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ME370 KDOM LAB: Experiment No. 6

Vibration Data Analysis

In this lab you will be analysing vibration data. Vibration data is acquired using measurement devices (sensors, data acquisition systems etc.). In order to properly acquire and analyse vibration data, certain aspects concerning the measurement process need to be kept in mind. This lab will introduce you to some of the issues taken into consideration in the analysis of vibration data. You will see how altering certain parameters of the measurement process affect the acquired vibration data and how signal processing is employed to condition the data and extract required information from it.

Sampling frequency. : The sampling rate or sampling frequency defines the number of samples per unit of time (usually seconds) taken from a continuous signal to make a discrete signal. Any digital measurement system acquires data in a discrete manner i.e. it does not continuously make measurements but only at particular time instants. Each such measurement made at a particular time instant is called a sample. These samples are used to reconstruct the signal we are trying to measure. The more such samples acquired, the more accurately will the reconstructed signal resemble the actual signal. As the sampling frequency is increased, it starts to approximate a continuous measurement closer. (The motor experiment conducted earlier, using Labjack acquired data at 500Hz). Sampling rates of several MHz is easily achievable today. However, there are certain overheads with high sample rates, such as cost, data management etc. Thus it is desirable to find out an optimum sampling rate which will accurately acquire the signal but will not prove too costly.

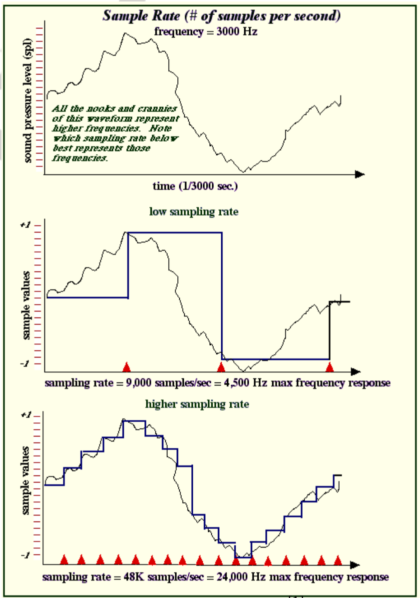


Fig 1. Sampling of a Signal [1]

If the sampling rate is too small, then the signal won't be measured correctly and a phenomenon known as aliasing occurs. In order to avoid this, frequency of sampling must be suitably chosen. The Nyquist–Shannon sampling theorem states that perfect reconstruction of a signal is possible when the sampling frequency is greater than twice the maximum frequency of the signal being sampled. In practice, since in most cases it is not known in advance what the maximum frequency in a measurement will be, a suitable initial sampling rate is chosen and a measurement is made. Subsequently the sampling rate is increased and the measurement is repeated. This process is continued till subsequent measurements don't show much difference.

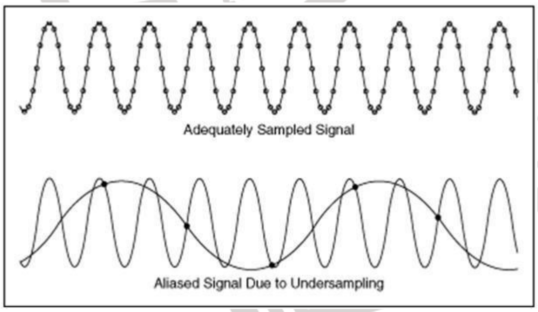
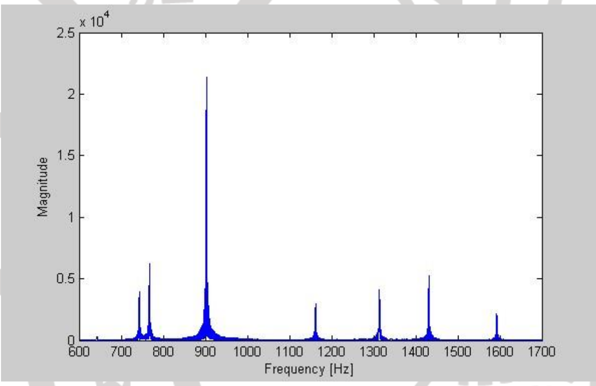


Fig 2. Aliasing [2]

<u>Signal Processing</u>: Often the data acquired in a measurement needs to be processed before analysis can be done. There are several reasons for this. One reason is that, along with the signal that we are interested in measuring, unwanted signals, known as noise are also captured by the measurement device. This could be due to a problem in the system being measured itself, but often it is a result of extraneous signals that get captured from the surroundings, the most common being the 50Hz line frequency noise, which is almost always present in any environment. Thus before analysing the data we would like to remove this noise. This process is known as filtering. Different kinds of filters exist classified based on their application. Based on the frequencies they filter out and allow to pass, they are classified as low-pass, high-pass or band pass.

Windowing: Another kind of operation is windowing. A measurement is started at a certain time instant arbitrarily chosen and similarly the measurement is stopped at some arbitrary instant, once we believe we have enough data or samples to analyse the system. In order to find out the frequency content of the signal from the data obtained, an operation known as Fast Fourier Transform (FFT) is performed which gives the various frequencies contained in a signal and their relative amplitudes. This data is typically plotted with frequency on the X-axis and magnitude on the Y-axis. The FFT computation presumes that the input data repeats over and over i.e. it concatenates the acquired data to form an endless data set. Since the measurement was started and ended at arbitrary time instants, it is possible that the initial and final values of the data set are not the same; the discontinuity causes aberrations in the spectrum computed by the FFT. In order to mitigate this effect, a "Windowing" operation is done which smoothens the ends of the data to eliminate these aberrations. A window is shaped so that it is exactly zero at the beginning and end of the data block and has some special shape in between. This function is then multiplied with the time data block forcing the signal to be continuous.



However, since a windowing function alters the data, it affects the frequency spectrum. Thus there are many windowing functions available (such as Hanning, Barlett, Tukey), each with their own advantages and disadvantages and are suitable for specific kinds of signals (such as random, sinusoidal)

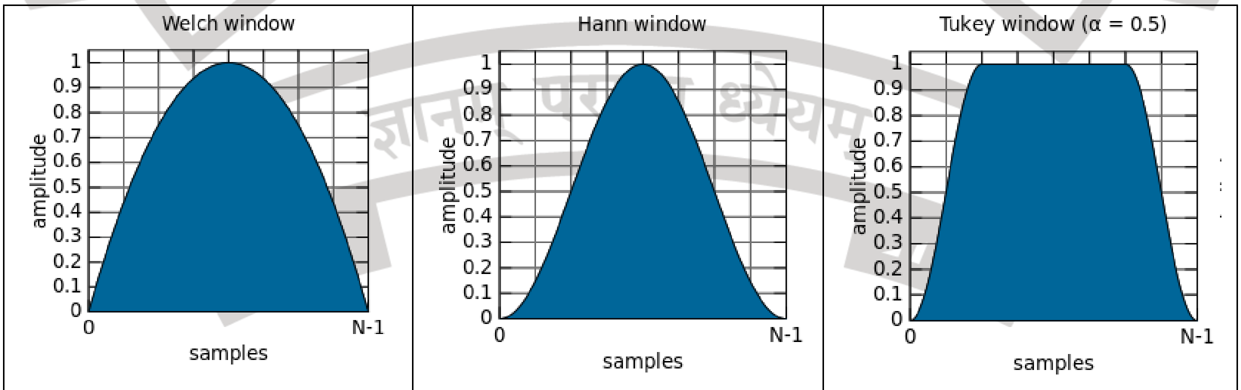


Figure 4. Different Windowing Functions – (a) Parabolic (b)Hann (c) Tukey (tappered cosine) [3]

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You have been given a file with simulated vibration data. It contains 500 samples, corresponding to 1 sec i.e. the sampling rate is 500Hz. The first column is time and each of the columns B-M contain a separate vibration measurement or sample. You have to choose the data set corresponding to your group no. (Group 1 – Sample 1, Group 2 – Sample 2 and so on). Import the data into MATLAB from the Excel file using the `xlsread` command. This will save the entire spreadsheet into a 2D array, from which you can select the relevant columns.

You are required to sample the data at different sampling rates and do a FFT analysis on this sampled data. Before performing the FFT analysis you must apply a suitable windowing function to take care of the problem explained earlier.
The data given to you is sampled at 500Hz. In order to sample at a lower rate, skip samples. For
example, if you consider only alternate samples, the sampling rate gets halved (250Hz) i.e. the data set you get by skipping alternate measurements corresponds to the situation in which you would be sampling the data at 250Hz

In order to perform the windowing operation, first choose a suitable windowing function. Apply this window function to the sampled data.

In order to do the FFT analysis, use the command `fft` in MATLAB. `fft(x)` will return the frequency content of the data stored in vector x. Read the following MATLAB Help page for details on how to use the `fft` function. http://www.mathworks.in/help/matlab/ref/fft.html

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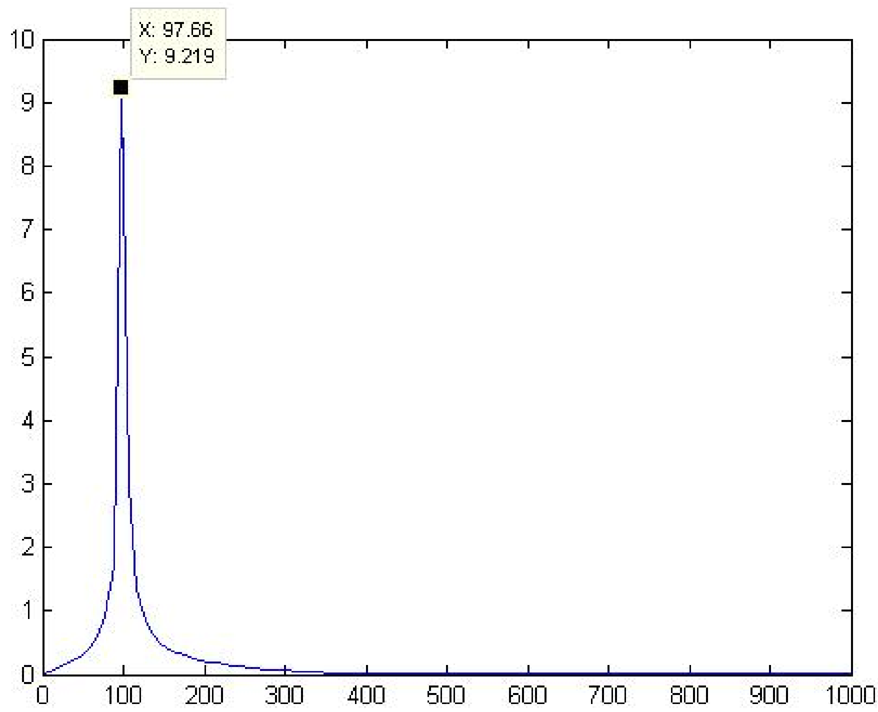
Details about FFT and Windowing

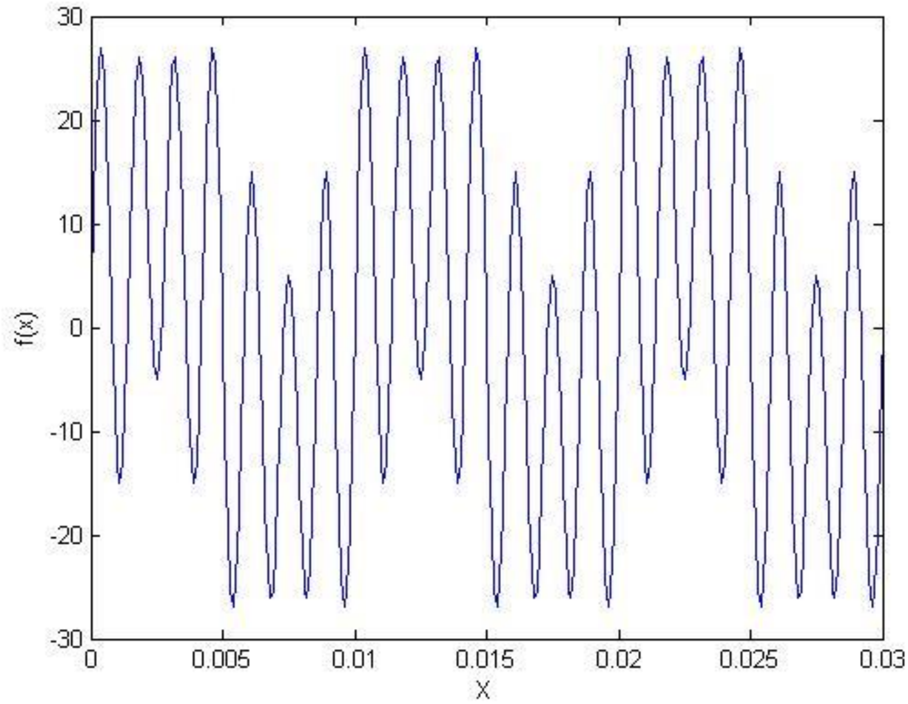
FFT stands for "Fast Fourier Transform" which is an algorithm to compute discrete Fourier Transform and its inverse. Many times analysing the signal in the frequency domain provides more information than time domain. Fourier analysis converts time domain signal to frequency domain signal and vice versa.
Generally, any periodic function f(x) can be written as infinite sum of sine and cosine functions with the help of Fourier series (as you would have studied in MA 10x in first year). FFT is the plot between frequencies (on X-axis) and amplitudes (on Y-axis) of those sine and cosine functions. For example, FFT of the function $f(x) = 10 * \sin(100x)$ is as shown below.



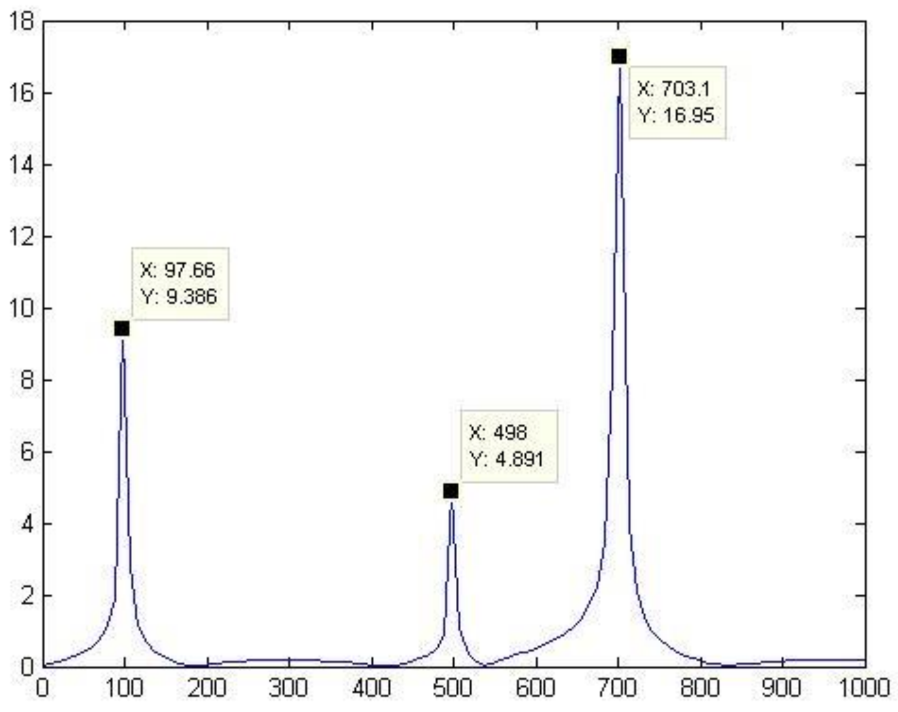
Here, X-axis is frequency and Y-axis is amplitude. So at 100Hz, we get one sharp peak whose amplitude is approximately 10.
Similarly, if function is the sum of different sines e.g.

$$f(x) = 10\sin(100x) + 5\sin(500x) + 20\sin(700x)$$

F(x) when plotted vs x, looks like as shown in the figure below.



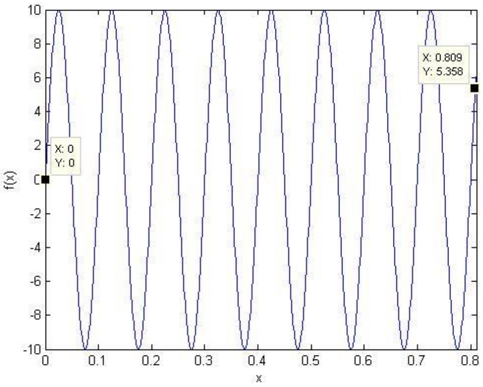
FFT of this function will have 3 peaks at frequency equal to 100, 500 and 700 Hz with corresponding amplitudes.



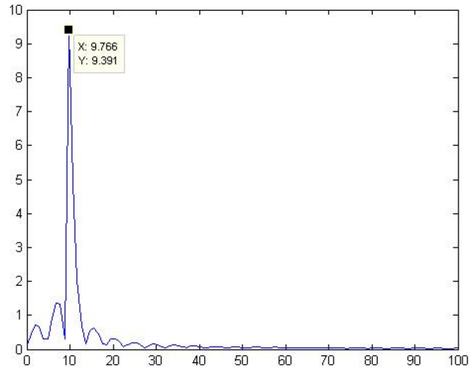
Since any periodic signal can be written in the infinite sum of sine and cosine functions, when plotted on FFT, it can have large number of frequency components.

Windowing: -

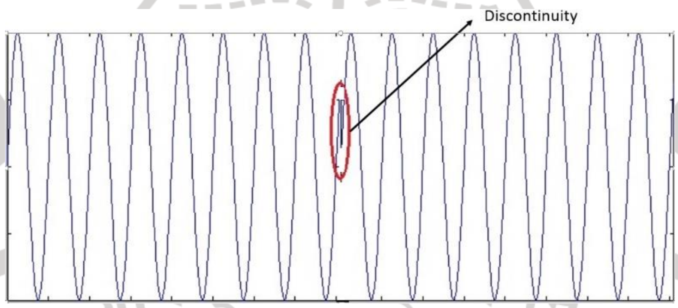
A fundamental assumption in Fourier analysis is that the signal is **periodic** from $t = -\infty\ to + \infty$. Since any measured signal has to be finite, the FFT computation presumes that the measured data repeats over and over till $t = \infty$, i.e. it concatenates the acquired data to form an endless data set. Measurement of the signal can start and stop at any arbitrary time instant, and need not be such that complete cycles of the signals (at different frequencies) within it should be present within the time range measured. So, it is possible that the initial and final values of the data set are not the same. When the FFT algorithm concatenates such a measured signal, the 'recreated' periodic signal deviates from the actual signal. This difference between the actual and recreated signals causes aberrations in the spectrum computed by the FFT as shown below; this is because the Fourier series of the recreated signal is computed, and not for the actual one.



Actual Signal



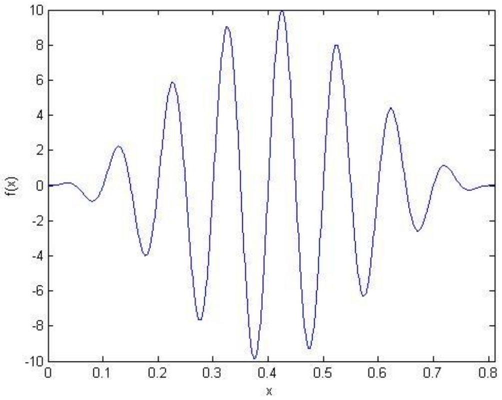
FFT of Recreated Signal



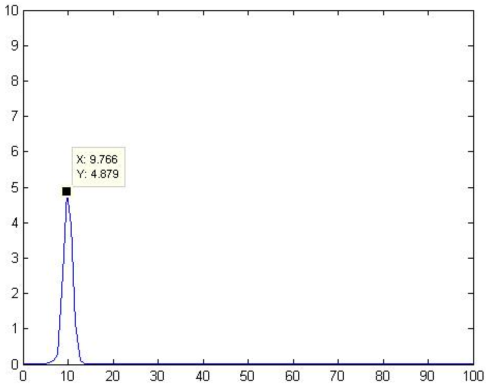
Recreated Concatenated Signal

In order to mitigate this effect, a "Windowing" operation is done which smoothens the ends of the measured data set, such that the aberrations due to concatenation are reduced/eliminated. However, such a windowing function alters the data, and thereby affects the frequency spectrum.

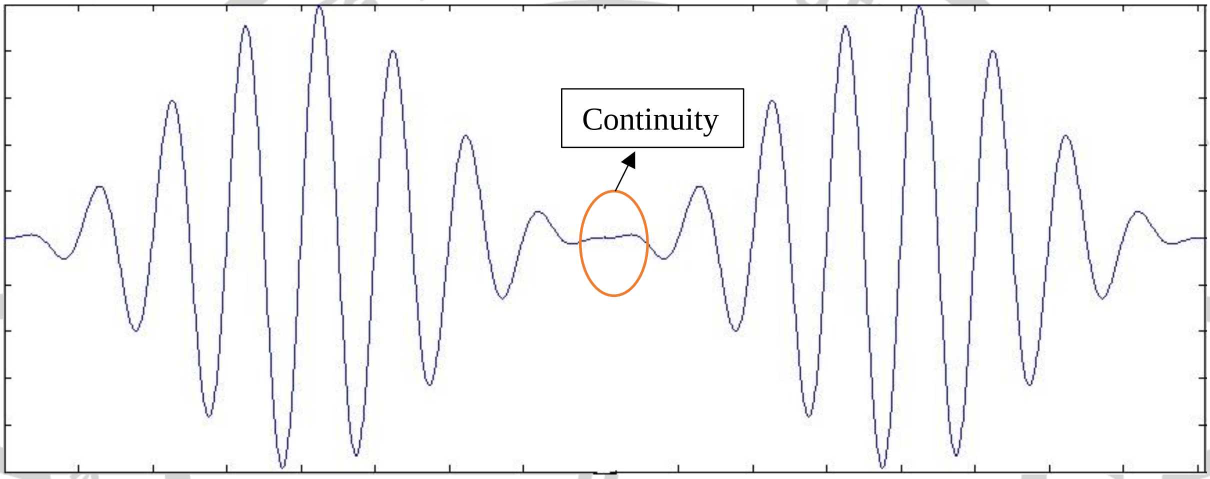
Signal is multiplied with Hanning function for demonstration. Modified signal and its FFT looks as shown below. Note that after windowing, the windowed signal on concatenation shows much lower aberrations as compared with the non-windowed ('raw') signal.



Signal after windowing



Modified FFT



Modified Periodic Signal after Windowing

Answer the following questions:

1. What changes do you observe as you alter the sampling rate?
2. How does the choice of a particular windowing function alter the FFT analysis output?

For further reading, you may refer the following:

http://www.physik.uni-wuerzburg.de/~praktiku/Anleitung/Fremde/ANO14.pdf

http://www.tmworld.com/electronics-news/4383713/Windowing-Functions-Improve-FFT-Results- Part-I

References

[1] http://www.indiana.edu/~emusic/etext/digital_audio/chapter5_rate2.shtml

[2] http://www.ni.com/white-paper/10669/en

[3] http://en.wikipedia.org/wiki/Window_function