/fs:D;
%Phase
P=-pi/3;
%Function
x1=A*cos(W*tt+P);
%Create autofile
audiowrite('S2 4 1.wav',x1,fs);
```

```
S2 4 1
type S2 4 2
%The input of the sound signal/sinusoid created for lab1.2.
%Script based on given instruction 2.4.2.
%2.4.2.
%Using fs=11025 samples/sec, instantiate a time vector of 0.8 second
%duration. Vector x2 of samples of a sinusoid with amplitude A=80,
%angular frequency W=2pi(1200), and phase = pi/4.
%Frequency sample
fs=2*11025;
%Amplitude
A = 80;
%Duration
D=0.8;
%Angular frequency
W=2*pi*(1200);
%Period
T=(2*pi)/W;
%Time
tt = 0:1/fs:D;
%Phase
P=pi/4;
%Function
x2=A*cos(W*tt+P);
%Create autofile
audiowrite('S2 4 2.wav',x2,fs);
S2 4 2
*Question: How does this sound compare to the output of the previous sound?
%The x2 sound has a higher pitch and longer duration than the x1 sound,
%but x1 sound is louder that the x2 sound.
type S2 4 3
%The input of the sound signal/sinusoid created for lab1.2.
%Script based on given instruction 2.4.3.
%Concatenate x1 and x2 into a vector x, leaving a duration of 0.1 seconds
%of silence in between.
%x1 signal
%Frequency sample
fs=11025;
%Amplitude
A1=100;
%Duration
D1=0.5;
```

```
%Angular frequency
W1=2*pi*(800);
%Period
T1=(2*pi)/W1;
%Time
tt1 = 0:1/fs:D1;
%Phase
P1=-pi/3;
%Function
x1=A1*cos(W1*tt1+P1);
%x2 signal
%Amplitude
A2 = 80;
%Duration
D2=0.8;
%Angular frequency
W2=2*pi*(1200);
%Period
T2=(2*pi)/W2;
%Time
tt2 = 0:1/fs:D2;
%Phase
P2=pi/4;
%Function
x2=A*cos(W2*tt2+P2);
%Silence for 0.1 sec
%fs/10=11025/10=1102.5
N=1103;
sil=zeros(1,N);
%Concatenation
x = [x1, si1, x2];
%Create autofile
audiowrite('S2_4_3.wav',x,fs);
%2.4.4
%Plot and identify the two input signals.
vt = (1/11025)*(1:length(x));
plot(vt,x)
```

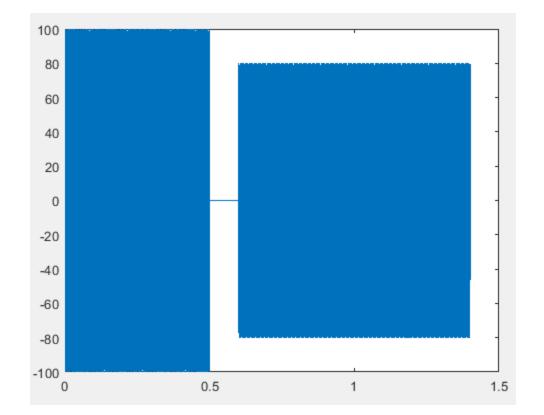


Figure 7: 2.4.4 Plot of x

%Plot and identify the two input signals.

 $\mbox{\ensuremath{\$}}\mbox{\ensuremath{Verify}}$ that the amplitude changes from 100 to 0, and then from 0 to 80 at $\mbox{\ensuremath{\$}}\mbox{\ensuremath{the}}$ to recet times.

%The first signal (x1) has a amplitude of 100 from time 0 to 0.5s which is %what the stated function calls for. There is a 0.1s of silence, the break %in between the signals. The second signal (x2) has an amplitude of 80 %starting from 0.6 to 1.4s which is what the stated function calls for.

%2.4.5 %Different sampling frequencies of fs=22050 samples per sec. fs2=22050;

audiowrite('S2 $_4$ 5.wav',x,fs2);

%The higher the frequency, sampling was 2xfs, means more frequencies in x. %The result is a higher pitch of the signals but has a shorter duration, %nearly half.

S2 4 3

%{Concatenate the sounds together (a musical scale) created for lab1.2. %Script based on given instruction 2.4.6.

%2.4.6 %Sampling frequency fs = 11025;

%Given scales frequency scale = [523, 587, 659, 698, 784, 880, 988, 1047];

```
%t from 1.4
tt = 0:1/fs:0.5;
%Concatenate scales, leaving a duration of 0.1 seconds of silence in
%between.
%Silence for 0.1 sec
%fs/10=11025/10=1102.5
N=1103;
sil=zeros(1,N);
%Initializes c vector.
c=[];
%Loop through all scale
for n=1:8
    %Signal function
    y=1.4*exp(j*pi/2)*exp(j*2*pi*scale(n)*tt);
    %Concatenate scales and 0.1s silence in between.
    c=[c,sil,y];
end
%Create autofile
audiowrite('S2_4_6.wav',c,fs);
```