Signals Assignment 1

In this assignment, a clean 1000 Hz sine wave and a voice recording of me attempting to replicate the same frequency are compared. The two figures I provide represent both in the time domain(Fig. 1), and the magnitude spectrum of both(Fig. 2).

Figure 1 displays the clean 1000 Hz sine wave and the singing recording in the time domain. As seen, the 1000 Hz signal is completely covered in blue, which represents a pure tone at a single frequency value. A pure tone has a constant amplitude over time, with no fluctuations, therefore, there is a solid blue rectangle. As for my singing recording, it clearly shows varying amplitudes over time – this tells us that even though I tried to match a 1000 Hz tone, it is not clean – in fact it is not possible, as the human voice produces multiple frequencies or harmonics. Furthermore, due to noise in the surroundings, there will be more distortion in the spectrum.

Figure 2 is the reproduction of the plot required, it shows plots of the magnitude spectrum of the 1000 Hz signal and my singing recording. This was obtained by using a Fourier transform to convert the time-domain signal into the frequency domain. The Fast Fourier Transform(FFT), a computational algorithm was applied to the earlier signals. Both of them were plotted on one graph for comparison, and we can see clearly that there is only a peak at 1000 Hz for the clean signal, whereas there are additional peaks in the magnitude spectrum surrounding the target 1000 Hz for my singing plot.

When working with signals like these, the sampling rate plays an important role. To avoid aliasing, which happens when higher frequencies are misinterpreted as lower ones, the sampling rate must be at least twice as high as the highest frequency in the signal(Nyquist theorem). Moreover, the sampling rates for the clean signal and the voice recording should match in order to compare them. In this case, since my recording was done on the voice recorder app on the laptop, the sampling rate was checked and found to be 48 kHz(48000 samples per second), so the clean 1000 Hz signal was also given the same sampling frequency. This also satisfies the Nyquist theorem.

In order to reproduce the plot, proper normalization of the FFT results and coversion of FFT magnitude to decibels(dB) is essential. The energy in the FFT is distributed across positive and negative frequencies, so multiplying by 2 (for positive frequencies) ensures that the amplitude correctly represents the original signal. Moreover, in order to ensure that individual samples are 2 Hz apart, the sampling interval should be taken to be 0.5 s.

To reproduce the signal through headphones, it is crucial to match the sampling rate for both signals, use high quality headphones(with flat frequency response), and ensure volume to set to level that does not cause distortion.

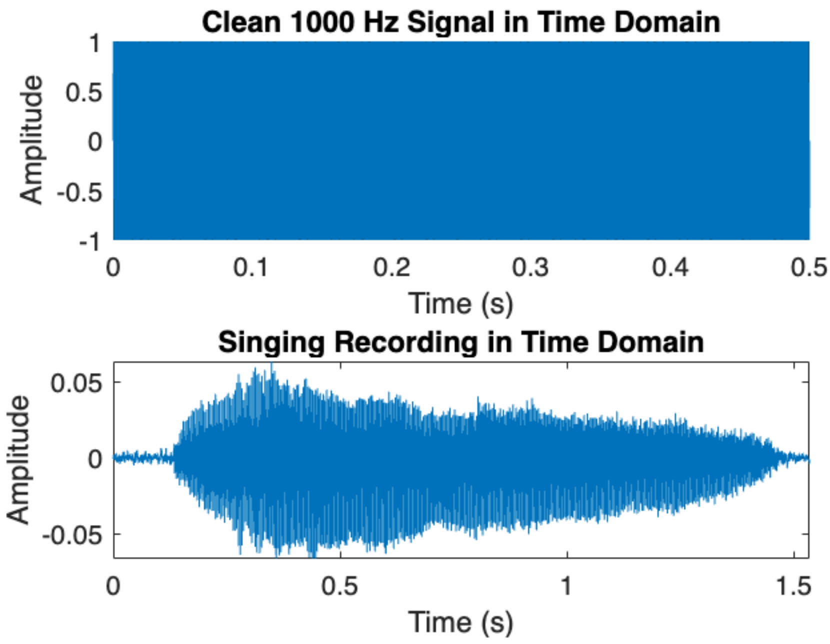

Figure 1: Clean 1000 Hz signal and my singing recording in the time domain

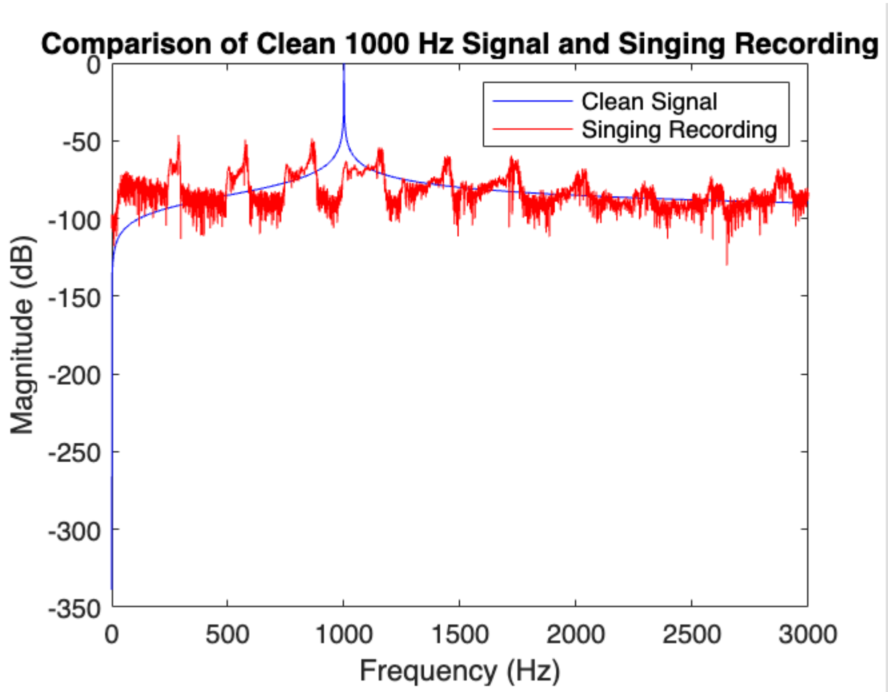

Figure 2: Magnitude spectrum of both signals