ME370: KINEMATICS & DYNAMICS OF MACHINERY LAB

Department of Mechanical Engineering
IIT Bombay

Lab 5: Signal Processing and Analysis

Group: 5
Section: A

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1. Aim of the Experiment

- Analyzing vibration data acquired experimentally using measurement devices, including sensors and data acquisition systems.
- Get introduced to parameters which should be taken into consideration while analyzing vibration data.
- See how the acquired vibration data can be post-processed using common signal processing techniques in order to extract the required information from the measurement.

2. Apparatus Used

- MATLAB with signal processing toolbox installed
- Data provided by Prof Salil Kulkarni

3. Introduction

- Sampling Frequency Overview:
  - In digital measurement systems, data is acquired discretely rather than continuously. The rate at which these samples are taken, known as the sampling frequency, plays a crucial role in accurately reconstructing the original continuous signal. Higher sampling frequencies lead to more accurate reconstructions.

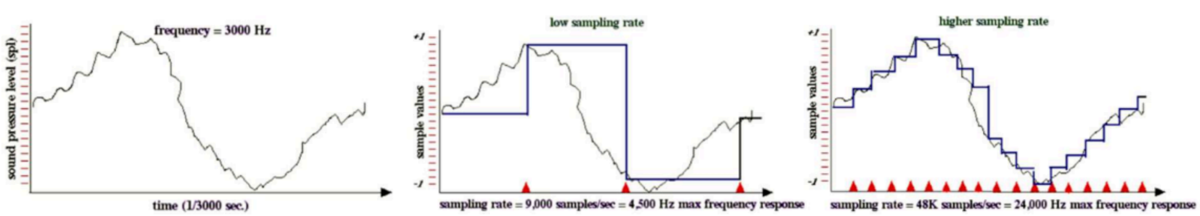


Fig. 1: A continuous signal sampled at a low frequency and high frequency to obtain a discrete output signal

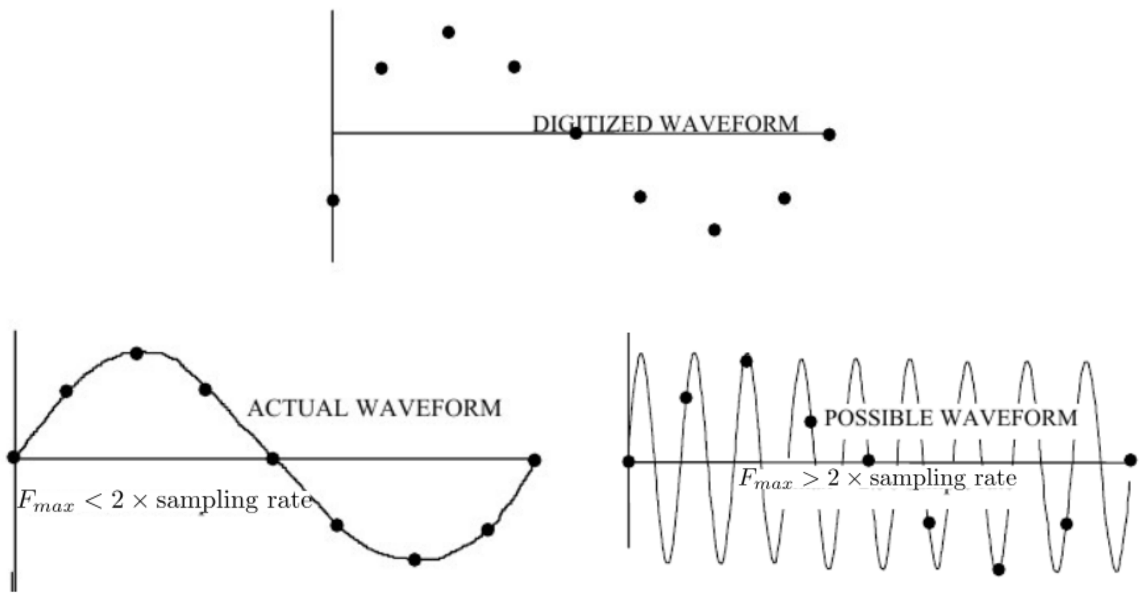
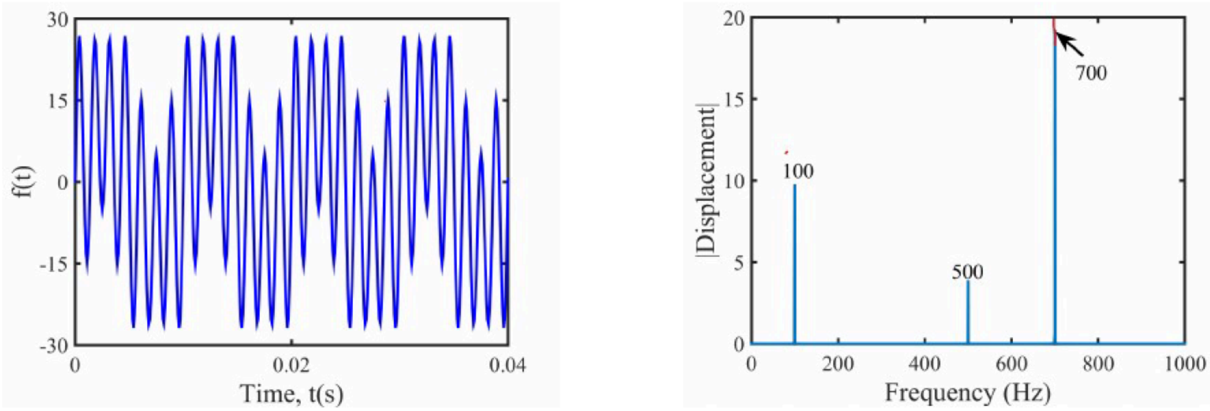


Fig 2: Illustration for the Nyquist–Shannon sampling theorem

- Aliasing and Sampling Frequency:
    - Aliasing occurs when a signal is improperly measured due to a low sampling rate. The Nyquist–Shannon sampling theorem guides determining a suitable sampling frequency, emphasizing it should be greater than twice the maximum frequency of the signal.
- Optimizing Sampling Frequency:
    - Balancing accuracy and cost, an optimal sampling rate needs to be identified. This involves selecting an initial sampling rate for signal reconstruction, incrementally increasing it until higher frequencies don't significantly improve reconstructions.
- Fast Fourier Transform (FFT) Introduction:
    - FFT is an algorithm crucial for computing discrete Fourier transforms and their inverses. Analyzing signals in the frequency domain through FFT often provides more insights than in the time domain. It enables representing periodic functions as sums of sine and cosine functions.
- Understanding FFT Output:
    - FFT output is typically displayed as a plot of amplitude versus frequencies of constituent sine and cosine functions, revealing peaks corresponding to different frequencies within the signal.



$f(t) = 10\sin(2\pi 100t) + 5\sin(2\pi 500t) + 20\sin(2\pi 700t)$

Fig 3: Working of FFT(fast fourier transform)

- Windowing Principle:
  - The assumption of signal periodicity from negative to positive infinity in Fourier analysis poses challenges with finite measured signals. Windowing is a technique used to mitigate discontinuities in reconstructed periodic signals caused by concatenation, improving FFT spectrum accuracy.

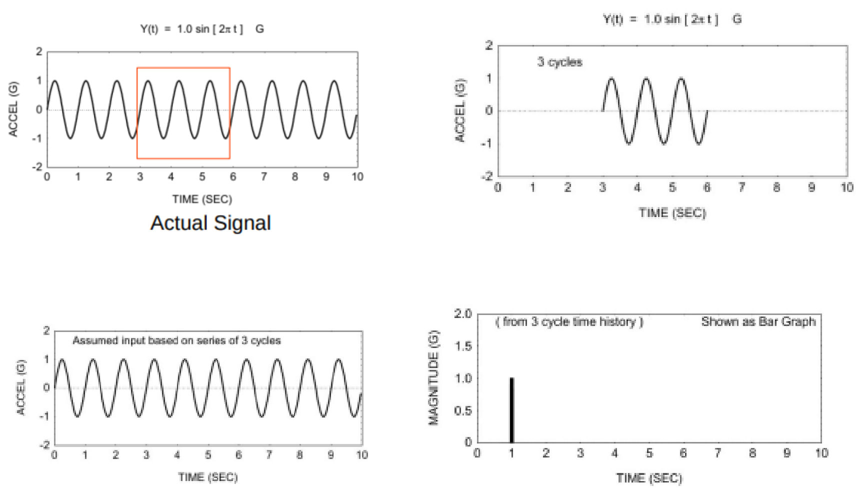
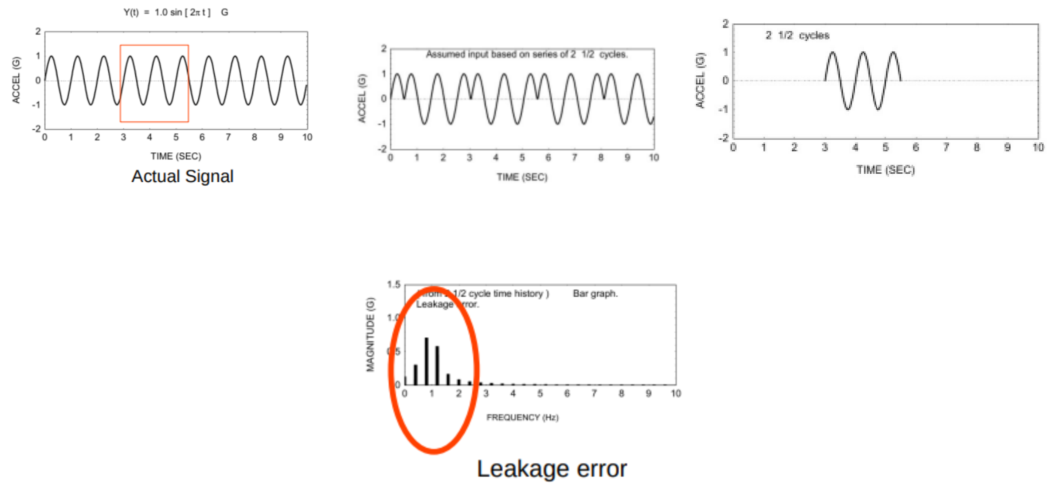


Fig 4: FFT of reproduction of a small signal



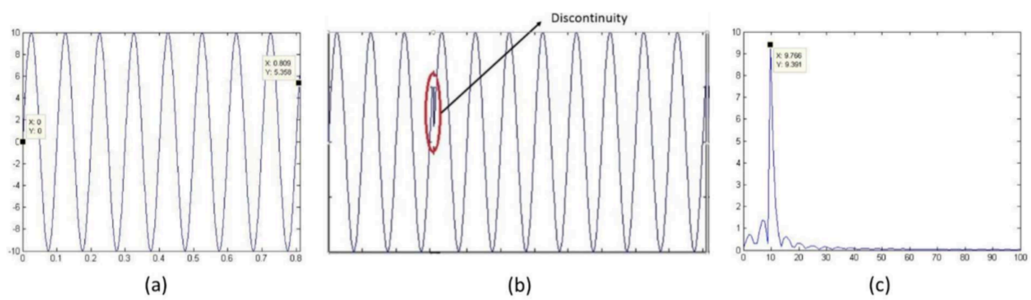


Fig. 5: (a) Measured signal with different start and end points (b) Recreated periodic signal displaying discontinuities due to concatenation (c) FFT of the reconstructed signal displaying aberrations

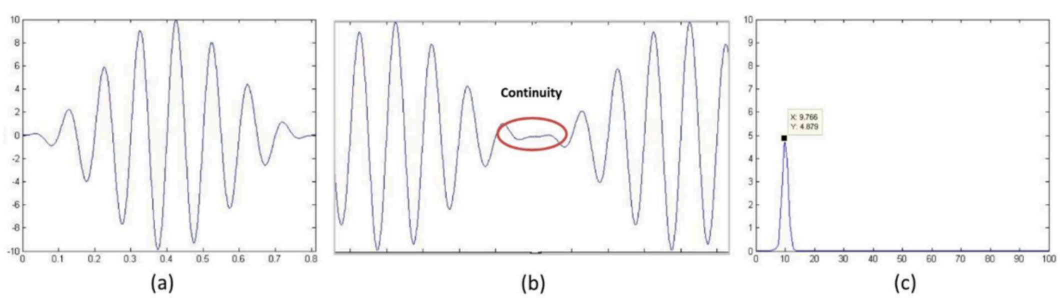


Fig. 6: (a) Windowing performed on the original measured signal in Fig. 5a (b) Recreated concatenated signal after windowing displaying no discontinuities (c) FFT of the modified reconstructed signal

- Windowing Effects and Considerations:
  - While windowing reduces aberrations in reconstructed signals, it also alters the original signal, impacting the frequency spectrum. Various

windowing functions, each with its advantages and limitations, can be chosen based on the characteristics of the input signals.

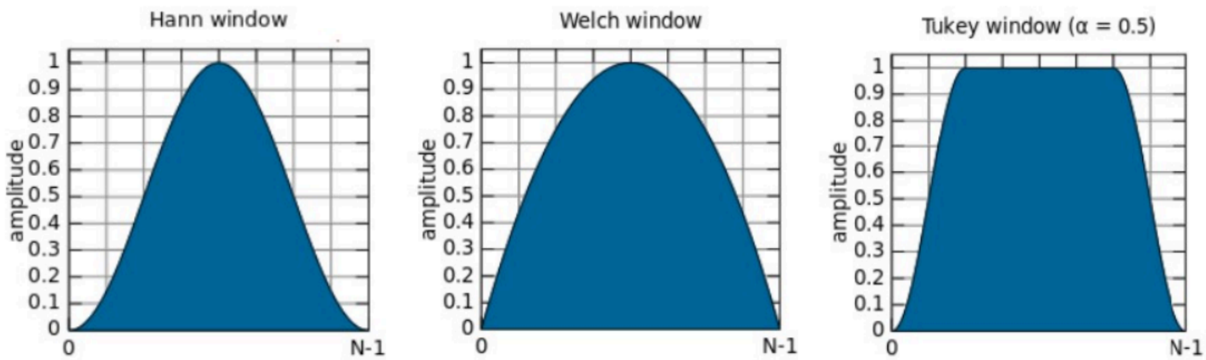


Fig 6: Various Windowing Functions

4. Methodology

- For part a, we had to generate a signal of the form $f(t) = A_1 sin(w_1 t) + A_2 sin(w_2 t) + A_3 sin(w_3 t) + A_4 sin(w_4 t)$ using MATLAB.

- The parameters w_i and A_i for i = 1, 2, 3, 4 were given as:

w_1(rads/s)	w_2(rads/s)	w_3(rads/s)	w_4(rads/s)
60	195	480	1350

A_1	A_2	A_3	A_4
200	120	350	135

Clearly, the bandwidth(highest frequency) is 1350 rads/s.

- FFT(fast Fourier Transform) of this signal was taken without windowing initially. This signal normally has losses due to ends being cut. FFT in MATLAB can be called using *fft(x)*.

- As a real acquired signal is never infinite in time, we apply windowing on the time-limited acquired signal to minimize losses. Windowing is applied to signals in order to mitigate spectral leakage, which occurs when a signal's frequency components spread into adjacent frequency bins during Fourier analysis. By tapering the signal with a window function, the abrupt transitions at the signal boundaries are softened, reducing spectral leakage and providing more accurate frequency analysis.

- We had the option to apply Hann, Hamming, Welch, Tukey windowing functions and see their effects on the frequency contents.

- Artificial Noise was added to see its effects on the FFT of the noisy signal. Adding artificial noise to a signal allows for the evaluation of the robustness and effectiveness of signal processing techniques in handling noisy environments. Filtering the noisy signal helps in isolating the original signal from the noise, enabling clearer analysis and interpretation of the signal's characteristics.

- For part b, we were given data from an accelerometer to perform signal processing and analysis.

- Applying the principles from part a, FFT of the filtered signal was computed while trying out different windowing functions.

5. **Results**

- We were able to observe the frequency content of each signal in both parts using the FFT command in MATLAB.

Part (a) –

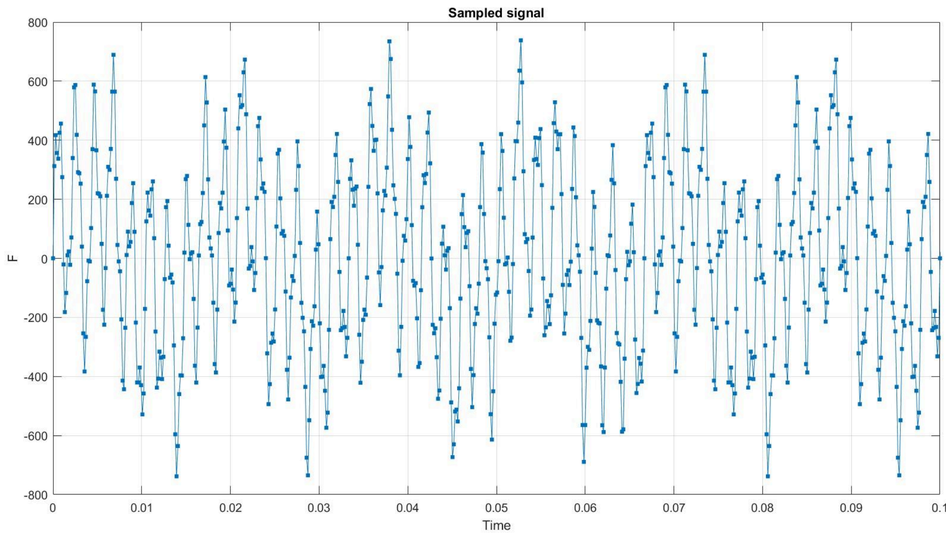


Fig 7: Sampled signal with sampling rate=10*Bandwidth

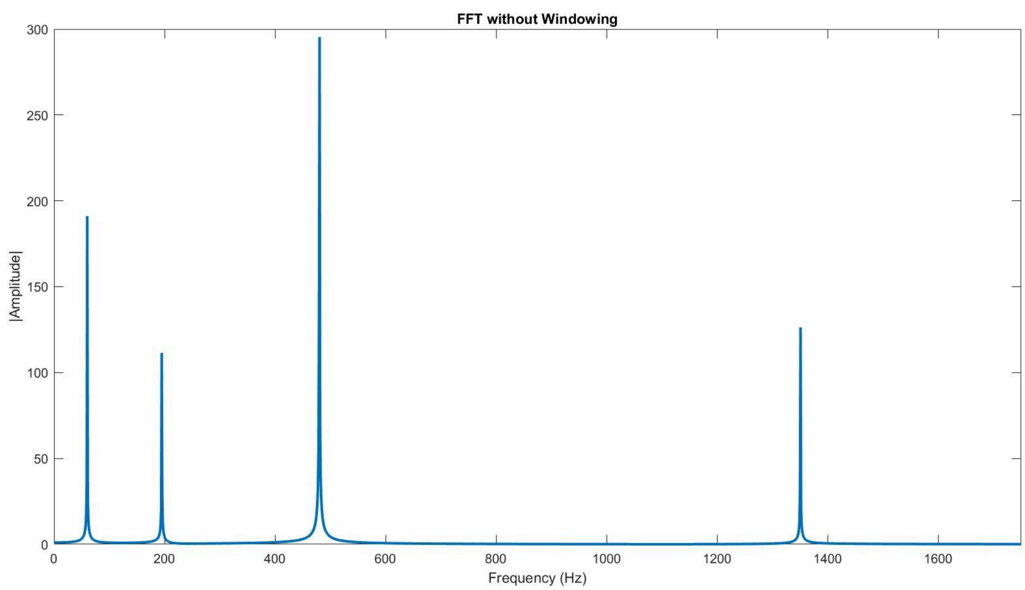


Fig 8: FFT without windowing. Losses are seen.

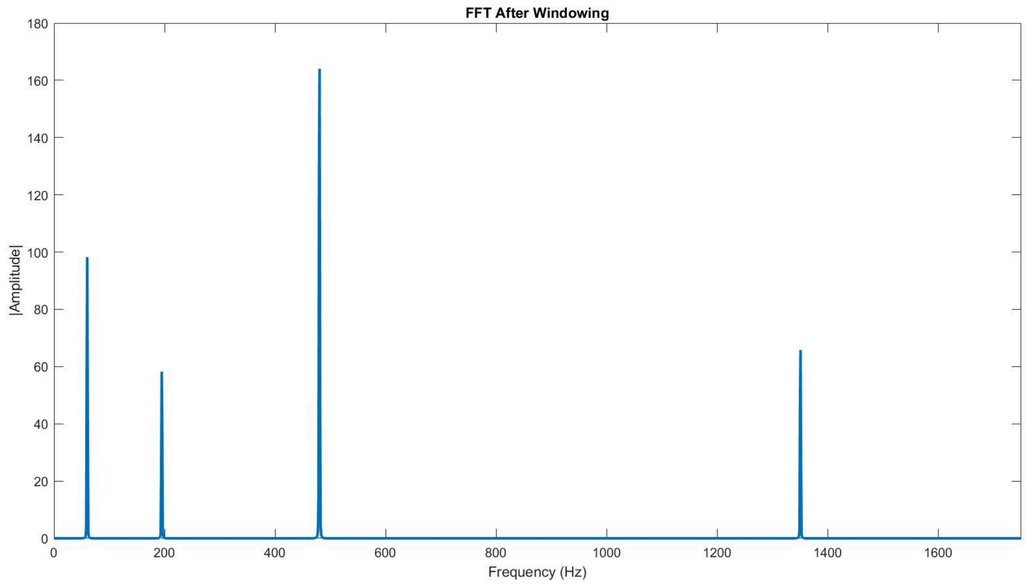


Fig 9: FFT after windowing(Hann). Losses are reduced.

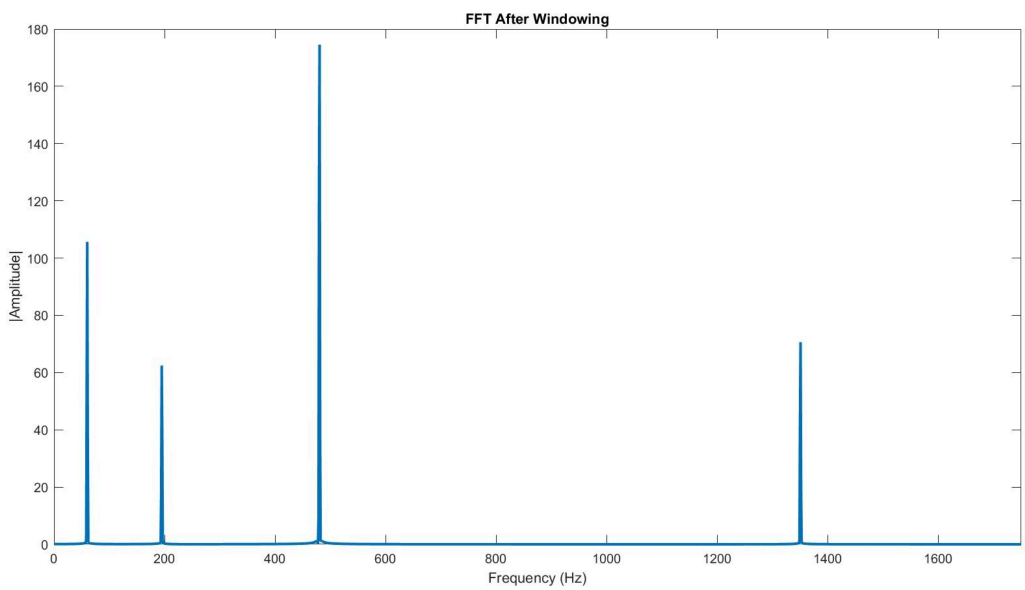


Fig 10: FFT after windowing(Hamming). Losses are reduced.

- Windowing is a necessary step as it reduces the losses in the Fourier Transform of the signal (shown in the images above).

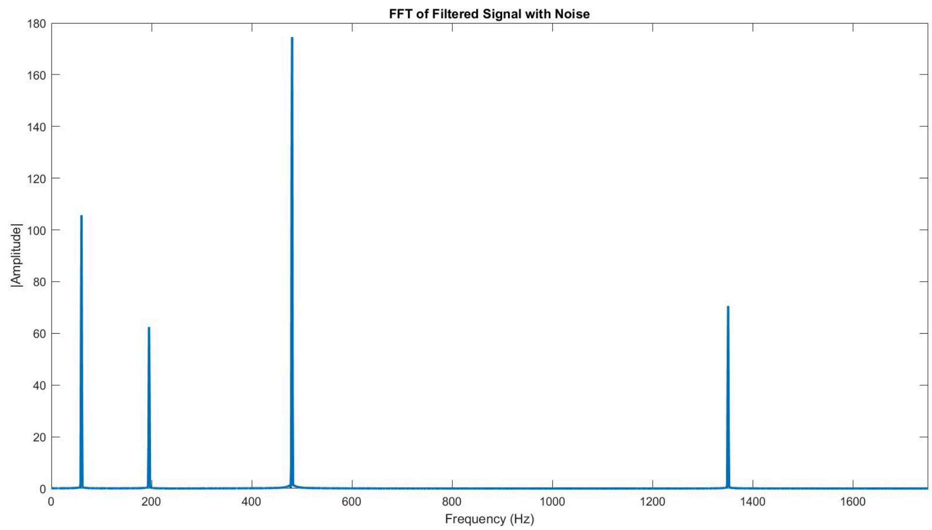


Fig 10: FFT of the filtered signal with noise

Part (b) –

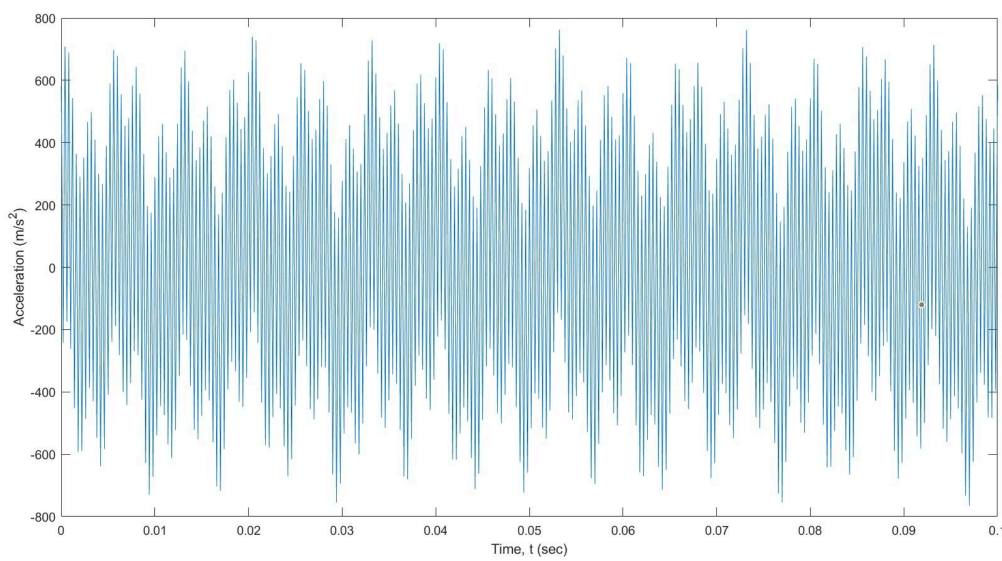


Fig 11: Acceleration vs Time from the accelerometer

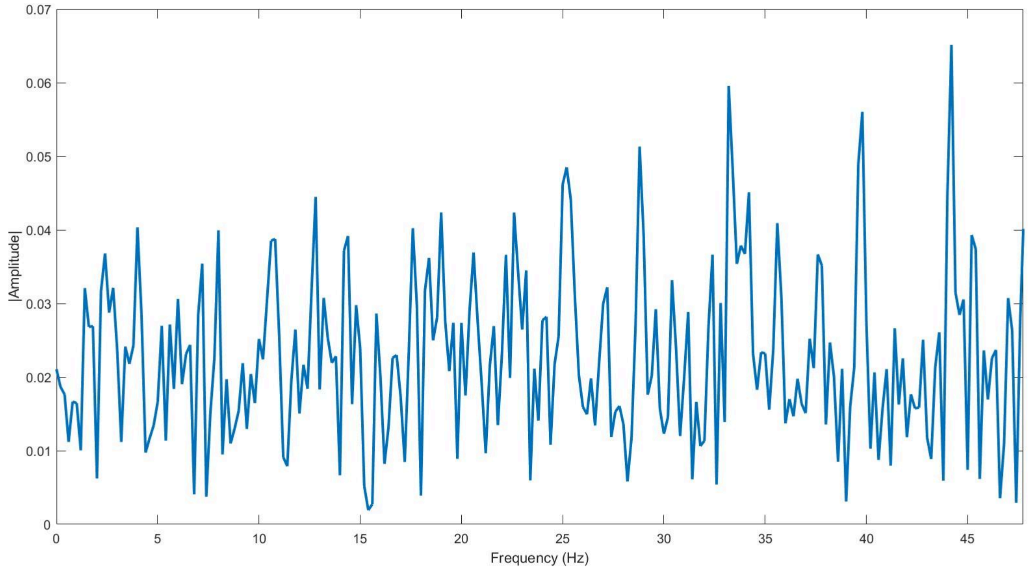


Fig 11: FFT of the filtered, windowed signal(Hann)

6. Discussion

- The windowing functions are necessary for real-time signals where frequency content has more than 4 discrete values(used in (a)). Without these, the losses would be quite large leading to errors in analysis.

- For part (a), lower sampling rates give poor reconstruction of the signal which is in accordance with the Shannon-Nyquist's sampling theorem. For best reconstruction or no aliasing, a factor greater than 5 needs to be multiplied with the highest frequency (1350 in our case).

- Out of all the windowing functions, Hamming seems to give the best frequency content as we want sharp spikes at the frequencies contained in the signal.

7. **Conclusion**

- Using these techniques, we can analyze any signal and its frequency content with great accuracy after applying required windowing functions.
- Shannon-Nyquist's Sampling Theorem is a theoretical lower bound for the sampling rate but in practice, a higher multiplying factor(5 in this case) should be chosen to avoid loss in information.
- Real world signals most likely contain noise from sensors, measurements and other disturbances. Filtering these signals thus, carries great significance to reduce its effects on the underlying signal.
- The results reflect the importance of robust signal processing algorithms in extracting meaningful information from noisy datasets.

8. **Sources of Error**

- External interference or inherent noise in the system can distort the signals, affecting the accuracy of the analysis.
- Insufficient sampling rates may lead to aliasing, where higher frequency components are incorrectly represented in the sampled signal.
- Systematic biases in the experimental setup or data processing methods can skew the results.

9. **Contributions**

Sr No.	Name	Contribution
1	Yash Salunkhe	Methodology, Results
2	Sanika Wagh	Aim, Apparatus, Introduction
3	Kavan Vavadiya	Conclusion
4	Shreya Biswas	Sources of Error
5	Samiksha Patel	Discussion
6	Mudit Sethia	Plots, Formatting

10. **Bibliography**

- https://www.sciencedirect.com/topics/computer-science/shannon-sampling-theorem
- https://en.wikipedia.org/wiki/Window_function
- ME370 slides shared by Prof. Salil Kulkarni