SIGNALS AND SYSTEMS
LAB: 12

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TASK :01

Task : 01 (Lab #12) Date

$x(t) = \sin(2\pi f_0 t)$

1) $T_s = 0.25s$

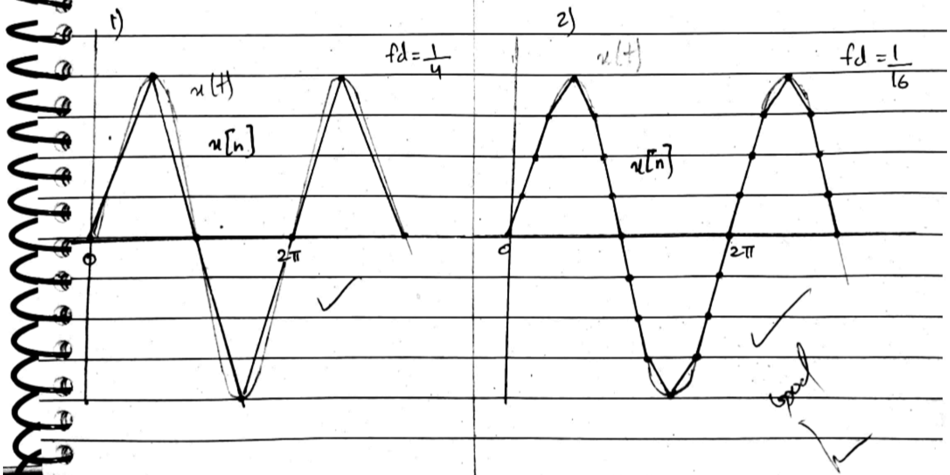
$f_s = \frac{1}{T_s} = \frac{1}{0.25} = 4Hz$ $f_0 = 1\ Hz$

$f_d = \frac{f_0}{f_s} = \boxed{\frac{1}{4}}$ → Means there will be four samples per cycle.

2)

$f_d = \frac{f_0}{f_s} \rightarrow \frac{1}{16} = \frac{1}{f_s}$ $\because f_d = \frac{1}{16}$

So, $\boxed{f_s = 16\ Hz}$ $f_d = \frac{1}{16}$ means there will be 16 samples per cycle.



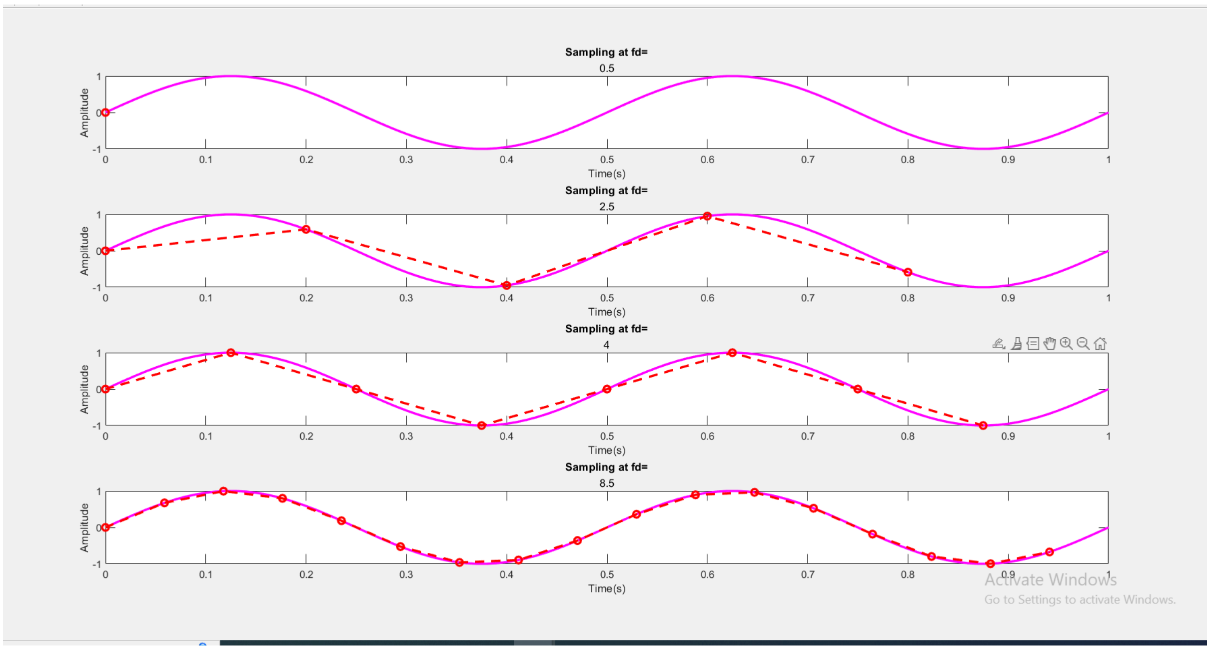
MAAZ

TASK :02

CODE:

```
clc
clear all
close all
f0=2; %fundamental frequency of the signal
dur=1; %signal duration
T0=1/f0; %time period of signal
stepsize=0.01;
t=0:stepsize:dur; %time vector
xx=sin(2*pi*f0*t);
samplingfreq=  [1 5 8 17]; %%different sampling frequencies to be used
for i = [1 2 3 4]
   fs = samplingfreq(i);
   Ts=1/fs; %samping time
   [x,tt]=sampled(f0,fs,dur);
   subplot(4,1,i)
   plot(t,xx,'m','Linewidth',2) %plotting the original signal
   hold on
   plot(tt,x,'r--o','Linewidth',2) %plotting the sampled signal
   title('Sampling at fd=',fs/f0);
   xlabel('Time(s)')
   ylabel('Amplitude')
end
el('Amplitude')
end
```

SNAPSHOT:



TASK :03

FILE SIZE:
Fs= 1000 → File size = 10 KB
Fs= 2000 → File size = 20 KB
Fs= 4000 → File size = 40 KB
Fs= 16000 → File size = 157 KB
Fs= 44100 → File size = 431 KB

OBSERVATIONS:

- By increasing the sampling frequency the audio gets clearer. This is obvious because the less samples we take the more informationof the audio we lose. So the recordings with lower frequency are much less accurate and it is harder to understand what is being spoken. Whereas the audio with larger sampling frequencies are very clear. This is because the we keep getting closer to the original spoken audio as we increase th sampling frequency.
- By increasing the sampling frequency the file size is also increasing. This is because increasing the sampling frequency increases the number of samples taken, and to store more samples we need more space. So greater frequency audios will be greater in size and accuracy.

Code will be the same as the one provided in lab manual with different value of Fs.

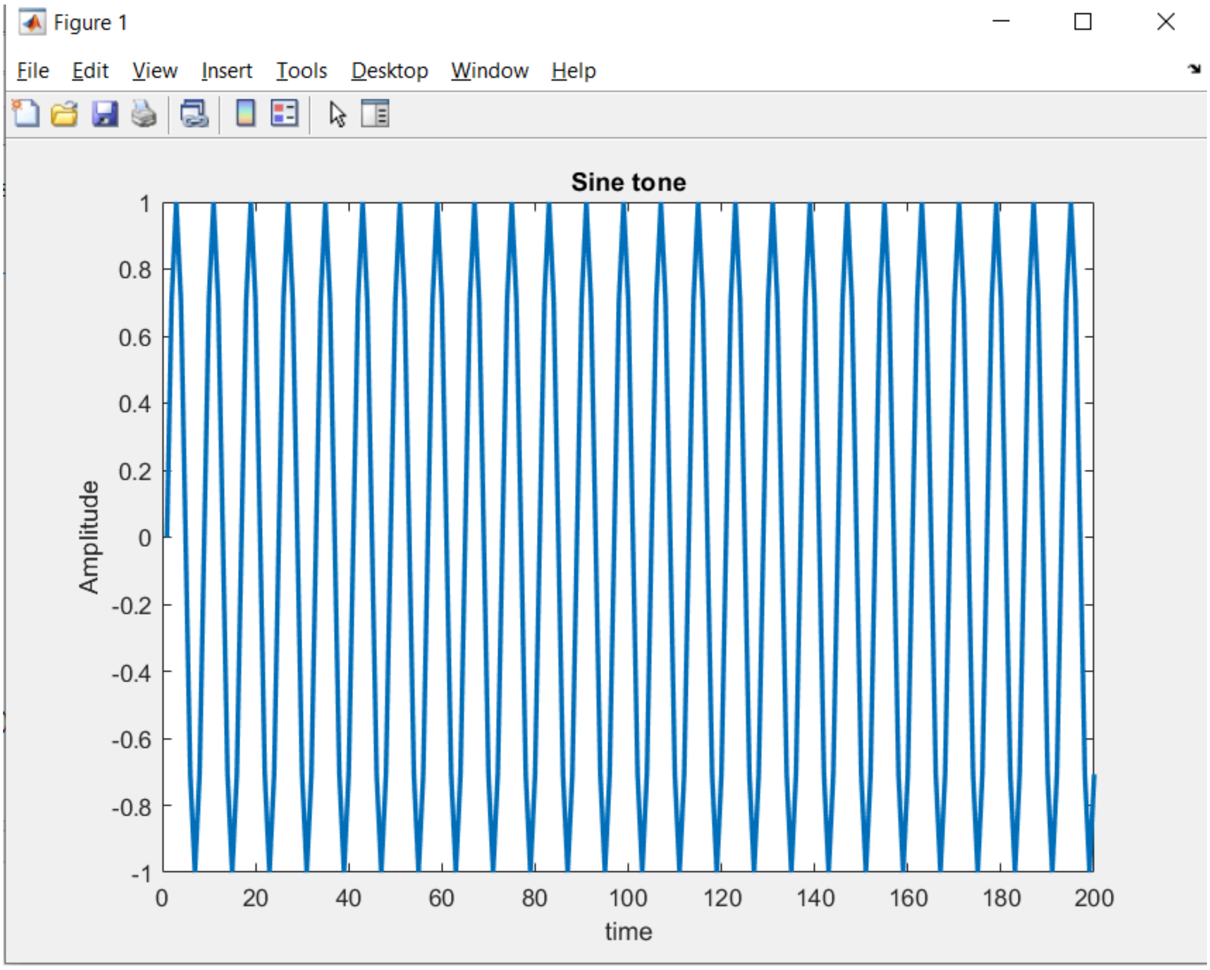
TASK :04

1)

CODE:

```
clc
clear all
close all
f0 = 1000;
fs = 8000;
t=0:1/fs:3;
xt= sin(2*pi*f0*t); %sin wave
soundsc(xt)
figure
plot(xt(1:200), 'LineWidth',2 ); %plotting first 200 samples
title("Original sine tone");
xlabel("time");
ylabel("Amplitude");
```

WAVE:

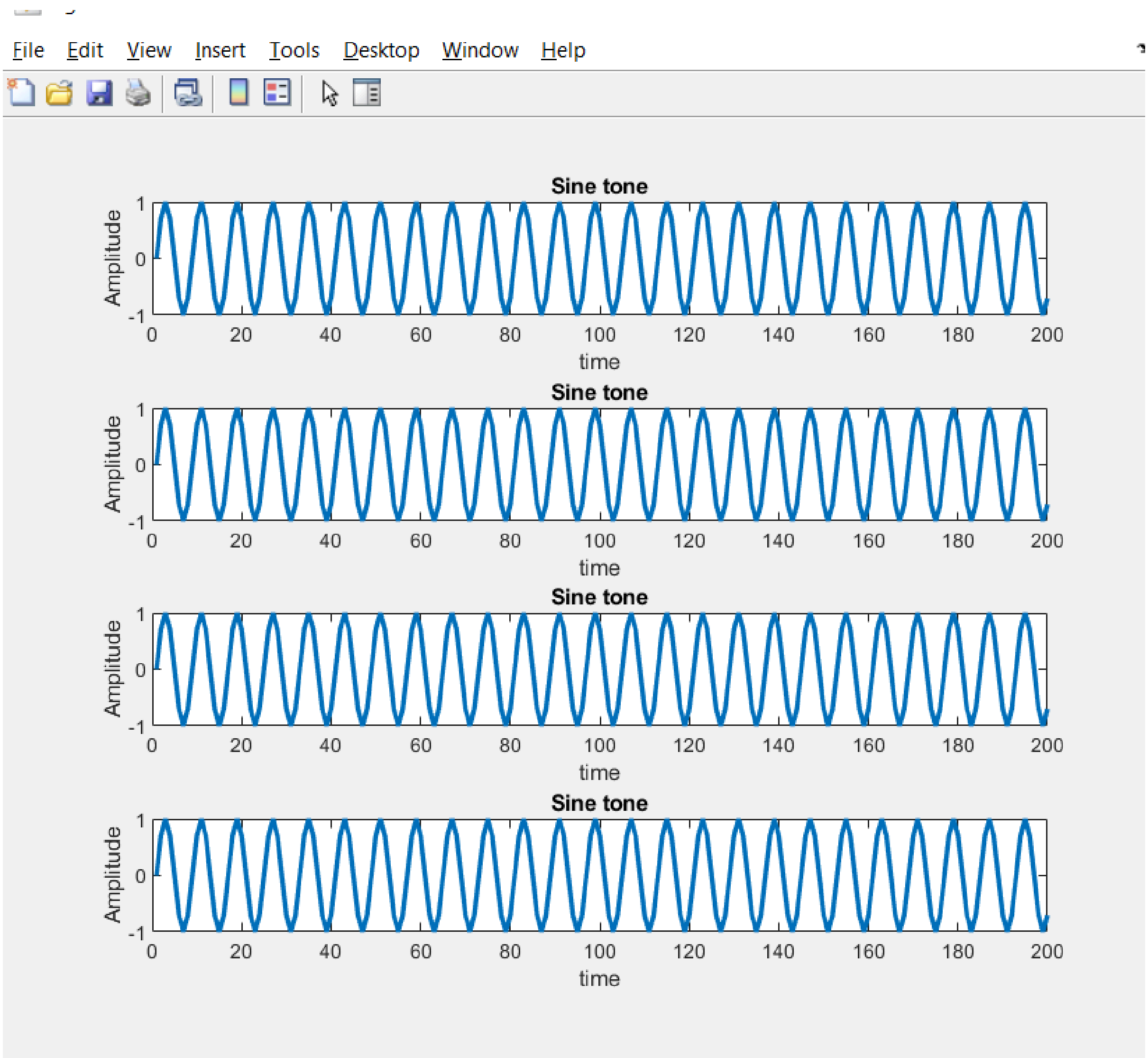


2)

CODE:

```
clc
clear all
close all
toneFreq = [1000 9000 17000 25000];
fs = 8000;
t=0:1/fs:3;
figure
for i = [1 2 3 4]
   f0 = toneFreq(i);
   xt= sin(2*pi*f0*t); %sin wave
   subplot(4,1,i)
   plot(xt(1:200), 'LineWidth',2 ); %plotting first 200 samples
   title("Sine tone");
   xlabel("time");
   ylabel("Amplitude");
End
```

SNAPSHOT:



OBSERVATIONS:
We are able to hear the 25kHz frequency because it is not 25Khz anymore. Its because the condition fs>2f0 is not satisfied and the sampled signal overlaps, which causes loss of information.

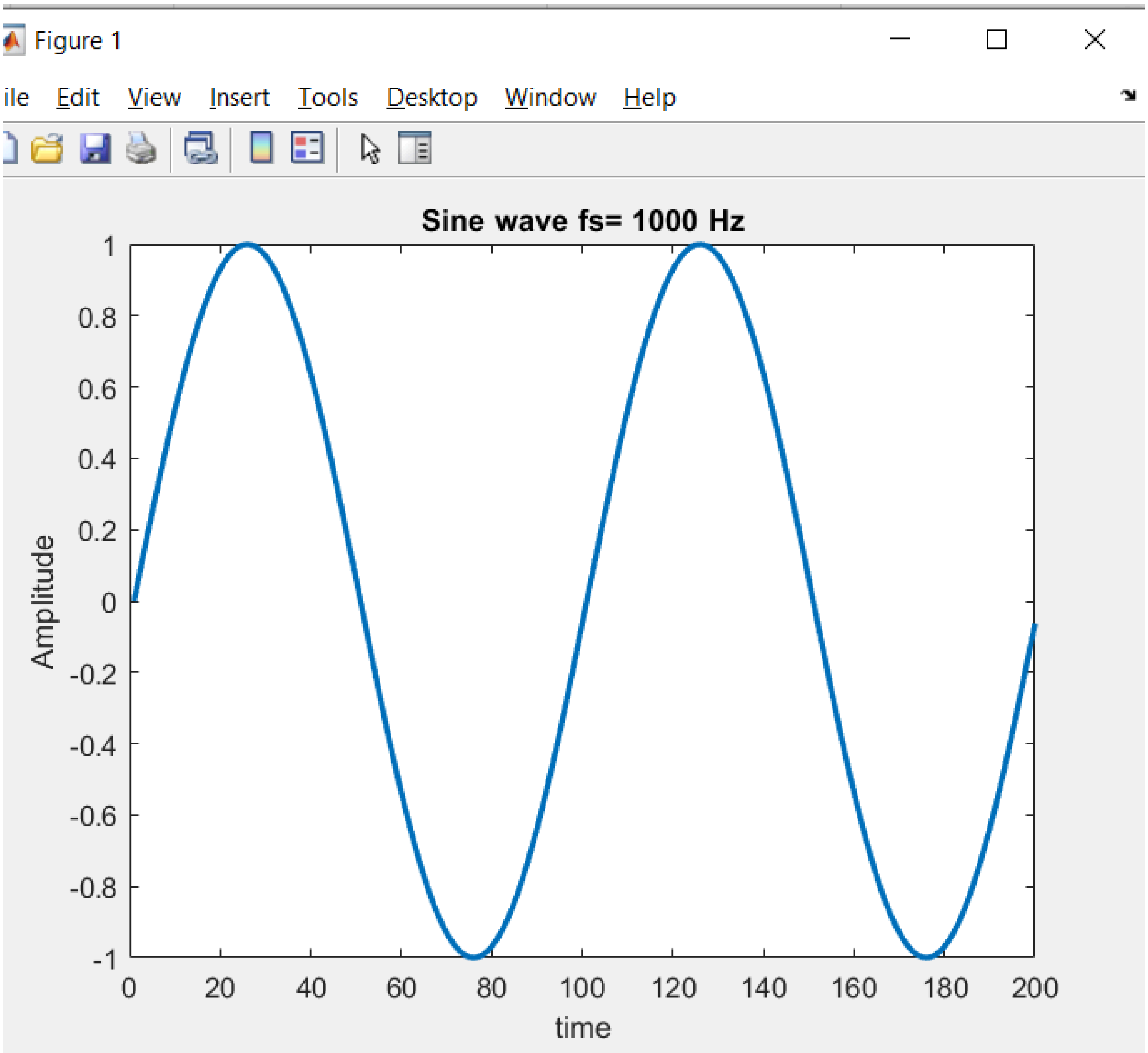
TASK :05

1)

CODE:

```
clc
clear all
close all
f0 = 10;
fs = 1000;
t=0:1/fs:3;
xt= sin(2*pi*f0*t); %sin wave
figure
plot(xt(1:200), 'LineWidth',2 ); %plotting first 200 samples
title("Sine wave fs= 1000 Hz");
xlabel("time");
ylabel("Amplitude");
```

PLOT:



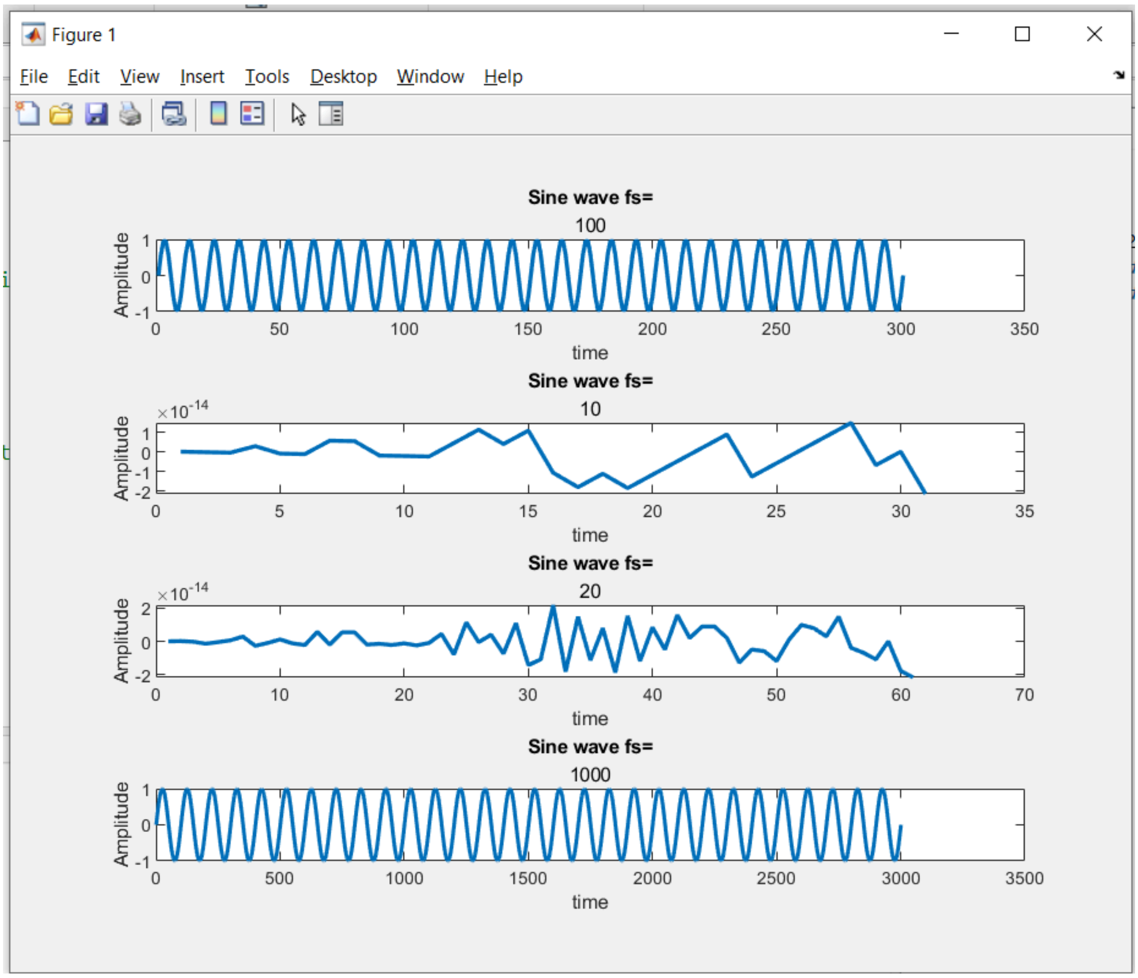
2)

CODE:

```
clc
```

```
clear all
close all
f0 = 10;
figure
samplingfreq = [100 10 20 1000]; %original signal, fs<2f0,fs=2f0 and fs>2fo
for i = [1 2 3 4]
    fs = samplingfreq(i);
    t=0:1/fs:3;
    xt= sin(2*pi*f0*t); %sin wave
    subplot(4,1,i)
    plot(xt, 'LineWidth',2 ); %plotting first 200 samples
    title("Sine wave fs=" , fs);
    xlabel("time");
    ylabel("Amplitude");
end
```

PLOT:



- fs=10<2f0
- fs=20=2f0
- fs=1000>2f0