Comparison of a Pleasant and Unpleasant Sound

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Abstract: This is a study to understand the mechanism of Human hearing which interprets sound as pleasant and unpleasant. The sound of a Piano is considered as Music whereas the sound of a Pressure Cooker is considered as Noise. The difference in the frequencies plays a major role in categorising a sound as Pleasant and Unpleasant which we can interpret using Discrete Fourier Transforms.

Keywords*: Discrete Fourier Transform, Fast Fourier Transform, Frequency spectrum.*

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I. INTRODUCTION

The Human Ear is capable of hearing sound in the range 20-20,000 Hz.

Audible sounds are classified into two groups, namely musical sounds and Noise. Three major factors that contribute to a sound being pleasant and unpleasant are Loudness (Amplitude), Pitch (frequency) and Quality or timbre.

Music is ordered sound. A musical Sound is that in which the vibrations of the sounding body are periodic, follow each other regularly and rapidly so as to produce a pleasing effect on the ear without any sudden change in loudness. The component frequencies of music are discrete (separable) and rational (their ratios form simple fractions with a discernible dominant frequency). A sound must have an identifiable Pitch, a good or pleasing quality of tone, and repeating pattern or rhythm to be music.

Noise is disordered sound. Noise has no identifiable pitch, no pleasing tone and no steady rhythm. The component frequencies of noise are continuous (every frequency will be present over some range) and random (described by a probability distribution) with no discernible dominant frequency.

The Fourier Transform(FT)[1] decomposes a function of time(a signal) into the frequencies that make it up, in a way similar to how a musical chord can be expressed as the frequencies (or pitches) of its constituent notes. The Fourier Transform of a function of time itself is a complex valued function of frequency whose absolute value represents the amount of that frequency present in the original function, and whose complex argument is the phase offset of the basic sinusoid in that frequency. The Fourier transform is called the frequency domain representation of the original signal. The term Fourier refers to both the frequency domain representation and the mathematical operation that associates the frequency domain representation to a function of time.

The Discrete Fourier Transform (DFT)[3] is a specific kind of discrete transform, used in Fourier analysis. It transforms the function into another, which is called the frequency domain representation of the original function (which is often a function in the time domain). The DFT requires an input function that is discrete. Such inputs are often created by sampling a continuous function, such as a person's voice. The DFT is widely employed in signal processing and related fields to analyse the frequencies contained in a sampled signal, to solve partial differential equations, and to perform the operations such as convolutions or multiplying large integers. A key enabling factor for these applications is the fact that the DFT can be computed efficiently in practice using a Fast Fourier Transform algorithm(FFT). FFT algorithms are so commonly employed to compute DFTs that the term FFT is often used to mean DFT in colloquial settings. Formally there is a clear distinction. "DFT" refers to a mathematical transformation or function, regardless of how it is computed, whereas "FFT" refers to a specific family of algorithms for computing DFTs. The sequence of N complex numbers $x_0, x_1, .., x_{N-1}$ is transformed into another sequence of N complex numbers according to the DFT formula:

$$\mathbf{F(k)=FFT\{f(n)\}=} \sum_{n=0}^{N-1} f(n) e^{-i2\pi kn/N} \quad , k = 0, 1, 2, ..., N-1$$

where $\frac{k}{n}$ – frequency measured in cycles per sampling interval

N - number of samples

n - the range of N

f - frequency represented by the frequency number

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For the comparison of two sounds, Piano music and sound of a Pressure Cooker are considered. Intuition tells us Piano music is pleasant and Pressure Cooker sound is unpleasant and thus we categorise them as music and noise respectively. As Piano music is considered good for mental health it gives morality boost and there is no doubt in our mind that it is pleasant. On the other hand, Pressure Cooker sound is common in our everyday life even though we are used to hearing it on a regular basis, the sound is always annoying and thus can be categorized as noise. Even though with simple argument, we are able to reason, and conclude that one is pleasant and other is unpleasant, generally speaking such a simple task when proving mathematically can become monumental at the same time monumental could become simple with the help of right software. For this study, MATLAB R2016b[2] software has been applied, for the purpose of recording as well as discerning. As the first step, the sound from a Piano and Pressure Cooker are recorded using a microphone into the software as audio files as shown in Figure 1 and 2.

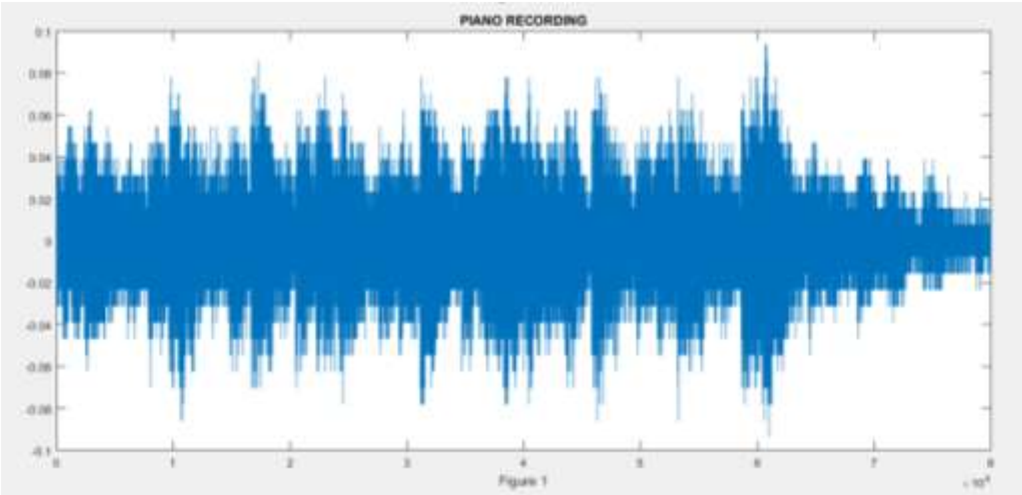

Figure 1

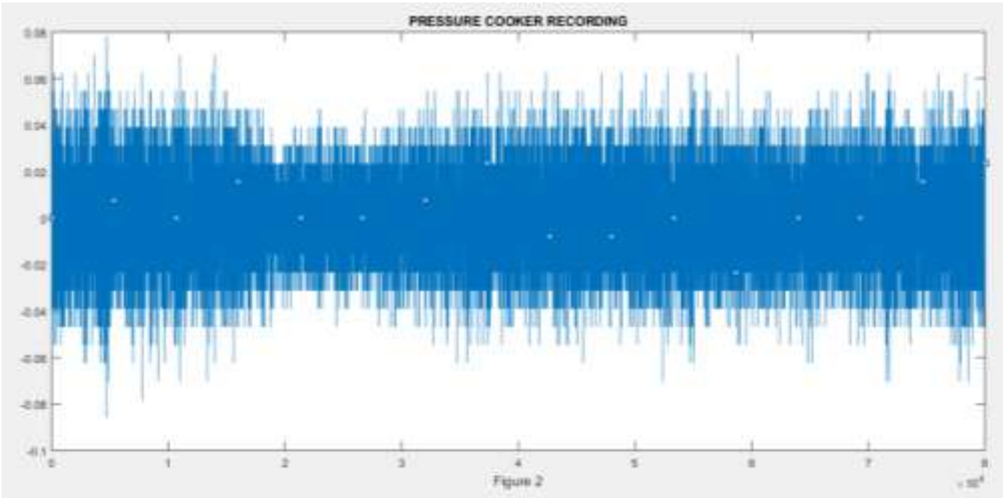

Figure 2

The recording time considered is 15 seconds. Since the software is able to produce the graphs, of the sound waves of Piano and Pressure Cooker, in real time, our task is simplified to understand and interpret only the graph. The converted audio files in wave form is shown here as Figure 3 and 4.

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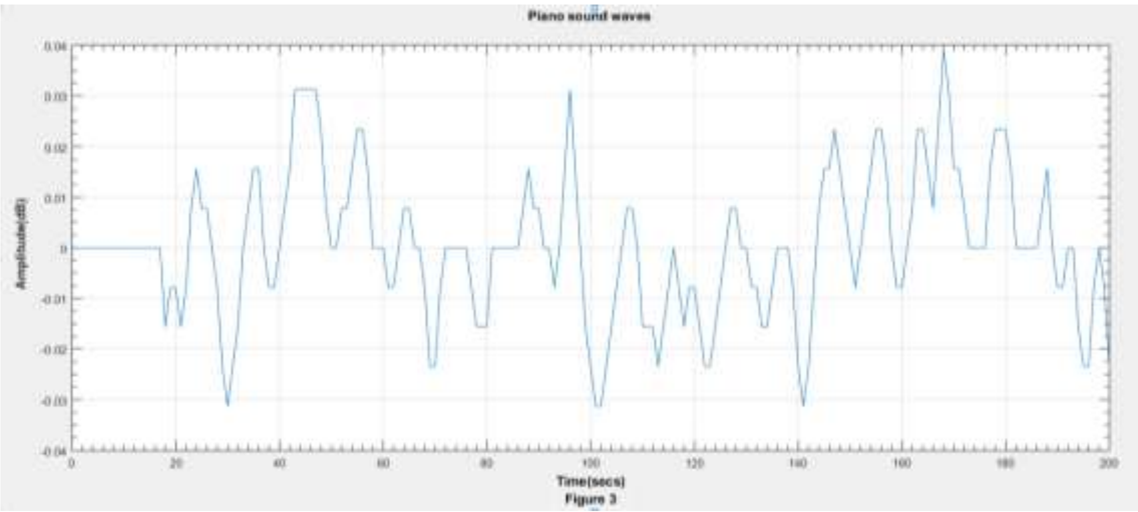

Figure 3

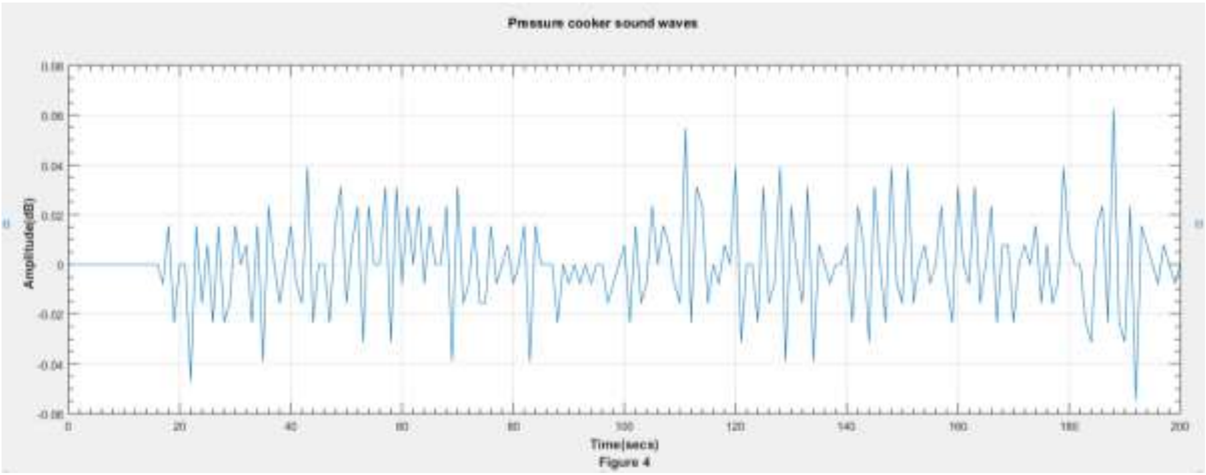

Figure 4

From the above two figures, we are unable to derive any conclusion. It is difficult to differentiate one from another. To understand the behaviour of any sound wave, using DFT we pick up finite points from each of the curve that will be our sample point N and then apply the DFT formula. Depending on the sound wave, the number of samples will be decided. For example, if we select a single sine wave with 1 Hz, generally N=8 will suffice. Here, as we have taken 15 seconds of recording of sound wave, when we start to analyse, the sample number becomes very large. Since calculating manually is tedious there is a provision in MATLAB to use the algorithm FFT. Once FFT is applied, we can make a visual representation using the magnitude obtained from the DFT calculations. MATLAB has abs function which calculates this magnitude automatically which can be plotted as a graph as shown in Figure 5 and 6.

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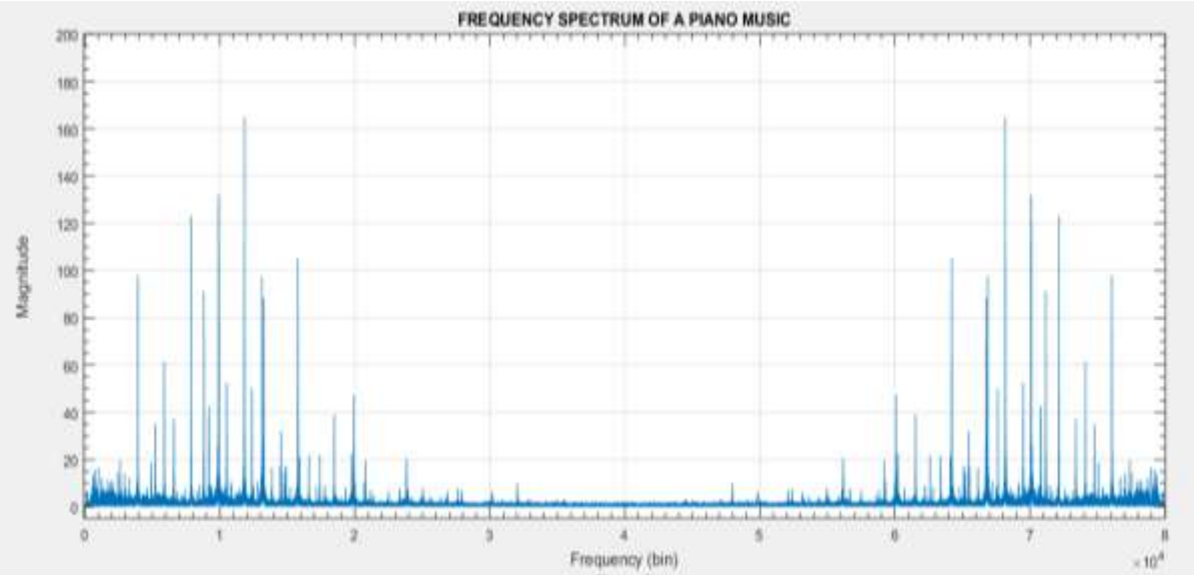

Figure 5

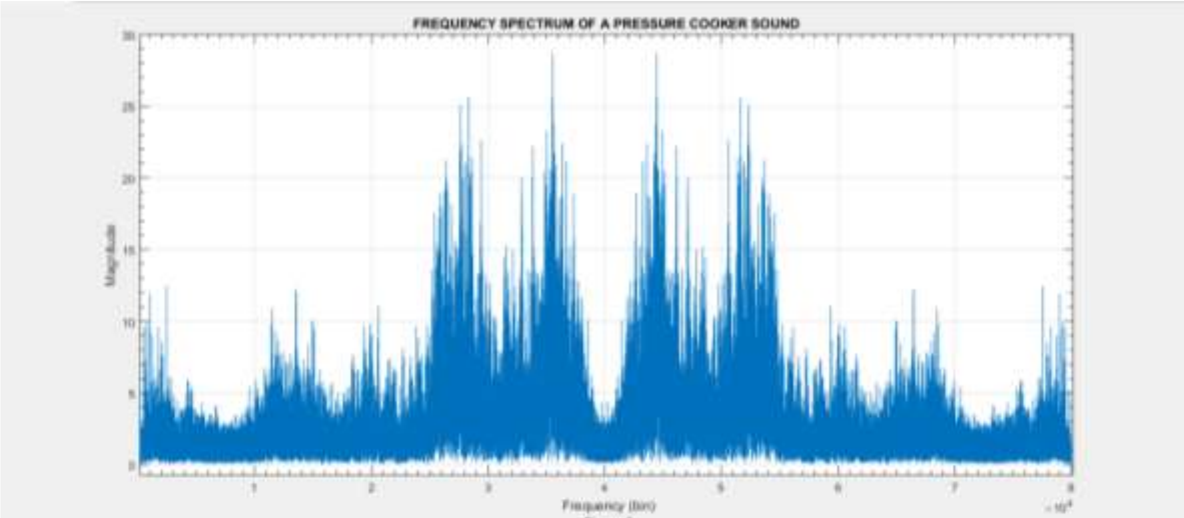

Figure 6

These are the frequency spectrum of both the sound waves. Through frequency spectrum we can analyse the behaviour or pattern of the sound waves. Looking at figures 5 and 6 the first thing we notice is that the graph is mirroring half way through, that is, after the midpoint of the frequency(bin), the first half is reflected on the second half. This is due to Nyquist effect.

In signal processing, the Nyquist rate named after Harry Nyquist is twice the bandwidth of a band limited function. When a continuous function x(t) is sampled, at a constant rate, fs samples per second there is always an unlimited number of continuous functions that fit the same set of samples. But only one of them is band limited to ½ fs cycles per second.(Hertz). It follows that x(t) is band limited to ½ fs. This is called Nyquist criterion.

Hence it is enough if we focus our attention on the first half of the frequency spectrum which is shown in figure 7 and 8.

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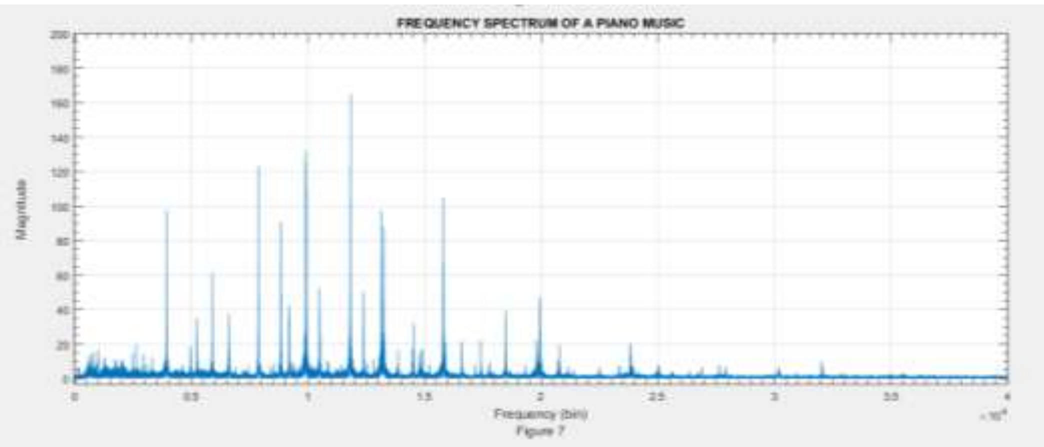

Figure 7

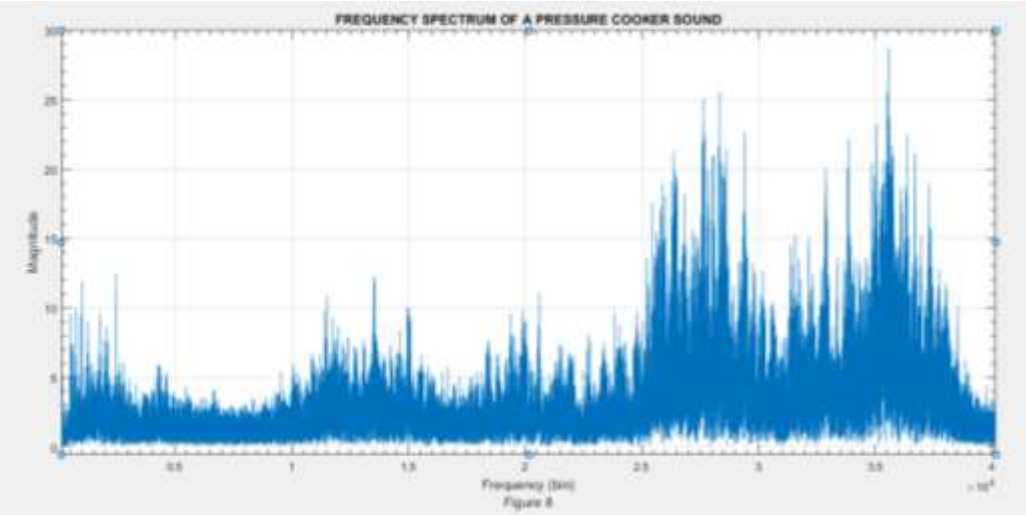

Figure 8

II. EXPERIMENTAL RESULTS

In the frequency spectrum each of the vertical lines are indicators of presence of sinusoids. The graph is tightly packed with many vertical lines, meaning there are thousands of sinusoids making up the sound waves of both sound. The prominent sinusoids are but few. It can be immediately observed that the major difference between the two spectrum is the number of prominent frequencies with low frequency bin and high frequency bin . For Piano, there are fewer peaks which are of lower Hertz with high magnitude but for Pressure Cooker, there are numerous peaks which are of higher Hertz. There are only 7 tall peaks for Piano above 60 magnitudes, with Hertz between 0×10^4 to 2×10^4 frequency(bins). For Pressure Cooker, there are some 23 peaks between 2×10^4 to 4×10^4 above magnitude 20 .Even though for both the waves, there is a continuous presence of sinusoids, from the start to end, but the sinusoids with higher magnitude occur at lower Hertz for Piano and at higher Hertz for Pressure Cooker which becomes the deciding factor that makes the sound pleasant for Piano and unpleasant for Pressure Cooker. Also contributing factor is that for Pressure Cooker, continuously from frequency bin 2×10^4 to 4×10^4 there are sinusoids with magnitude higher than 10 and up to 15 whereas for Piano, above 10 there are fewer peaks. From this frequency spectrum it is possible for us to deduce the magnitude, frequency and phase value of each of the sinusoids but the number will go in thousands. The sinusoids corresponding to the tallest peak of the Piano and Pressure Cooker may be found since the number is small.

III. CONCLUSION

Based on our discussion above it is a straight forward to conclude that sound waves with lower Hertz make pleasant music and sound waves with higher Hertz make unpleasant music. This is also in keeping with the findings of researchers[4] before us that pleasant healing music is of low frequency and harsh loud bursts of sound are of higher frequency. Hence it is of no surprise in our own conclusion. But this has been a personal find in line with the existing knowledge and hence a fruitful and interesting work. This work can further be continued by figuring out the magnitude, frequency and phase value of the sinusoids that make up both the wave forms. It will be challenging work as obviously there are thousands and thousands of sinusoids

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present in these waves. But it will complete the problem we started.

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