Experiment 2

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1 Aim of the experiment

To familiarize with the concept of time-domain sampling and alias.

2 Results/Graphs

2.1 Question 1

Plot the following waveforms for $0 \leq t \leq 1$
(a) xa1(t)=2cos(10πt)
(b) xa2(t)=cos(20πt).
(c) xa3(t)=0.5cos(60πt) .

Code

```
xa1 = inline('2*cos(10*pi*t)','t');
xa2 = inline('cos(20*pi*t)','t');
xa3 = inline('0.5*cos(60*pi*t)','t');

t = [0:0.001:1];

figure('Name','Task 1')

plot(t,xa1(t),'b',t,xa2(t),'r',t,xa3(t),'k');
xlabel('t');
title('xa1(t),xa2(t) and xa3(t) vs t');
legend('xa1','xa2','xa3');
saveas(gcf,'output/task1.png')
```

Graph

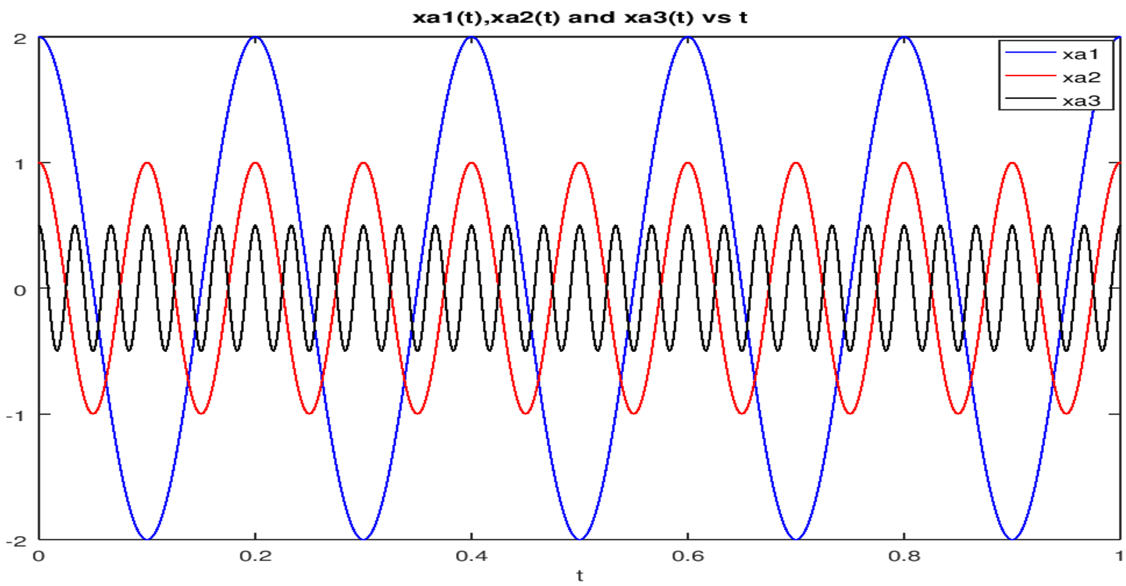

Figure 1: For Question 1

i) Code

```
% task i
xa = inline('xa1(t)+xa2(t)+xa3(t)','t');
figure
plot(t,xa(t));
xlabel('t');
ylabel('xa(t)');
title('xa(t) vs t');
saveas(gcf,'output/task1i.png')
```

Graph

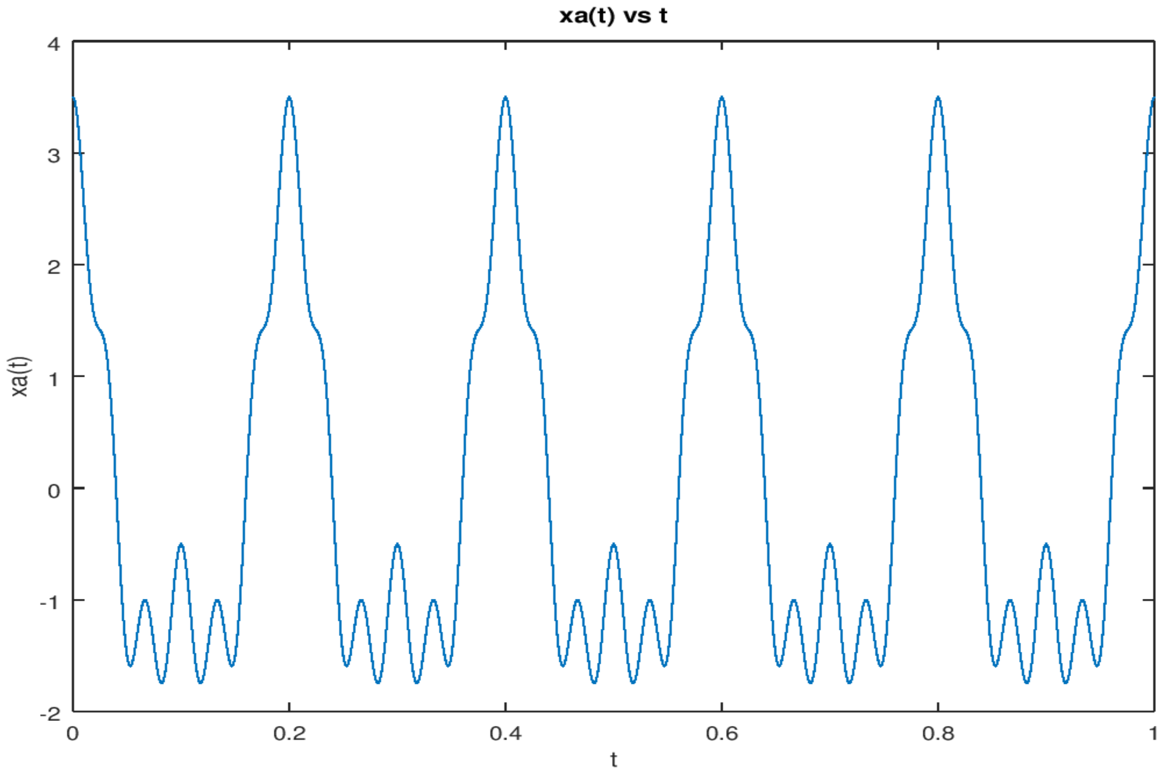

Figure 2: For Question 1 (i)

Observations

- The three cosines have frequencies of 5, 10 and 30 Hz respectively.
- They also have time periods of 0.2, 0.1 and 0.033 seconds respectively.
- The sum of the three waveforms is not a sine/cosine wave.

Conclusion

The Periodicity of the xa is the LCM of the periodicity of the three functions that form it, 0.2s and the nyquist frequency is the maximum frequency of the signals xa1,xa and xa3 times 2 = 60 Hz.

ii) Code

```
%task ii a)
figure
subplot(3,1,1);
Fs= 50;
t1=[0:1/Fs:1];
stem(t1,xa(t1));
title('Fs = 50');
xlabel('t');
ylabel('xa(t)');

%task ii b)
subplot(3,1,2);
% LCM = 60 pi
Fs=60;
t2=[0:1/Fs:1];
stem(t2,xa(t2));
title('Fs = Nyquist rate');
xlabel('t');
ylabel('xa(t)');

%task ii c)
subplot(3,1,3);
t3=[0:1/(5*Fs):1];
stem(t3,xa(t3));
title('Fs = 5 times Nyquist rate');
xlabel('t');
ylabel('xa(t)');
```

Graph

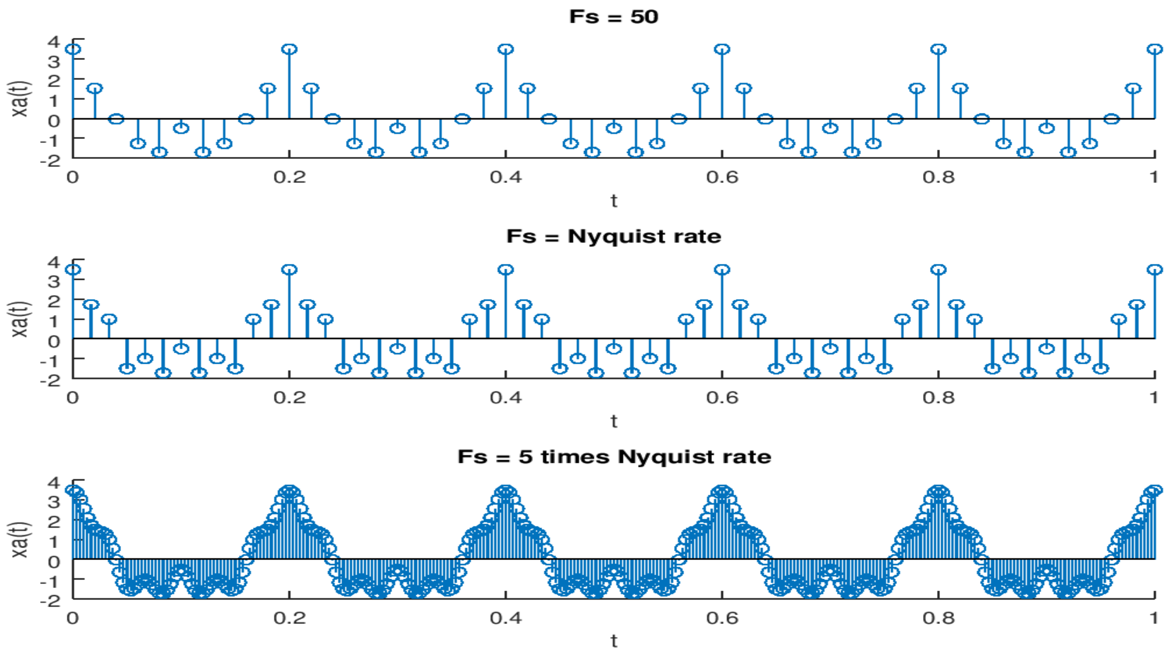

Figure 3: For Question 1 (ii)

Observations

- The one sampled with 300 samples/cycle resembles the original continuous time function the most.
- The signal shown with Fs=50 has missing peaks which is not lost at Nyquist rate sampling or higher sampling rate.
- Aliasing is noticed in Fs=50 sampled signal.

Conclusion

As we go on increasing the sampling rate, the stem of the sampled signal resembles the original signal more and more.

iii) Code

```
% task iii a)
figure
subplot(3,1,1);
Fs= 50;
t1=[0:1/50:1];
plot(t1,xa(t1));
title('Fs = 50');
xlabel('t');
ylabel('xa(t)');

%task iii b)
subplot(3,1,2);
% LCM = 60 pi
Fs=60;
t2=[0:1/Fs:1];
plot(t2,xa(t2));
title('Fs = Nyquist rate');
xlabel('t');
ylabel('xa(t)');

%task iii c)
subplot(3,1,3);
t3=[0:1/(5*Fs):1];
plot(t3,xa(t3));
title('Fs = 5 times Nyquist rate');
xlabel('t');
ylabel('xa(t)');

saveas(gcf,'output/task1iii.png')
```

Graph

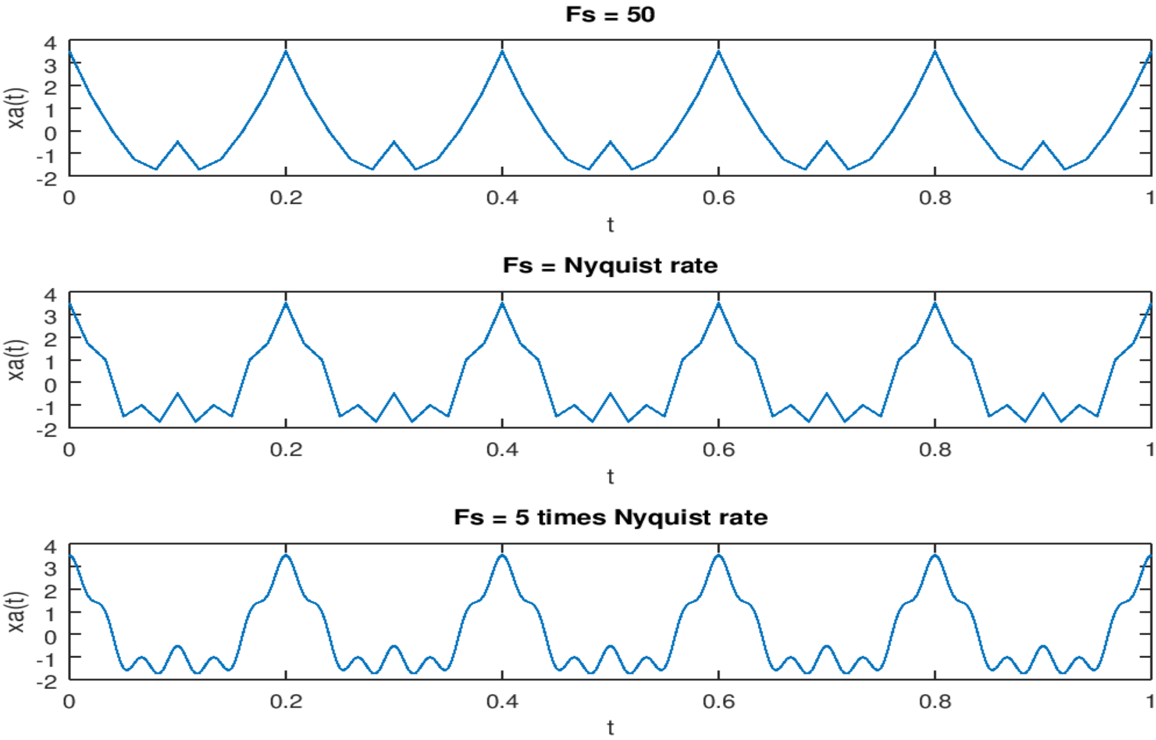

Figure 4: For Question 1 (iii)

Observations

- With the linear interpolation we ca one sampled wihat theth 300 samplen clearly see ts/cycle resembles the original continuous time function the most and the signal shown with Fs=50 has missing peaks which is not lost at Nyquist rate sampling or higher sampling rate.
- The curve is smoother for higher frequencies.
- No important data of the signal is lost at Nyquist rate or higher.

Conclusion

The minimum sampling rate required to correctly represent a signal is equal to the Nyquist rate which is equal to twice the bandwidth. F>=2B, Where, B is the bandwidth or frequency of the highest frequency component in the signal. F=2B is known as the Nyquist Rate. Here, Nyquist Rate=2 times 30=60 samples/cycle.

iv) Code

```
% task iv
figure
Fs=50;
N=Fs+1;
X=fft(xa(t1));
Xsq=(abs(X).*abs(X))/N;
Fk= [0:N-1]*(Fs/N);
bar(Fk,Xsq);
title('Fs = 50');
xlabel('Fk');
ylabel('Magnitude');

saveas(gcf,'output/task1iva.png')

figure
Fs=60;
N=Fs+1;
X=fft(xa(t2));
Xsq=(abs(X).*abs(X))/N;
Fk= [0:N-1]*(Fs/N);
bar(Fk,Xsq);
title('Fs = Nyquist rate');
xlabel('Fk');
ylabel('Magnitude');

saveas(gcf,'output/task1ivb.png')

figure
Fs=60*5;
N=Fs+1;
X=fft(xa(t3));
Xsq=(abs(X).*abs(X))/N;
Fk= [0:N-1]*(Fs/N);
bar(Fk,Xsq);
title('Fs = 5 times Nyquist rate');
xlabel('Fk');
ylabel('Magnitude');

saveas(gcf,'output/task1ivc.png')
```

Graphs

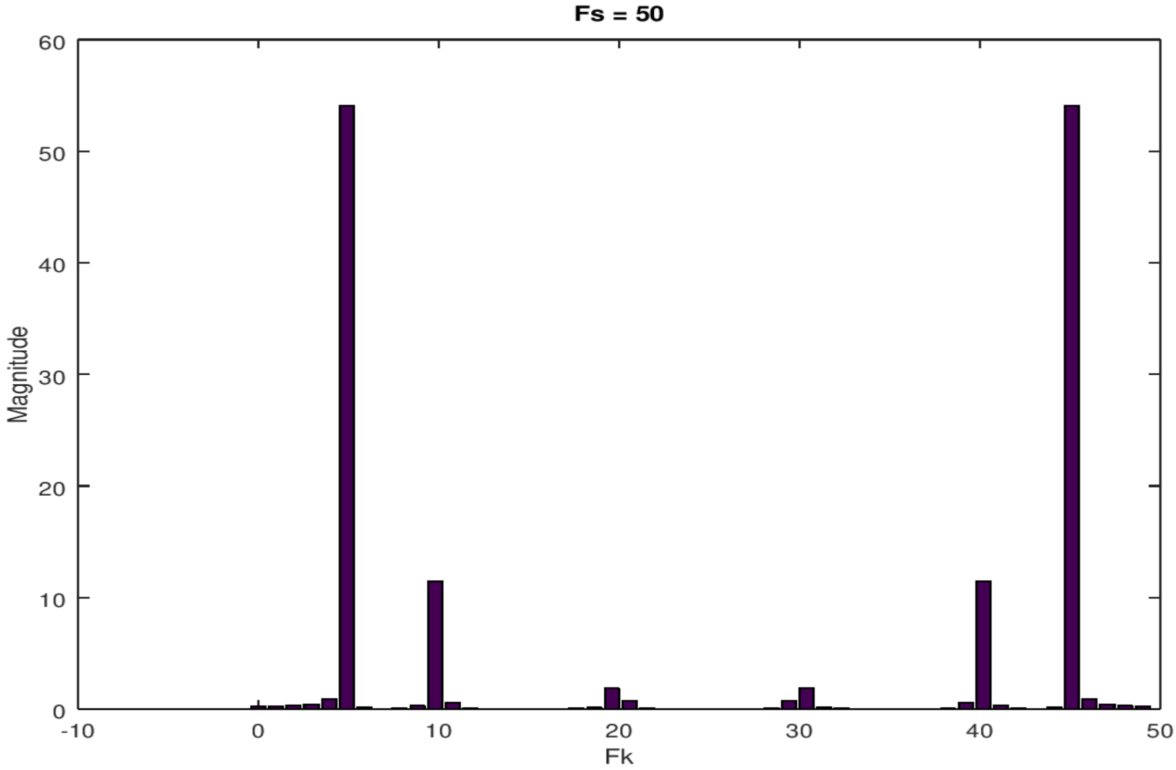

Figure 5: For Question 1 (iv) a

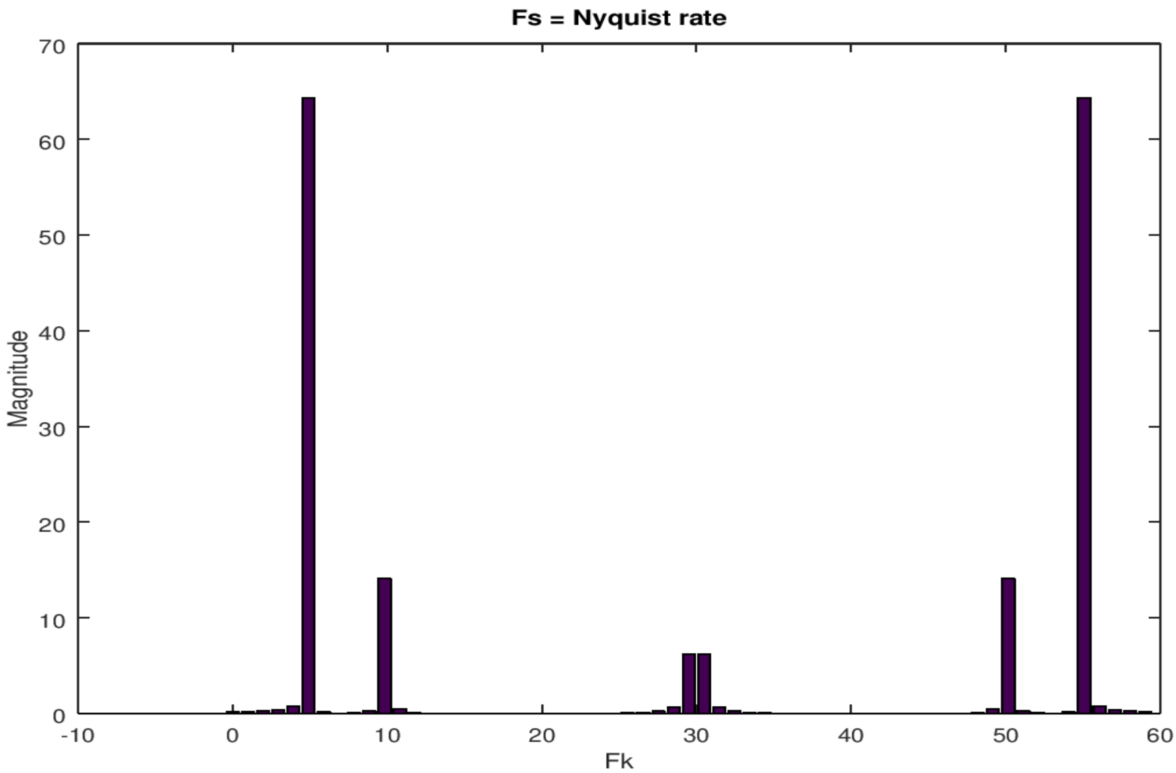

Figure 6: For Question 1 (iv) b

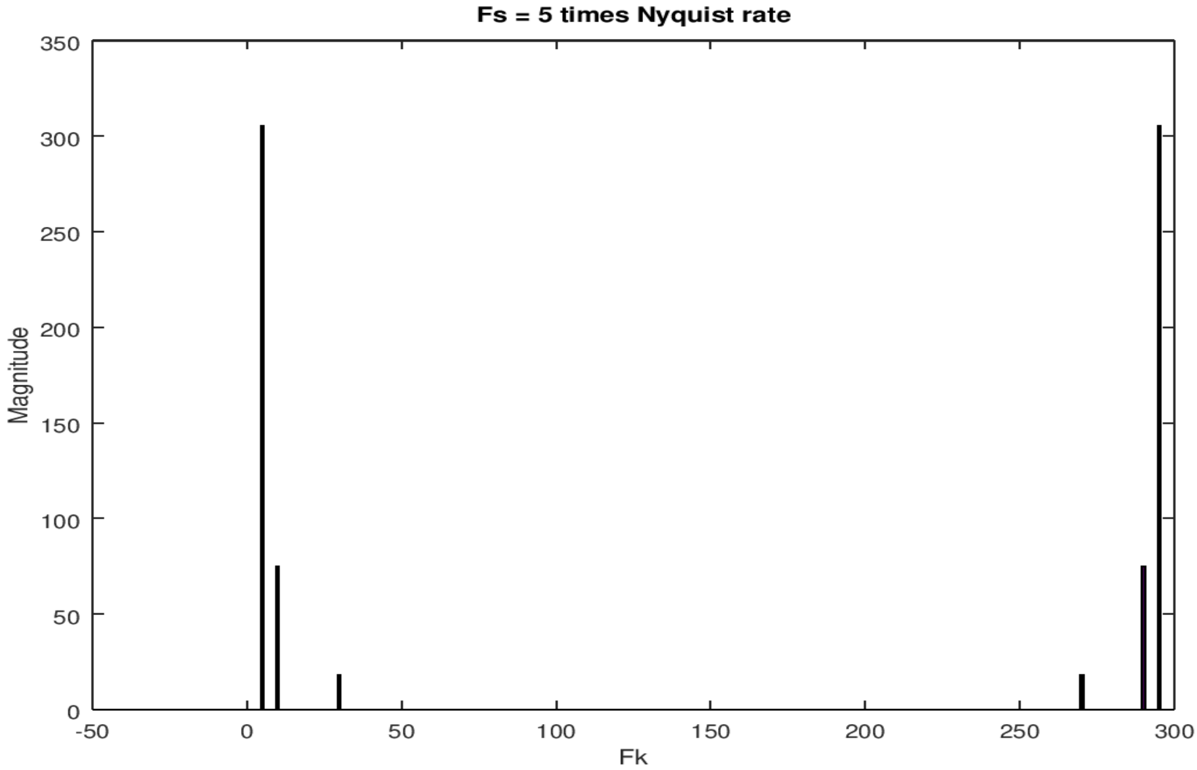

Figure 7: For Question 1 (iv) c

Observations

- The peaks occur at the frequencies of the component signals, i.e. at 5, 10 and 30 Hz for sampling rate 60 and 300 Hz.
- They occur at 5, 10 and 20 Hz for sampling rate=50 Hz.
- There is some noise even in the Nyquist rate spectrum which clears out at 5 times of it.
- There are additional peaks mirroring about Fs/2 which is the folding frequency.

Conclusion

50 Hz $<$ Nyquist Frequency, and -Fs/2 $<$ F $<$ Fs/2 or -25 $<$ F $<$ 25 Hz. So the peak at 30 Hz gets folded back or aliased to 20 Hz.

2.2 Question 2

Use the program for generating the sa-re-ga-ma... notes in Experiment No 0and try using different sampling rates (example: 1000Hz, 1600 Hz, 2200 Hz etc.) and write them in to different wav files and listen to see the difference

Code

```
F=[525 590 664 704 790 885 995 1055];
s=['1','2','3','4','5','6','7','8','9','10'];
for ii=[1000:500:2500]
        Fs=ii;y=[];
        for i=1:length(F)
                t=0.5*(i-1):1/Fs:(0.5*i-1/Fs);
                y=[y sin(F(i)*2*pi*t)];
        end
        wavwrite(y,Fs,['output/naveen' s((ii-500)/500) '.wav']);
        sound(y,Fs)
end
```

Observations

- The sound Sa Re Ga Ma tune becomes clearer as the frequency rises.
- For lower frequencies, Sa Re Ga Ma Pa cannot be heard properly (1000 Hz, 1500 Hz and 2000 Hz)

Conclusion

The sampling frequency should be higher than the twice the maximum frequency for the complete thing to be heard. 2500 > 2*1055 Hz, so it can be heard completely. As frequency gets higher, the sound clip becomes clearer.

2.3 Question 3

Load Track001.wav and generate different wav files with several values of the sampling rate (for example, half the original sampling rate, 1/3rd of the original sampling rate etc.) and see the effect of this different sampling rate on the audio. Remember: you need to do x1=x(1:K:end) to get every K th sample for a 1/K rate. Use wavwrite(x1,Fs/K,FileName.wav) to write it.

Code

```
[track,Fs]=wavread("data/Track001.wav");

wavwrite(track,Fs,'output/task3_org.wav');

track_1_2=track(1:2:end);
wavwrite(track_1_2,Fs/2,'output/task3_half.wav');

track_1_3=track(1:3:end);
wavwrite(track_1_3,Fs/3,'output/task3_athird.wav');

track_1_6=track(1:6:end);
wavwrite(track_1_6,Fs/6,'output/task3_onesixth.wav');
```

Observations

- As K increases there are lesser variations in the sampled sound and much of the information is lost. The tune also starts to sound very different.
- For K=2, 3 and 6 the song becomes less understandable and much more distorted.

Conclusion

The sampling frequency of any signal should be greater than twice the maximum frequency of the signal so that information is not lost. This is known as the Nyquist rate.