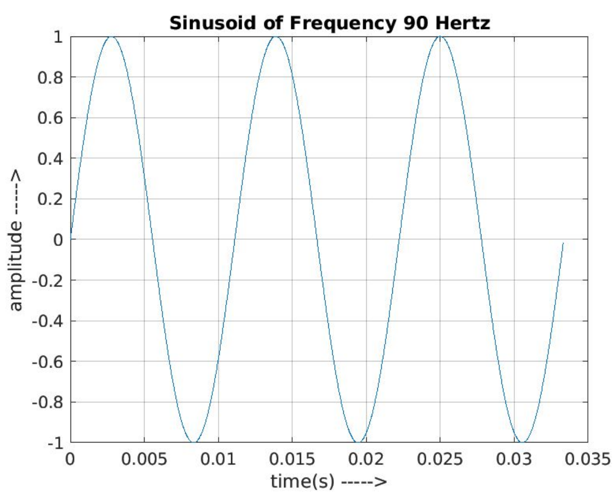
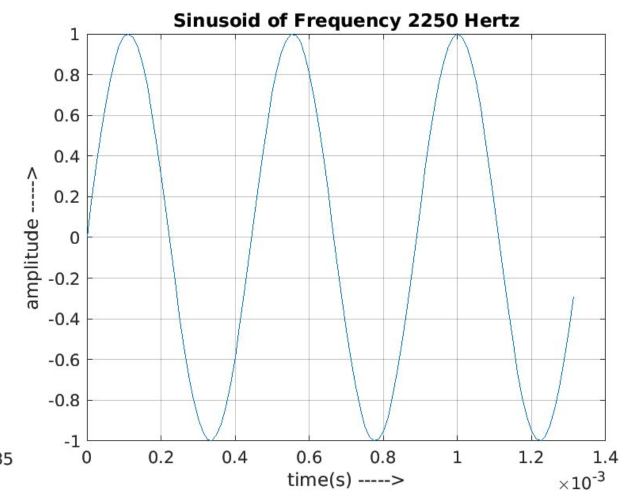
LAB ASSIGNMENT 1

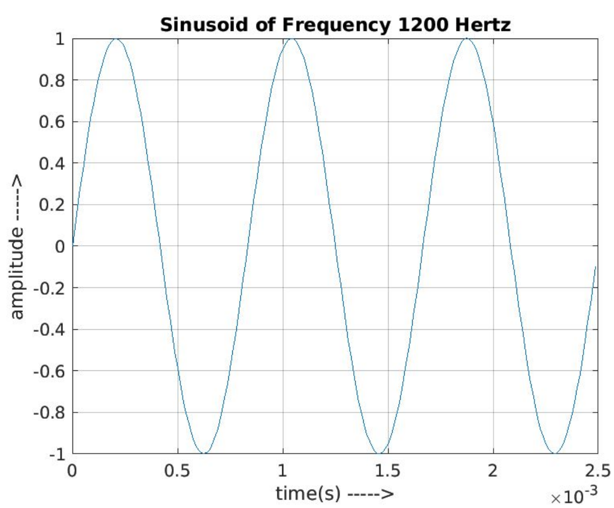
Kommineni Aditya - EE17B047 || V Sai Krishna - EE17B035

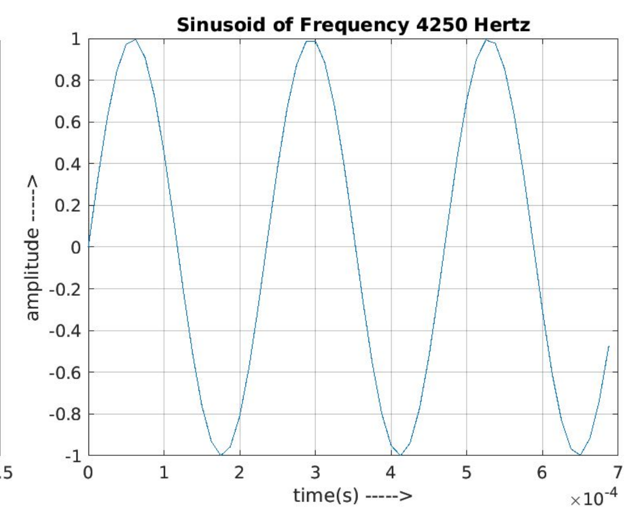
Question 1 :

The generated sinusoids were sampled at 80,000 hz which made sure that the signals were sufficiently sampled and none of them were undergoing aliasing.









When we generated sinusoids of a given frequency, as the frequency was varied we observed the following:

- The frequency of sinusoids which were audible ranged from 150 Hz to 16,000 Hz.
- As the frequency of the sinusoid increased, the sound starts to become shriller.
- Although the amplitude of the generated sinusoids is the same throughout, the loudness of the sinusoid initially increases upto a frequency of about 6,000 Hz and then, the loudness gradually decays.

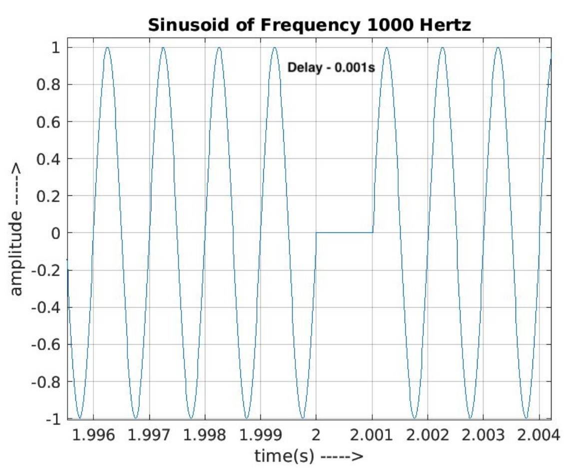
Inference:

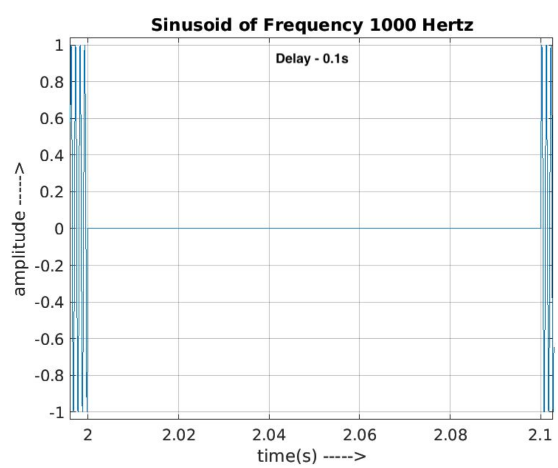
As the frequency of the applied sinusoid is increased, the sound becomes more shrill. Moreover, the human ear has a nonlinear relation to the loudness of the sound perceived as a function of frequency for signals of same intensity. This implies that although the signals of same intensity are played, the loudness perceived is dependent on the frequency content of the signal.

Question 2 :

We generated the delay using a vector composed of repeating the last sampled value of the sinusoid (which is approximately zero) and concatenating on both sides with sinusoids of zero phase so that the continuity isn't lost.

We generated various values of delay and noted down our observations. It is expected that for large values of delays, we should be able to hear two distinct sounds separated by a perceivable silence. However, if the delay is smaller than a given threshold, then both the effect of delay is unperceivable to the ear and the waveform is heard as a continuous sound.





The following are the observations pertaining to the experiment performed:

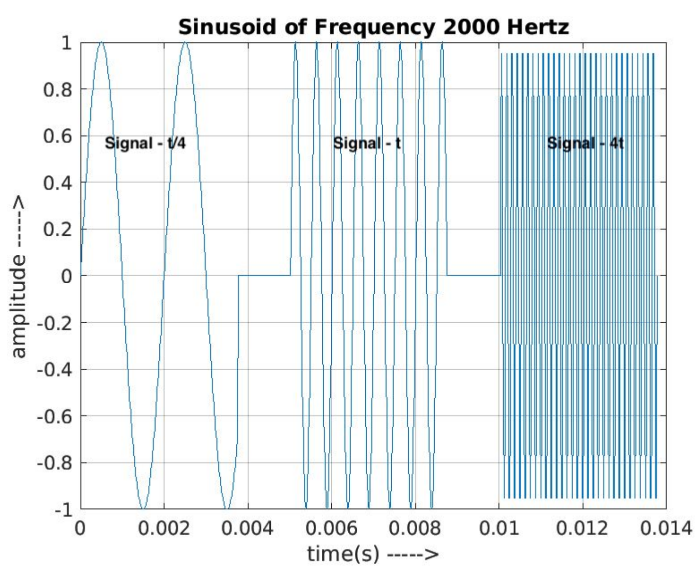
- For large delays we can distinguish the end of the first signal and the beginning of the second signal. However, when the delay is gradually reduced, below a certain threshold of delay the distinction between the end of the first signal and the start of the second signal is lost and it appears as though there is one continuous signal alone.
- The value of the threshold below which the delay isn't perceivable is about 0.05s. This value holds was observed for a majority of the frequencies which were tested out.

<u>Inference:</u>

The delay becomes unperceivable below a certain value because the persistence of hearing of the human ear is about 0.05s which implies that the sensation of any heard signal is present in the human brain for about 0.1s. therefore , if any signal delay is smaller than that, then the brain could perceive them to be the same continuous signal.

<u>Question 3 :</u>

The new signals generated by t/4 and 4t scaling of the signals will have different frequencies from the original signal we have picked.



The above picture is just a representation to indicate that the signals undergo a change in frequency. While observing, each of the signals were extended to about 2 seconds each.

The observations when the experiment is performed over a range of frequencies are as follows:

- From the plot below it is evident that the t/4 signal has a longer time period indicating a smaller frequency over the original and the 4t signal has a shorter time period indicative of a larger frequency over the original signal.
- The signal generated by t/4 transformation will be less shrill compared to the original signal and the signal generated by 4t will be more shrill. This reiterates the above point of them corresponding to lower and higher frequencies over the original signal.
- From the plot we can say that the frequencies are scaled by a factor of 4 i.e. t/4 signal corresponds to 500 hz and 4t signal corresponds to 8,000 hertz. This is the case with all the frequencies over the audible range.
- We should ensure that the sampling frequency of the signal is at least 4(scaling factor) times the normal sampling frequency of the original signal so that the higher frequency signal doesn't undergo aliasing.

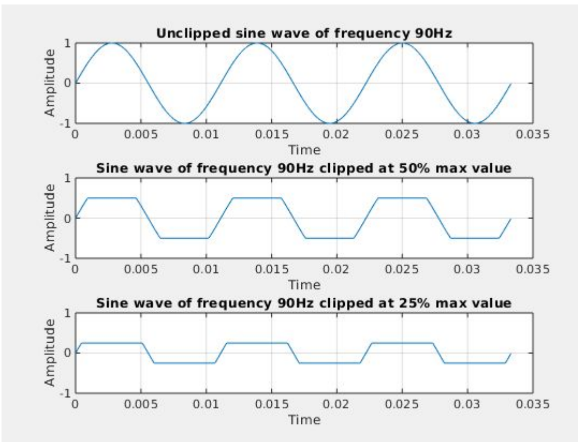
Inference: Time scaling will result in a corresponding change in the frequency spectrum of the signal. If the time domain signal is compressed then the frequency domain dilates and vice versa. This is in accordance to the Fourier Transform formula as below:

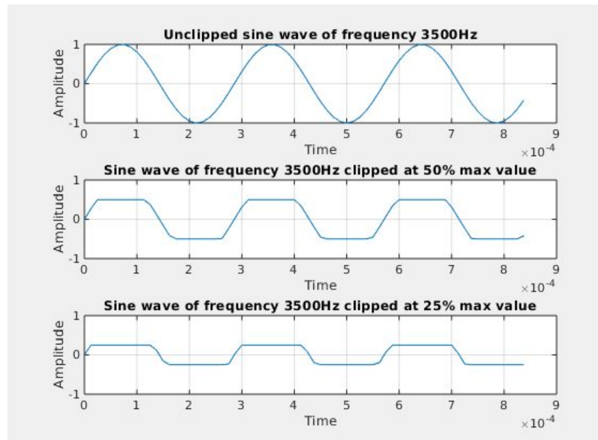
$$x(t) \leftrightarrow X(\omega)$$

$$x(at) \leftrightarrow \frac{1}{|a|} X\left(\frac{\omega}{a}\right)$$

Question 4:

For this part of the experiment, we will be using two sinusoids of frequency 90Hz and 3500Hz and we shall clip both of them at 50% and 25% peak value. The plots are shown below.





The observations are as follows:

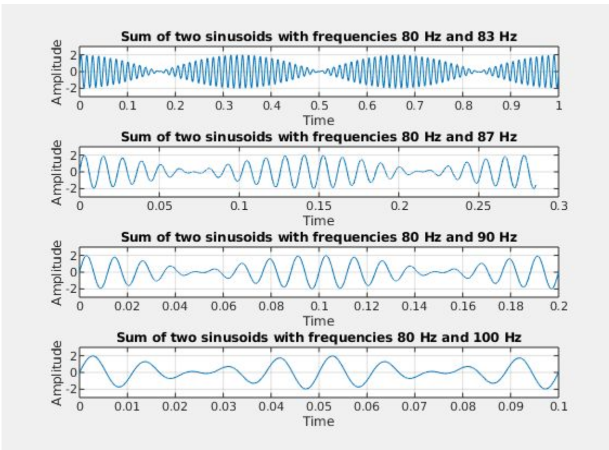
- While listening to the sounds plotted above in both cases, the clipped ones were less loud compared to the unclipped ones i.e. 25% max value was feebler than the 50% max value, which in turn was feebler than the unclipped sinusoid.
- The clipped sound waves are slightly noisy when you hear them i.e you could hear higher frequency tones.

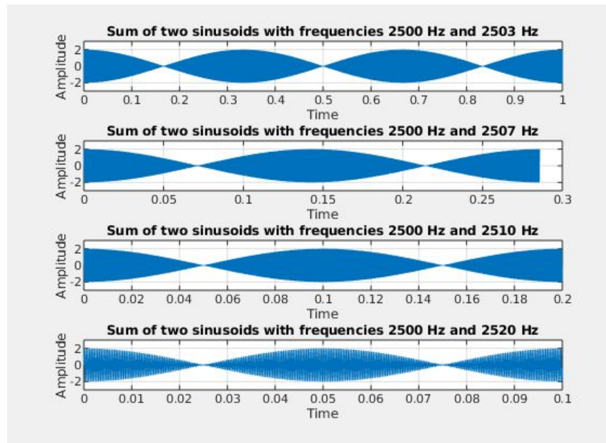
Inference:

- Clipped sinusoids are feebler than unclipped ones. This is because loudness is a measure of intensity and since waves with higher amplitude are more intense, clipped sinusoids sound less loud (feebler).
- Clipped sinusoids contain higher frequency terms. This can be explained better if we look at the Fourier Series of this signal (periodic signal hence Fourier series exists). Since the signal contains abrupt changes (non-differentiable points) we will require an infinite number of non zero Fourier terms to represent the signal. In other words, it will not contain just one frequency like an unclipped sinusoid but instead, have multiple frequency components.

Question 5:

The x values we chose for this part of the experiment are 80 Hz, and 2500 Hz and the Δx values we chose are 3 Hz, 7 Hz, 10 Hz, and 20 Hz.





The following are the observations pertaining to the experiment performed:

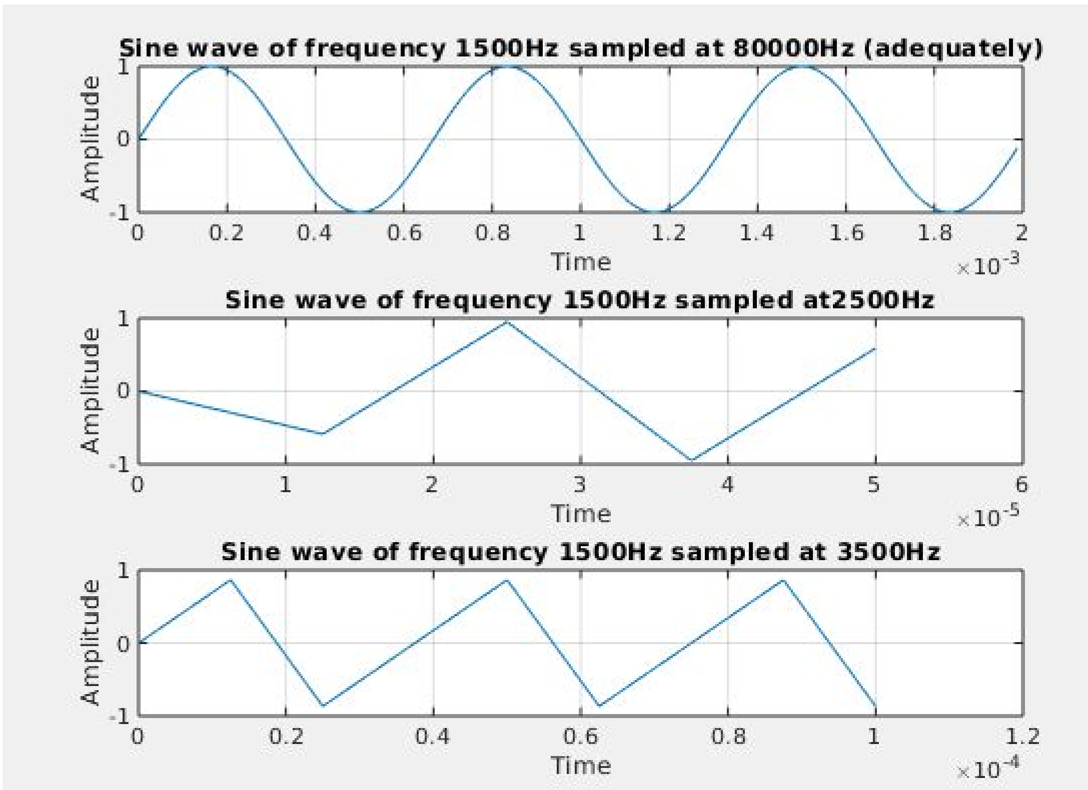
- In both cases, for Δx = 3 Hz and 7 Hz, we could witness the loudness of the sound wobbling periodically. However, as we continue to increase Δx, we fail to observe this event.

Inference:

- The phenomenon that we observed is called beats and it occurs when two signals of different frequencies interfere. Since they are of different frequencies, they interfere constructively and destructively at different points. This alternating constructive and destructive interference causes the sound to be alternatively soft and loud, which we perceive as wobbling.
- As Δx increases, the periodicity of the wobbling increases. Hence, we stop observing beats as we keep increasing Δx because the periodicity of wobbling would eventually cross the persistence of the human ear.

Question 6:

We will generate a tone of 1500 Hz and sample it adequately. Then we will sample the signal at lower/higher than Nyquist rates and listen to all three of them.



The observations are as follows:

- From the above plot, we can see that the signal is constructed quite bad when it isn't adequately sampled i.e it looks like a triangular signal instead of a sinusoid.
- While listening to the sound clips plotted above, we perceive the undersampled signal as a low-frequency sound. However, there is no difference while listening to the adequately sampled sinusoid and the one sampled higher than the Nyquist rate.

Inference:

The reason for perceiving the undersampled sinusoid with a lower frequency is aliasing. An undersampled signal will overlap in the frequency domain. As a result, the periodicity of the signal will decrease and in turn, we perceive it with a lower frequency. This doesn't happen when we sample it at more than the Nyquist rate, as there will be no aliasing.

REFERENCES :

This is the link to the codes for each question: https://drive.google.com/open?id=1Sc2zdFCzgWesOkhc4kmBqtkQ3pT-CVMx