UNIVERSITI MALAYSIA PAHANG

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BORANG PENGESAHAN STATUS TESIS

JUDUL: <u>NOISE EXTRACTION USING FREQUENCY DOMAIN ANALYSIS</u>

SESI PENGAJIAN: <u>2011/2012</u>

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NOISE EXTRACTION USING FREQUENCY DOMAIN ANALYSIS

RIDUAN BIN ABDUL RAHMAN

Report submitted in partial fulfilment of the requirements
for the award of Bachelor of Mechanical Engineering

Faculty of Mechanical Engineering
UNIVERSITI MALAYSIA PAHANG

JUN 2012

UNIVERSITI MALAYSIA PAHANG

FACULTY OF MECHANICAL ENGINEERING

I certify that the project entitled "Noise Extraction Using Frequency Domain Analysis" is written by Riduan bin Abdul Rahman. I have examined the final copy of this project and in my opinion, it is fully adequate in terms of scope and quality for the award of the degree of Bachelor of Engineering. I herewith recommend that it be accepted in partial fulfilment of the requirements for the degree of Bachelor of Mechanical Engineering.

Examiner	Signature
MR. LEE GIOK CHUI	………………………

SUPERVISOR'S DECLARATION

I hereby declare that I have checked this project report and in my opinion this project is satisfactory in terms of scope and quality for the award of the degree of Bachelor of Mechanical Engineering.

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Date	: 25 JUNE 2012

STUDENT'S DECLARATION

I hereby declare that the work in this report is my own except for quotations and summaries which have been duly acknowledged. The report has not been accepted for any degree and is not concurrently submitted for award of other degree.

Signature :
Name : RIDUAN BIN ABDUL RAHMAN
ID Number : MA 09005
Date : 25 JUNE 2012

Dedicated to my father, Mr. Abdul Rahman bin Haji Ahmad, my beloved mother, Mrs. Salamiah binti Hamzah, my brothers Mohd Tajuddin and Mohd Nazlah, my sister Jurahana, Badriana, Nor Azimah and Ainol Mardiah, my love one and last but not least to all my fellow friends

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ABSTRACT

Noise is considered a hindrance in every vibrations signals including in an automotive suspension systems. Therefore methods of noise extraction were introduced in order to extract noise in the vibration signals. In this study noise are extracted from an automotive suspension system by frequency domain analysis. The vibrations frequency of the automotive spring is set to 8 Hz, 9 Hz and 10 Hz after that the automotive spring vibrations signals data were collected by using an accelerometer which connected to the suspension test rig which it functions were to measure the displacement of the spring, by using DASYLab® software. The vibrations signals it is then undergoes a low pass and high pass filter which is then interpreted in the form of power spectrum density which is done by fast-Fourier transforms which is then from the power spectrum density it is analyze to conduct noise extraction. The result is based on the ripple produce in power density spectrum of all the different frequency and also a different low-pass filter and high-pass filter. After that finding the most suitable frequency conditions for the low pass and high pass filter based on power spectrum density produce after filter process. The most optimum condition for noise extraction which achieved the most free noise vibration is when the low-pass filter is set to a frequency of 8 Hz and the high-pass filter is set to a frequency of 10 Hz.

ABSTRAK

Bunyi hingar dianggap sebagai penghalang dalam setiap isyarat getaran termasuk dalam sistem suspensi automotif. Oleh itu, kaedah pengekstrakan bunyi hingar telah diperkenalkan untuk mengeluarkan bunyi hingar di dalam isyarat getaran. Dalam kajian bunyi hingar ini, ia diekstrak daripada sistem suspensi automotif melalui analisis frekuensi domain. Kekerapan getaran automotif spring ditetapkan diantara 8 Hz, 9 Hz dan 10 Hz, selepas itu isyarat data getaran automotif spring dikumpulkan dengan menggunakan meter pecut yang disambungkan ke suspension test rig yang berfungsi untuk mengukur sesaran spring, dengan menggunakan perisian DASYLab®. Kemudian isyarat getaran melalui proses penapisan lulus rendah dan laluan tinggi yang kemudiannya diterjemahkan dalam bentuk ketumpatan kuasa spektrum yang dilakukan oleh jelmaan Fast Fourier, kemudian dari ketumpatan kuasa spektrum ia akan dianalisis untuk mendapatkan pengekstrakan bunyi hingar. Hasil kajian ini adalah berdasarkan kepada kekerapan dalam ketumpatan kuasa spektrum untuk kesemua frekuensi yang berbeza dan juga kepada penurasan laluan rendah dan laluan tinggi yang berbeza. Selepas itu, usaha untuk mendapatkan keadaan frekuensi yang paling sesuai untuk penapisan lulus rendah dan lulus tinggi berdasarkan kepada ketumpatan kuasa spektrum yang dihasilkan selepas melalui proses penapisan. Keadaan yang paling optimum bagi pengekstrakan bunyi hingar yang telah mencapai bunyi getaran paling bebas ialah apabila penapisan laluan rendah ditetapkan pada frekuensi 10 Hz dan turas laluan tinggi ditetapkan frekuensi 8 Hz.

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LIST OF SYMBOLS

ω	Natural frequency
CF	Crest Factor
M_3	Moment of Stage-3
M_4	Moment of Stage-4
N	Number of data
$r.m.s$	Root Means Square
SD	Standard Deviation
t	Time
$\bar{x}$	Means
x_i	Initial Value
X	Amplitude

LIST OF ABBREVIATIONS

DFT	Discrete Fourier Transform
FFT	Fast Fourier Transform
IDFT	Inverse Diverse Fourier Transform
IFFT	Inverse Fast Fourier Transform
PSD	Power Spectral Density

CHAPTER 1

INTRODUCTION

1.1. INTRODUCTION

Noise is defined as any unpleasant or unexpected sound created by a vibrating object. Noise are present in every moving object and considered a hindrance and unwanted data therefore noise are usually neglected when performing vibration analysis.

Vibration is an oscillation wherein the quantity is a parameter defining the motion of a mechanical system. Oscillation is the vibration, usually with time, of the magnitude oa a quantity with respect to a specified reference when the magnitude is alternately greater and smaller the reference. More often, vibration is undesirable, wasting energy and creating unwanted sound (noise). For example, the vibrational motions of engines, electric motors, or any mechanical device in operation are typically unwanted. Such vibrations can be caused by imbalances in the rotating parts, uneven friction, the meshing of gear teeth, etc. Careful designs usually minimize unwanted vibrations.

A signal is a real (or complex) valued function of one or more real variable(s). When the function depends on a single variable, the signal is said to be one dimensional. Signal is a series of numbers that come from measurement, typically obtained using some recording method as a function of time. A signal can be extracted from many sources such as vibrating machines, sound and movement.

Vibration signal is present in all moving object whether it is rotating or translating. The motion of a mechanical system can consist of a single component at a single frequency as with the system or it can consist of several components occurring at different frequencies simultaneously, as for example with the piston motion of an internal combustion engine. The motion signal is here split up into its separate components both in the time domain and in the frequency domain.

1.2. PROJECT BACKGROUND

Frequency domain signal analysis covers a wide variety of techniques involving the Fourier transformation of the signal. The signal's frequency domain representation is then manipulated, decomposed, segmented, classified, and interpreted. One central idea is that of a filter, that is a linear, translation-invariant system that allows one band of frequencies to appear in the output and suppresses the others. Where signal elements of interest occupy a restricted spectrum, filters invariably enter into the early processing of candidate signals. In other ways often purely theoretical frequency-domain analysis is important.

Unit of measure for frequency is called Hertz and it is equivalent to 1 cycle per second. So if the time it takes for a wave to pass is 1/2 second, the frequency is 2 per second. If it takes 1/100 of an hour, the frequency is 100 per hour. The Figure 1.1 and Figure 1.2 below show the different frequency of wave.

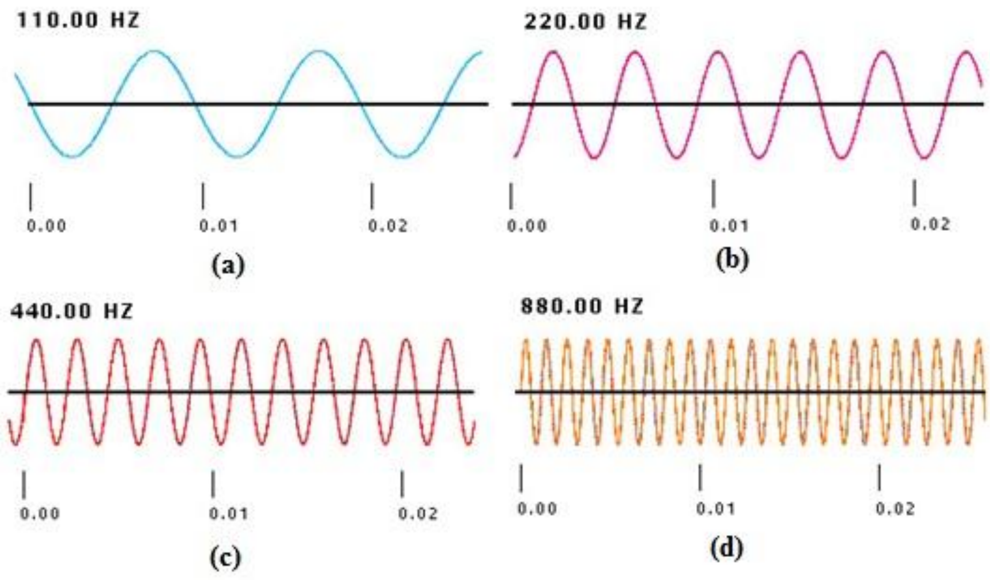


Figure 1.1: The wave of frequency. a) 110 Hz, b) 220 Hz, c) 440 Hz, d) 880 Hz

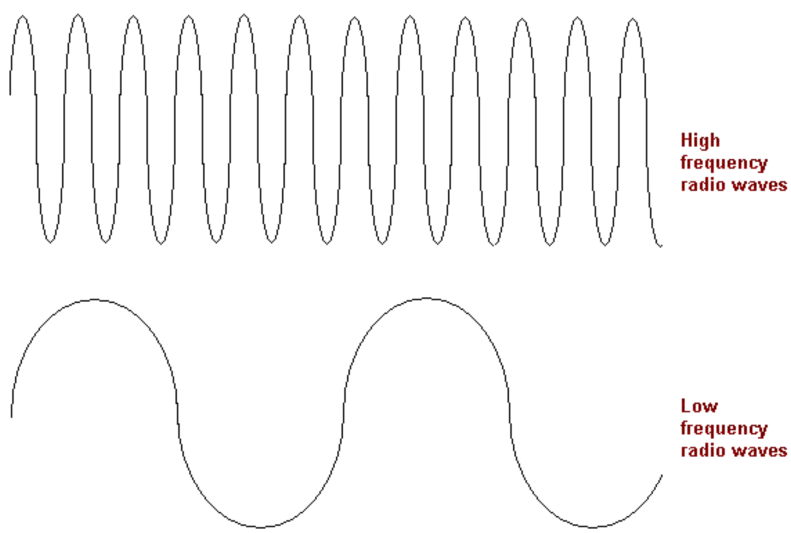


Figure 1.2: Example of the different frequency of wave

1.3. PROBLEM STATEMENT

Noise presence in any signal is the common phenomenon during recording the data. This noise can be considered unwanted data without meaning that is, data that is not being to transmit a signal, but is simply produce as an unwanted by-product of other activities. A method is known as noise extraction can be apply to eliminate noise in order to reduce analysis time, but at same time produce the similar results as the original signals. Thus, it is a critical requirement to optimize the noise value (in term of power) that should be eliminating to provide an accurate result.

1.4. OBJECTIVE

The objectives of this study are:

i. To extract noise from the vibration signals.
ii. To perform statistical analysis.

1.5. HYPOTHESIS

A time domain graph shows how a signal changes over time, whereas a frequency domain graph shows how much of the signal lies within each given frequency band over a range of frequencies. A frequency-domain representation can also include information on the phase shift that must be applied to each sinusoid in order to be able to recombine the frequency components to recover the original time signal. So, the hypothesis for this project is to use the frequency domain method to extract the noise using DASYLab® software.

1.6. SCOPE

The scopes of the project are limited to:

i. Record vibration signals at different type of frequency.
ii. Frequency domain analysis to extract noise from the signals.
iii. Optimization of the noise extraction.

CHAPTER 2

LITERATURE REVIEW

2.1. INTRODUCTION

This chapter discussed about suspension system, signals analysis, time domain analysis, frequency domain analysis and noise.

2.2. SUSPENSION SYSTEM

According to Donald Bastow *et. al.* (2004), the word suspension is the term given to the system that contains spring, shock absorbers and linkages that connects a vehicle to wheels. Suspension system isolates the people or cargo from severe levels of vibration and shock induced by the road surface. This isolation from road-induced shock and vibration is very important to improves and increase the longevity and durability of the vehicles. Figure 2.1 shows the suspension system in a vehicle body. The suspension basically includes the springs, damper and the wheel axle.

Figure 2.1: Suspension system on the vehicle

Source: Charles C. Roberts, Jr. (2005)

According to Selim Hasagasioglu *et. al.* (2011), the suspension system also enables the wheels to maintain contact with the road surface, assuring the stability and control of the vehicle because all the forces acting on the vehicle do so through the contact patches of the tires. The suspension system is an important factor in determining the comfort of a car because the suspension system is the pivot between the wheels with the weight of the car and also serves to dampen shocks and engine sound. In other words, the job of a vehicles suspension is to maximize the friction between the tires and the road surface, to give the stability of handling the vehicles and to provide the comfort of the passengers. If the road is flat with no irregularities, the suspension maybe might not be possible. But the flat road can said to be impossible. It’s means that the suspension was very important part in order to reduce the effect regarding to the flatness of the road surface. In Figure 2.2 show the basic concept of a suspension system. The suspension basically main objective is to supporting the sprung mass and the unspring mass.

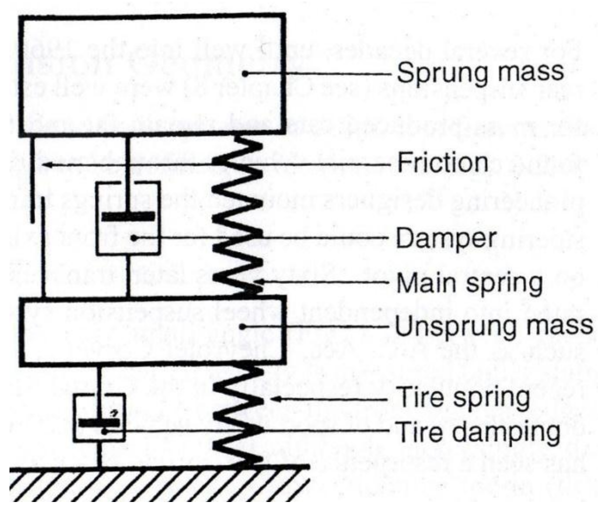


Figure 2.2: Basic elements of suspension system

Source: Donald Bastow, (2004)

A bump or subtle imperfections on the road surface causes the wheel to move up and down perpendicular to the road surface. In this situation, the vehicles can be loose handle and make the driving unsafe. This time, the suspension can play its role that ensure the tires always contact with the road surface and maintain the control over the vehicle and drive it safely. The suspension is located at the wheels of the vehicles. So, the most important thing to consider while building a suspension is the suspension is used to support a load from above such as the body of the vehicles, the loadings, the passengers and so on. The spring is what actually support the weight of the vehicle and will determine how the vehicle's weight changes when braking, acceleration and cornering.

According to Singiresu S. Rao (2004), when building a suspension, three most crucial elements must be considered. The first thing is flexibility. It is refers to designs of the suspension system that can adapt or giving the good respond to potential internal or external changes affecting its value delivery. Flexibility is given by a spring (on the suspension system) that distort and recovers (typically compress and expands) as the wheel traverses disturbances in the road surface. The second thing is damping which is essentially to restrain the body and wheel resonant bouncing motions. Damping is defined as the mechanism by which the vibrational energy is gradually converted into heat or sound. The damper also assume to have neither mass nor elasticity, and damping force exists only if there is relative velocity between the two ends of the dampers and the third one is the location of the wheel.

2.2.1. The Principle of Suspension System

According to Keith Worden *et. al.* (2008), the vehicle suspension systems basically consist of wishbones, the spring, and the shock absorber to transmit and also filter all forces between body and road. The task of the spring is to carry the body-mass and to isolate the body from road disturbances and thus contributes to drive comfort. Table 2.1 discusses the suspension component, properties (composition and position) and its function. The damper contributes to both driving safety and comfort. Its task is the damping of body and wheel oscillations, where the avoidance of wheel oscillations directly refers to drive safety, as a non-bouncing wheel is the condition for transferring road-contact forces. Considering the vertical dynamics and taking into account the vehicle's symmetry, a suspension can in a first step be reduced to the so-called quarter-car model as shown in Figure 2.3. Here, elements for modeling the Coulomb friction and an additional force resulting from active or semi-active components are added. The tire is typically modeled by a single spring.

The terms of driving safety and comfort are defined. Driving safety is the result of a harmonious suspension design in terms of wheel suspension, springing, steering and braking, and is reflected in an optimal dynamic behavior of the vehicle, whereas driving comfort results from keeping the physiological stress that the vehicle occupants are subjected to by vibrations, noise, and climatic conditions down to as low a level as possible. It is a significant factor in reducing the possibility of miss actions in traffic. Typically, the acceleration of the body as an obvious quantity for the motion and vibration of the car body and the tire load variation as indicator for the road contact are used for determining quantitative values for driving comfort and safety, respectively.

Table 2.1: The examples of components of automotive system

Component	Properties	Function
Shockbreaker	This component is made of steel so that it has more endurance and strength. However Shock breaker seriring will wear with time or usage period and the use of inappropriate.	These devices served to absorb shocks when the car drove and bulldoze variety of track conditions. Shock breaker made of steel that assists the spring or as to support the weight of the car following the charge that he had plundered.
Arm bushing	This form of rubber suspension components is located at the fulcrum between the wheels and arms clamp.	Bushing duties dampen vibration at the connection between the components of the suspension of the metal. When the car is often bulldoze the streets potholes or broken street, which is sustained load device, is also increasingly heavy.
Tierod and balljoint	Tierod, tierod end and balljoint is made of metal material	Tierod has a continuing function of the steering wheel turning force to the wheels. While balljoint useful to sustain Knuckle arm.

Source: Recent patents on mechanical engineering, (2008)

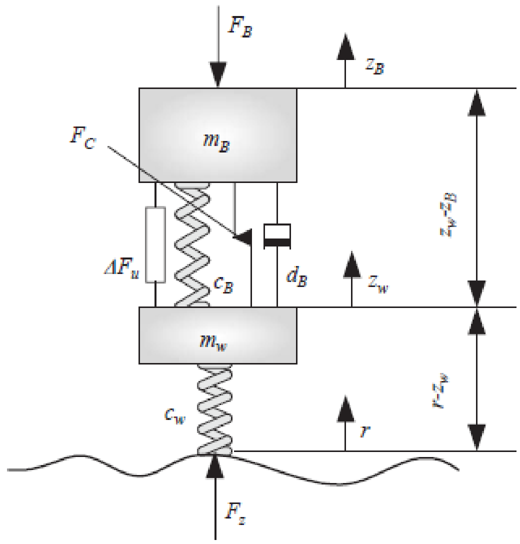


Figure 2.3: One-Dimensional Vertical Vehicle Representation the Quarter Car Model

Source: Shaohua, (2003)

2.3. SIGNALS

According to A.G. Ambekar (2006), signal is a series of numbers that come from measurement, typically obtained using some recording method as a function of time. A signal can be extracted from many sources such as vibrating machines, sound and movement. All these cause unbalance, misalignment, and looseness, dry friction between two rubbing surface, wind-induced vibration (self-induced vibration), oil whirl and external excitation. Some wavelength as contains high frequency and some contains low frequency waveform. The signal we have from experiment have disturbance such as noise.

2.3.1. Signal Characteristic

Signal characteristic always perform in vibration signal, below this show some of characteristic of signal in vibration.

i. Some features have a long time duration but narrow bandwidth, for example, rub & buzz noise.
ii. Some features have short time duration but wide bandwidth, for example, spikes and breakdown points.
iii. Some features have a short time duration and narrow bandwidth, for example, decayed resonance.
iv. Some features might have a time-varying bandwidth, for example, the imbalance bearing generating noise dependent on RPM.

2.3.2. Types of Signals

Signal analysis is fundamental to vibration testing. Consequently, understanding it and its proper use should be high priority to any practioner. Dynamic signal from a data analysis viewpoint, divide time history signals into two broad categories, each with two subcategories which are:-

a. Deterministic data signals:-
   i. Steady-state or periodic signals.
   ii. Transient signals.

b. Random data signals
   i. Stationary signals.
   ii. Non-stationary signals.

According to John Wiley & Sons (1987), the chaotic signal is recently recognized phenomenon where a random appearing signal is controlled by a deterministic process. Chaotic signals are receiving more attention in an effort and analyze them. Just how this research will impact future signal classification is not clear at this time; thus, the question marks in the diagram. However, it must be recognized that some random appearing signal analysis purpose until ways are found to clearly distinguish chaotic signal analysis purpose until ways are found to clearly distinguish chaotic signal from random signals. Chaotic signal are not considered in this book beyond cursory reference to them. In Figure 2.4 show the dynamic signals are generally classified as deterministic and random.

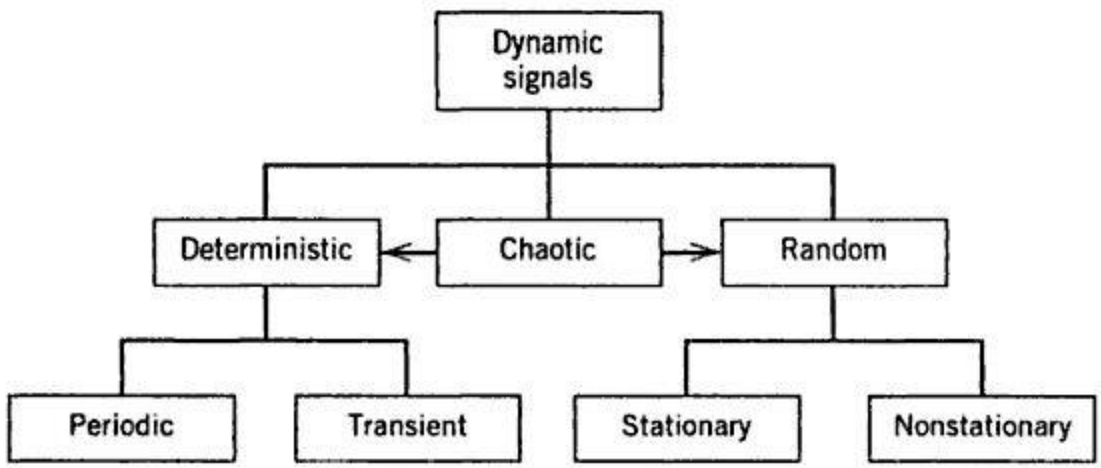


Figure 2.4: Dynamic signal classification

Source: Vibration testing theory and practice, (1995)

2.3.3. Deterministic Data

Deterministic data signals are those for which one can, in theory, predict future time history values of the signal (within reasonable error) based upon a knowledge of the applicable physics or past observations of the signal. Deterministic signal are further classified as either periodic or transient. A periodic signal is one that repeats itself in time and is a reasonable model for many real processes, especially those associated with constant speed machinery.

A transient signal is one that has no significant variation occurring for long periods of time with short periods of intense activity. Ideally, infinite time occurs before and after the transient event. In practical terms, it is necessary only that all vibration ceases before another event occurs for a signal to be classified as a transient.

2.3.4. Random Data

Random signals are characterized by having many frequency components present over a wide range of frequencies. The amplitude versus time appears to vary rapidly and unsteadily with time.

Stationary signal are ones that have constant parameters to describe their behavior, while non-stationary signal have time dependent parameters. This is easily seen when one thinks of engine exited vibration where the engine's speed varies with time; in this case, the fundamental period changes with time as well as with the corresponding dynamic loads that cause vibration.

Random vibrations are broadly defined as those that are not deterministic, that is, where it is not theoretically feasible to predict future time history values based upon knowledge of the applicable physics or past observations.

2.3.5. Signals Analysis

According to Burrus, C. S. and Parks, T. W., (1985), in signal analysis, we determine the response of a system under a known excitation and present it in a convenient form. Often, the time response of a system will not give much useful information. However, the frequency response will show one or more discrete frequencies around which the energy in concentrated. Since the dynamic characteristics of individual components of system are usually known, we can relate the distinct frequency components (of the frequency response) to specific components.

2.3.6. Statistical Analysis

Global signal statistics are frequently used to classify random signals. The most commonly used statistical parameters are the mean value, the standard deviation value, the root-mean-square (r.m.s.) value, the skewness, the kurtosis and the crest factor (Hinton 1995). For a signal with a number n of data points in the sampled sequence$n$, the mean value $\bar{x}$ is given by

$$\bar{x} = \frac{1}{n}\sum_{i=1}^{n} x_i \tag{2.1}$$

The standard deviation *(SD)* is mathematically defined as

$$SD = \left\{\frac{1}{n}\sum_{j=1}^{n}(x_i - \bar{x})^2\right\}^{\frac{1}{2}} \tag{2.2}$$

for data of more than 30 samples. If the data is less than 30 samples, standard deviation is given by

$$SD = \left\{\frac{1}{n-1}\sum_{j=1}^{n}(x_i - \bar{x})^2\right\}^{\frac{1}{2}} \tag{2.3}$$

The standard deviation value measures the spread of the data about the mean value. The r.m.s. value, which is the 2nd statistical moment, is used to quantify the overall energy content of the signal. For discrete data set, the r.m.s value is given by Equation 2.4. of the signal have an average is zero, the rms value is equal to the value of standard deviation.

$$r.m.s = \left\{\frac{1}{n}\sum_{j=1}^{n} x_i{}^2\right\} \tag{2.4}$$

The moment the average value of r at $\bar{x}$ for discrete data values are given by Equation 2.5

$$M_r = \left\{\frac{1}{n}\sum_{i=1}^{n}(x_i - \bar{x})^r\right\} \tag{2.5}$$

where *n* is the number of data and *r* is the moment. Skewness *(S)*. Where is the moment that the three statistics used to measure the symmetry of the distribution data on the average. Skewness of a signal is given by

$$S = \frac{1}{n(SD)^3}\sum_{j=1}^{n}(x_i - \bar{x})^3 \tag{2.6}$$

easily it is written as

$$S = \frac{M_3}{(SD)^3} \tag{2.7}$$

where M_3 is the moment of stage-3 and $(SD)^3$ are the three standard deviation. Skewness of the distribution of symmetry, such as sinusoidal or Gaussian random signal is equal to zero. Negative skewness value indicates the probability distribution is skewed to the left while a positive value of skewness shows the distribution is skewed to the right, which refers to the value average.

Kurtosis which is the statistical moments of the four is the statistical sensitive to the distribution of high or transient data. For discrete data set, the value kurtosis is given by

$$K = \frac{1}{n(SD)^4} \sum_{i=1}^{n} (x_i - \bar{x})^4 \tag{2.8}$$

and is easy it is written as equation 2.9 (Nuawi et al., 2007).

$$K = \frac{M_4}{(SD)^4} \tag{2.9}$$

the M_4 is a moment of stage-4 and $(SD)^4$ is the power of four standard deviations. For signals with a Gaussian distribution or normal distribution, the kurtosis is close to 3.0. This value indicates the signal is stationary. High kurtosis value indicate the presence of the signal amplitude is too high or extreme compared signal amplitude found in the Gaussian distribution which represents a signal is not stationary. Kurtosis value is used to detect damage in engineering due to the statistical parameters are sensitive to signals a high amplitude (Qu & He 1986).

The peak factor which is the usual statistical parameters found in engineering applications, defined as the ratio between the maximum value in time domain and the r.m.s.

$$CF = \left|\frac{x_{j\,max}}{r.m.s}\right| \tag{2.10}$$

2.3.7. Spectrum Analyzers

Spectrum of frequency analyzers can be used for signal analysis. A spectrum or frequency analyzers is a device that analyzes a signal in frequency domain by separating the energy of the signal into various frequency band. The separation of signal energy into frequency bands is accomplished through a set of filters. The analyzers are usually classified according to the type of filter employed.

In recent years, the digital analyzers have become quite popular for real-time signal analysis. In the real-time frequency analysis, the signal is continuously analyzed over the entire on the frequency band. Thus the calculation process not takes more time than the time taken to collect the signal data. Real-time analyzers are especially useful for machinery health monitoring since a change in to the noise or vibration spectrum can observe at the same time that change in the machine occurs. There are two types of real-time analysis procedure: the digital filtering method and the Fast Fourier Transform (FFT) method which will be discussed further afterwards.

2.3.8. Time Domain Analysis

The traditional way of observing signals is to view them in the time domain. The time domain is a record of what happened to a parameter of the system versus time. An example of time response is the displacement of the mass of the spring-mass system versus time. Typically when we investigate the performance of a dynamic system we use as the input to the system a step input. The resulting graph is a record of the displacement of the mass versus time, a time domain view of displacement. A simple way to express time domain analysis is by getting the simple harmonic motion equation. A body is said to execute simple harmonic motion when it oscillates about a mean equilibrium position with its acceleration always directed towards and proportional to the displacement from mean equilibrium position (F.C. Moon et al 1987).

The movement of a spring upward and downward will trace out sinusoidal patterns as a function of time that will generates a harmonic motions as shown in Figure 2.5. Furthermore if the damping effect is neglected it will become a continuous simple harmonic motion as shown in Figure 2.6, and if considering the damping effect therefore the sinusoidal wave will experienced an exponential decayed such as in Figure 2.7**.**

i. The Simple Harmonic Motion Equation

$$x(t) = X\,sin\,(\omega t + \theta) \qquad (2.11)$$

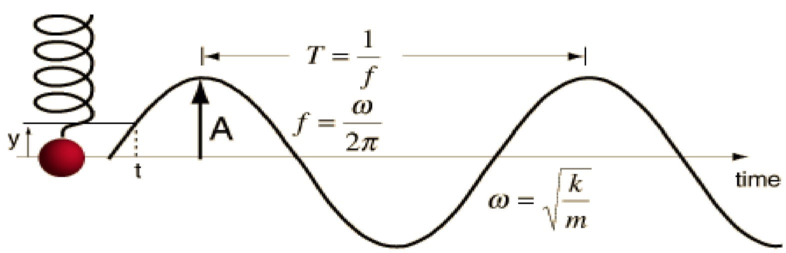


Figure 2.5: Experimental demonstration of simple harmonic motion

Source: Single degree of freedom system, (2006)

ii. Harmonic without damping

$$m\ddot{x} \; + kx \qquad (2.12)$$

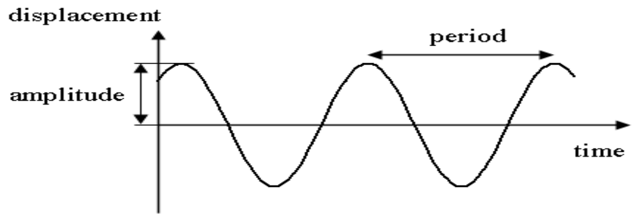


Figure 2.6: Simple harmonic motion without damping

iii. Harmonic with damping

$$m\ddot{x} + c\dot{x} + kx \tag{2.13}$$

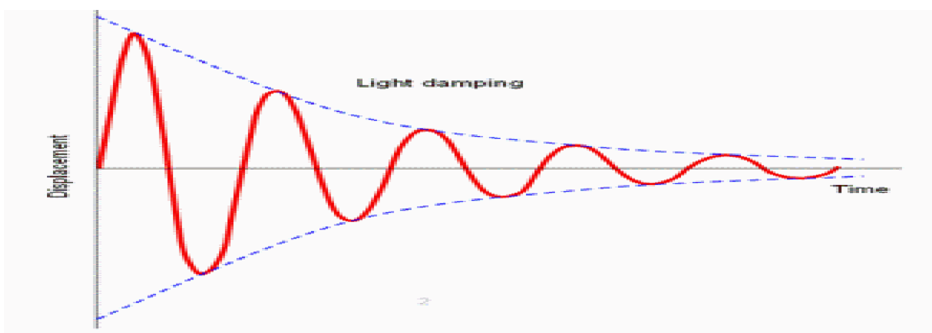


Figure 2.7: Simple harmonic motion with damping

2.3.9. Frequency Domain Analysis

According to J. W. Cooley and J. W. Tukey (1965), frequency analysis is performed in order to convert a time domain signal into the frequency domain. The results of a frequency analysis are most commonly presented by means of graph having frequency on the x-axis and amplitude on the y-axis. The algorithm that is used to split the time history into its constituent sinusoidal components is the Fourier transform. This transform was first defined by the French mathematician and engineer who postulated that any periodic function could be expressed as the summation of sinusoidal waves of varying frequency, amplitude and phase.

The spectral analysis is the understanding of a signal by examining the amplitude, frequency and phase of its component sinusoids. For a periodic time function, *x (t)*, frequency analysis can be performed using the classical Fourier transform defined by the mathematical definition. All signals have a frequency domain representation such as that any real world waveform can be generated by the addition of sinusoidal waves in Figure 2.8. The following diagram shows an example of this process:

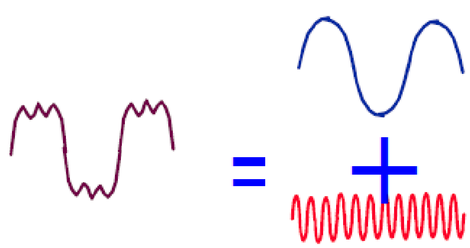


Figure 2.8: Real world waveform can be generated by the addition of sinusoidal waves.

According to Gabrielli C *et. al.* (1990), signals can be transformed between the time and the frequency domain through various transforms. The signals can be processed within these domains and each process in one domain has a corollary in the other, as shown Figure 2.9.

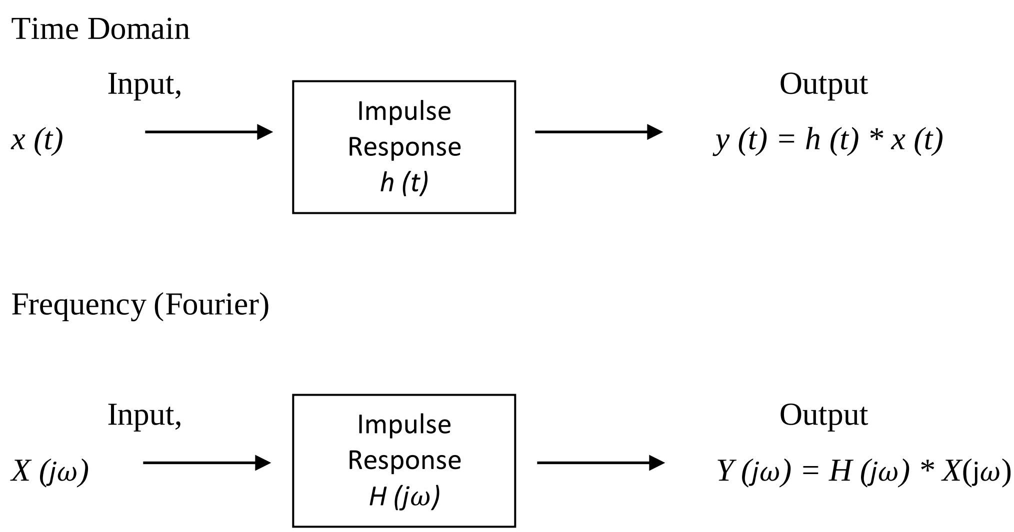


Figure 2.9: Transformation of signal

Source: Principles, algorithms and applications, (1992) The highest frequency will be captured and displayed by the instrument as digitized time waveform to produce the FFT in a series of points as shown in Figure 2.10.

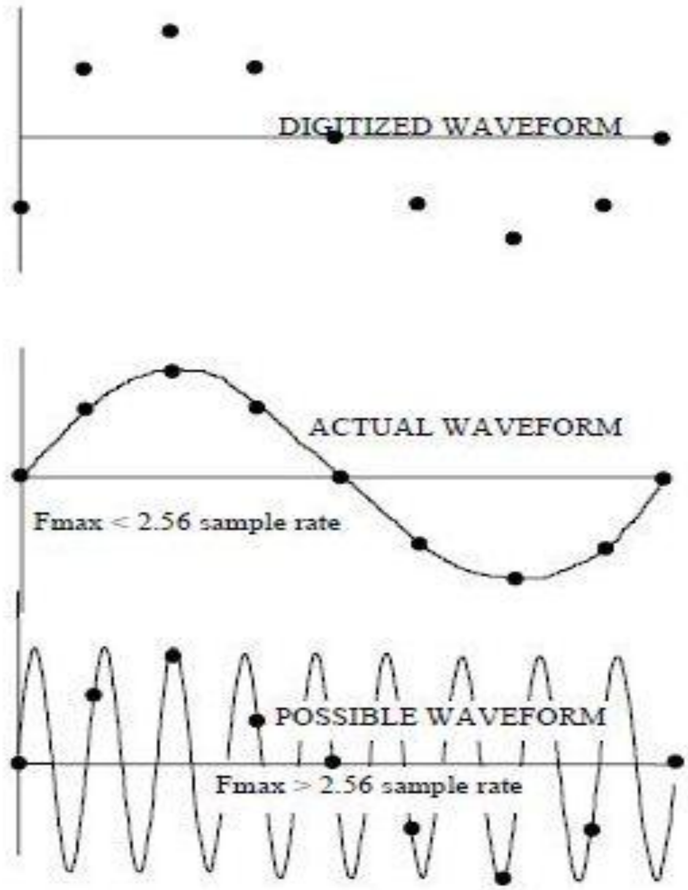


Figure 2.10: Digital sampling and analyzing

Source: Signal processing for effective vibration analysis, (1995)

2.3.10. Discrete Fourier Transform (DFT)

The Fourier transform relates the continuous-time representation *x (t)* to the continuous-frequency representation *X (f).* For processing on digital computers, it is necessary to have a similar transform that relates the discrete-time representation *x (n)* to the discrete-frequency representation *X (k).* The resulting transformation is known as the discrete Fourier transforms (DFT).

When computing spectra on a computer it is not possible to carry out the integrals involved in the continuous time Fourier transform. Instead a related transform called the discrete Fourier transform is used.

The discrete Fourier transform (also known as the finite Fourier transforms) relates two finite sequences of length *N*. Given a sequence with components *x[k]* for *k = 0, 1.N-1*, the discrete Fourier transform of this sequence is a sequence *X[r]* for *r = 0, 1… N-1*

The fast Fourier transform is defined by:

$$F(s) \equiv \int_{-\infty}^{\infty} f(x) e^{-2\pi i s x}\, dx \qquad (2.14)$$

Notice that we shall adopt the definition in which there is a factor of *1=N* in front of the forward transform. Other conventions place this factor in front of the inverse transform (this is used by MATLAB®) or a factor of *1=pN* in front of both transforms. But as in order to have a faster DFT calculation of the time series FFT was introduced (J. W. Cooley and J. W. Tukey, 1965).

2.3.11. Fast Fourier Transform (FFT)

According to A.G. Ambekar (2006), discrete Fourier transforms is used when calculating the discrete Fourier transforms which involves *O (N logN)* operations. This is known as the fast Fourier transform (FFT) algorithm. The discrete Fourier transform of a sequence of *N* points requires *O (N2)* arithmetic operations to compute if a straightforward implementation of its definition is carried out. For large N, this can become prohibitive.

Many variants of the FFT algorithm exist. We shall discuss the simplest form known as the decimation in time algorithm. The central insight which leads to this algorithm is the realization that a discrete Fourier transform of a sequence of *N* points can be written in terms of two discrete Fourier transforms of length *N = 2*. Thus if *N* is a power of two, it is possible to recursively apply this decomposition until we are left with discrete Fourier transforms of single points. Consider an unnormalized discrete Fourier transform of *N* points which we can write as

$$X[r] \equiv \sum_{k=0}^{N-1} x[k]\, W_N^{-rk} \qquad (2.15)$$

The Fourier transform is a reversible, linear transform with many important properties. For any function *f(x)* (which in astronomy is usually real-valued, but *f(x)* may be complex), the Fourier transform can be denoted *F(s),* where the product of *x* and s is dimensionless. Often *x* is a measure of time *t* (i.e., the time-domain signal) and so s corresponds to inverse time, or frequency (i.e., the frequency-domain signal).

$$X(s) \equiv \int_{-\infty}^{\infty} f(x) e^{-2\pi i s x}\, dt \qquad (2.16)$$

2.3.12. Power Spectral Density (PSD)

Power spectral density function (PSD) shows the strength of the variations (energy) as a function of frequency. In other words, it shows at which frequencies variations are strong and at which frequencies variations are weak. The unit of PSD is energy per frequency (width) and you can obtain energy within a specific frequency range by integrating PSD within that frequency range. Computation of PSD is done directly by the method called FFT or computing autocorrelation function and then transforming it.

PSD is a very useful tool if you want to identify oscillatory signals in your time series data and want to know their amplitude. For example let assume you are operating a factory with many machines and some of them have motors inside. You detect unwanted vibrations from somewhere. You might be able to get a clue to locate offending machines by looking at PSD which would give you frequencies of vibrations.

PSD is still useful even if data do not contain any purely oscillatory signals. For example, if you have sales data from an ice-cream parlor, you can get rough estimate of summer sales peak by looking at PDF of your data. We quite often compute and plot PSD to get a "feel" of data at an early stage of time series analysis. Looking at PSD is like looking at simple time series plot except that we look at time series as a function of frequency instead of time. Here, we could say that frequency is a transformation of time and looking at variations in frequency domain is just another way to look at variations of time series data. PSD tells us at which frequency ranges variations are strong and that might be quite useful for further analysis.

In a real-word application, one would typically average this single-measurement PSD over several repetitions of the measurement to obtain a more accurate estimate of the real PSD underlying the observed physical process. This computed PSD is sometimes called periodogram. One can prove that this periodigram converges to the true PSD when the averaging time interval T goes to infinity to approach the Power Spectral Density (PSD). (Robert Grover *et. al.* 1997)

The equation of PSD

$$x(t) = \frac{1}{2\pi} \int_{\infty}^{\infty} S_x\ (\omega) d\omega \qquad (2.17)$$

Equation of PSD (cumulative power spectrum)

$$S_x = \lim_{T \to 1} \frac{|F_x\ (\omega)|^2}{T} \qquad (2.18)$$

2.3.13. Inverse Diverse Fourier Transform (IDFT)

Since the DFT is invertible, it is possible to synthesize any desired time domain waveform by constructing the complex spectrum of the desired signal and applying the inverse DFT. This is usually known as the forward transform

$$f(s) \equiv \int_{-\infty}^{\infty} F(x) e^{-2\pi i s x} ds \qquad (2.19)$$

2.3.14. Inverse Fast Fourier Transform (IFFT)

Finding IFFT from FFT is complex. From FFT we synthesize IFFT without considering the time domain (usually combine with frequency domain)

$$f(x) \equiv \int_{-\infty}^{\infty} X(s) e^{2\pi i s x}\, dt \qquad (2.20)$$

2.3.15. Advantages

According to Boukamp BA (2004), a large frequency domain can be achieved, from mHz to GHz and higher, with high accuracy. Data processing is easier since ordinary processes are involved instead of convolution in the time domain. Powerful data analysis programs are available (e.g., complex non-linear least square fit or CNLS-fit). Error estimations for model parameters can be obtained. More visual information on various contributing processes is provided. Furthermore, often the time response of a system will not give much useful information. However, the frequency response will show one or more discrete frequencies around which the energy in concentrated.

2.4. NOISE

The 19th century physicist Hermann von Helmholtz employed the term 'noise' to describe sound composed of nonperodic vibrations (e.g. the rustling of leaves), consist of periodic vibrations. Noise is defined as any unpleasant or unexpected sound created by a vibrating object. Noise are present in every moving object and considered a hindrance and unwanted data therefore noise are usually neglected when performing vibration analysis. The result of noise present in a machine will result in reduction in machine working performance thus which include vibratory system machine such as automotive suspension system. Although noise cannot be fully suppress but it can be extracted and a counteractive measure can be taken to reduce the noise. Noise has it unique frequencies which can be filtered which allow the noise to be extracted by allowing the noise to undergo a filter process.

2.4.1. Narrow Band

The most basic frequency-domain analysis task involves detecting and interpreting more or less isolated periodic components of signals. For some signals, or at least for some part of their domain, a few oscillatory components contain the bulk of the energy. It is such narrowband signals sinusoids (tones) and dual-tones, mainly, but we could also allow complex-valued exponentials into this category, that we begin our study of Fourier transform applications. Narrowband signals can be distinguished from wideband signals, where the energy is spread over many frequencies. A signal that contains sharp edges, for example, will generally have frequency-domain energy dispersed across a wide spectral range.

Although basic, a tone detection application leads to important practical concepts: noise removal, filtering, phase delay, group delay, and windowing. Filters are frequency-selective linear translation invariant systems. Filtering a signal can change the time location of frequency components.

2.4.2. White Noise

White noise is a sound that contains every frequency within the range of human hearing (generally from 20 hertz to 20 kHz) in equal amounts. Most people perceive this sound as having more high-frequency content than low, but this is not the case. This perception occurs because each successive octave has twice as many frequencies as the one preceding it. For example, from 100 Hz to 200 Hz, there are one hundred discrete frequencies. In the next octave (from 200 Hz to 400 Hz), there are two hundred frequencies.

White noise can be generated on a sound synthesizer. Sound designers can use this sound, with some processing and filtering, to create a multitude of effects such as wind, surf, space whooshes, and rumbles.

2.4.3. Bandwidth reduction

Reducing the system noise bandwidth *(B*n*)* by reducing noise. This approach works well if the frequency spectra of the noise and signal do not overlap significantly, so that reducing the noise bandwidth does not affect the signal. With random white noise, the output noise is proportional to $\sqrt{B\text{n}}$. with nonwhite noise, other relationships will apply.

2.4.4. Averaging or integrating techniques

Successive samples of the signal are synchronized and added together. The signal will grow as the number *(n)* of added samples; with random white noise, the noise will grow as $\sqrt{n}$. This is only the case, if the signal characteristics are stationary for the duration of the extraction process.

2.5. FILTERS IN THE FIELD OF VIBRATION

Vibration signal from a transducer requires signal processing to produce the data that we need. The process usually involves filtering, setting sample rates and resolution, windowing, etc. It is important to understand what filters do and how they are used in the field of vibration.

Filtering is a process that removes some frequencies from a signal in order to suppress interfering frequencies and reduce noise. Refer to Figure 2.11 show the types of commonly used filters in vibration.

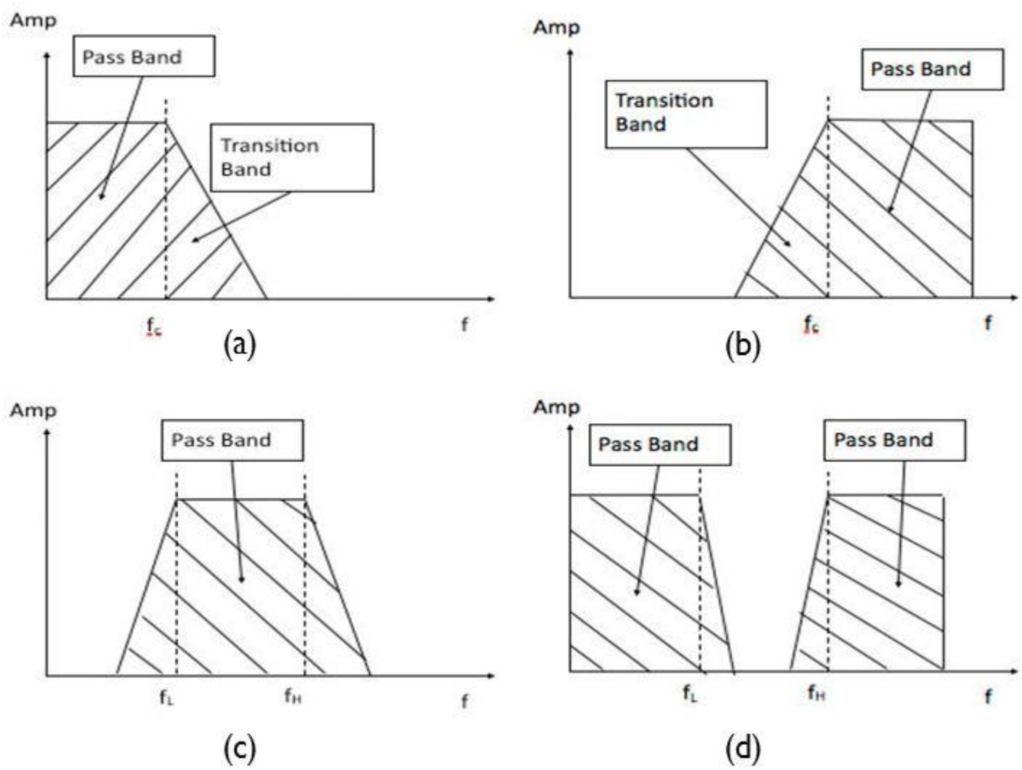


Figure 2.11: Type of filter: (a) Low-pass filter, (b) High-pass filter, (c) Band-pass filter, (d) Band stop filter.

a) Low-pass filters

Low-pass filters allow low frequencies to pass through. Low pass filters are the most common filter type because of the popularity in removing alias signals, and for other aspects of data acquisition and signal conversion.

b) High-pass filters

High-pass filters allow high frequencies to pass through. High pass filters are normally used in early bearing wear detection. A high pass filter is useful to block the high amplitude, lower frequencies to enable to "amplify" to the low amplitude levels of early bearing wear in the higher frequencies.

c) Band-pass filters

Band-pass filters allow frequencies within a band to pass through. Band pass filters transmit only those signal components within around a center frequency. Band pass filters are usually applied in situations that require extracting a specific tone, such as a test tone, from adjacent tones or broadband noise.

d) Band-stop filters

Band-stop filters block frequencies within a band from passing through. Band stop filters transmit all signals except those between specified ranges.

There are applications where a particular band, or spread, or frequencies need to be filtered from a wider range of mixed signals. Filter circuits can be designed to accomplish this task by combining the properties of low-pass and high-pass into a single filter. The result is called a band-pass filter. Creating a band-pass filter from a low-pass and high-pass filter can be illustrated using block diagrams.

CHAPTER 3

METHODOLOGY

3.1. INTRODUCTION

This chapter discussed about the methods that were used in this project and the flow of this project. There are two methods used in these projects which are experimental and simulation. The experimental will be focused automotive suspension system while the simulation it will be focused on how noise are extracted. The details of the methodology can be referred in Figure 3.1.

3.2. FLOW CHART

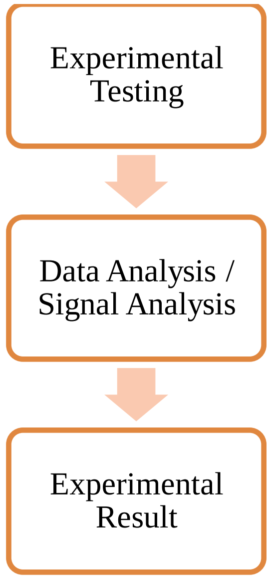


Figure 3.1: Flow chart of the methodology

3.3. EXPERIMENTAL TESTING

The design of shock absorber test rig has been developed for vibration measurement system. This product actually developed to test and indicates the condition of shock absorber in automotive vehicle. As it functioning, this product can be used as a tool to verify the capability of shock absorber. Figure 3.2 shows the complete of the design of shock absorber test rig.

This shock absorber test rig is a rigid structure with two main components connected vertical. The upper vertical is the shock absorber while the lower connection to the base structure is the pneumatic cylinder. The upper and lower component is divided by the middle plate. This middle plate is supported with two units of guide shaft for smooth movement. The shaft holder is placed at each end of the guide shaft for protecting and secures the guide shaft joints. The complete shock absorber test rig is system consist of a few important parts which are: shock absorber, guide shaft, linear guide bushes, air cylinder, air regulator and air pilot valve. This test rig is design for interchangeable shock absorber testing. Therefore, it can be used to test the shock absorber according to vehicle 850cc and 1600cc capacity.

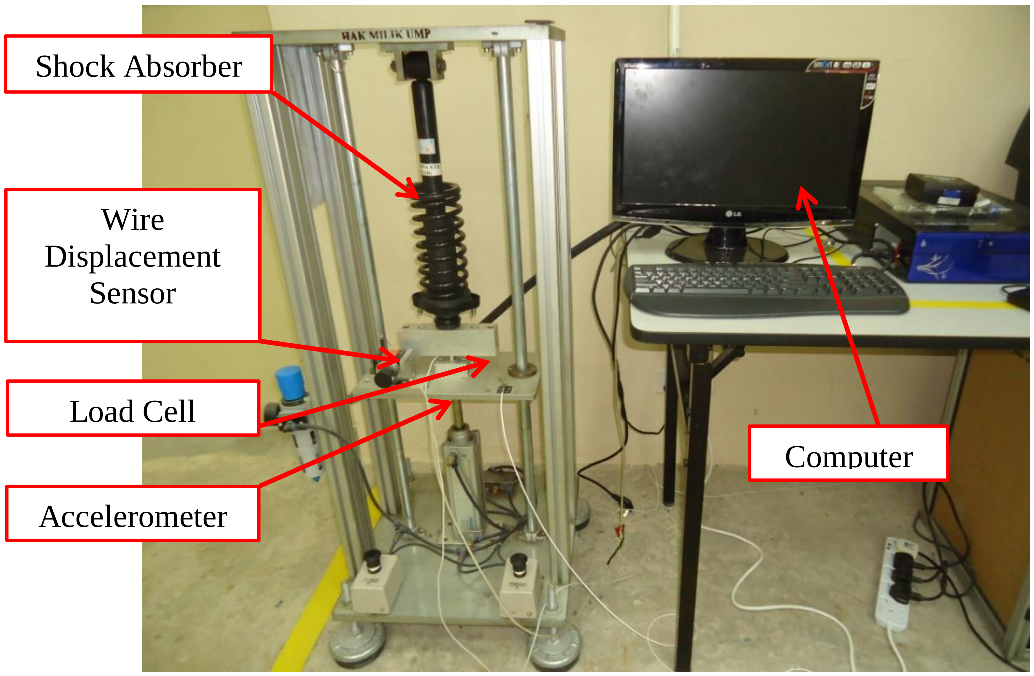


Figure 3.2: Shock absorber test rig system

In order to collect signals generated from the test rig components, there are the sensors positioned on the test rig. The unit of accelerometer is secured on the middle plate to record signal from the vibration caused from the cylinder when activated to compress the shock absorber. The accelerometer is low impedance, voltage mode designed for vibration measurement

According to Nakara, *et. al.* (2004), the sensing element (accelerometer) actually is in contact with the process and gives an output which depends in some way on the variable to be measured. Than the element that take the output of accelerometer and convert it into more suitable for further processing is a signal conditioning elements. Therefore, it is suitably processed and modified in the signal conditioning element so as to obtain the output in desired form. Figure 3.3 shows the picture of accelerometer.

As vibration can be expressed as a function of displacement, a unit of wire displacement sensor is installed at the middle plate. This wire is pulled and secured to the top plate. So, the compression of shock absorber from the activation of cylinder will show the displacement. The figure of wire displacement sensor is show in Figure 3.4.

Figure 3.3: Accelerometer

Figure 3.4: Wire displacement sensor

Furthermore, one unit of force sensor is mounted to the cylinder shaft at middle plate. This is to measure the force generated from the cylinder when activated to push up the shock absorber. The data acquisition system used is the digital type using a digital computer and has multiple channels for measurement of various physical variables. The computer controls the addressing, data input and processes the signals as desired for display and storage. The computer, control the addressing and data input and processes the signal as desired for display and storage (Nakara, *et. al.* 2004).

3.3.1. Data Acquisition System

The data acquisition system is required to collect, measure and analyze data from the test rig. Sensors are positioned on the test rig in order to collect signal generated from the test rig's component. The data acquisition system and components are:

i. Personal Computer
ii. 8 Chanel Rack
iii. Analog to Digital Acquisition Card
iv. Software (DEWESoft 6.6.7)
v. Amplifier

Refer to Figure 3.5 shows the 8 channel rack signal analyzer. The sensors that are secured on the shock absorber test rig are connected to the computer using this 8 channel rack. This 8 channel rack is a signal conditioning element. The output from the sensors is converted into more suitable output for further processing. It is because the output of the transducers or sensors element is usually too small to operate an indicator or a recorder. Therefore, it is suitably processed and modified in signal conditioning element to obtain the output in desired form.

Figure 3.5: Signal conditioning 8 channel

This shock absorber test rig is designed for vibration measurement to analyze the capability of shock absorber. Before the analysis on the shock absorber is done, this test rig must be tested. In order to perform testing on shock absorber test rig, the shock absorber was mounted on the test rig. For activation of the cylinder, the minimum of air supply from 2 to 7 bars is required. This air is directly supplied to air regulator then to solenoid valve. The right button is pressed to activate the pneumatic cylinder to compress the shock absorber. The other button was pressed to bring back the pneumatic cylinder to home position. When the test rig is at home position, the cylinder piston is retracted.

The input signal of all sensors is recorded by the Data Acquisition System after generated from the test rig. The force generated from the cylinder is measured by the load cell. The displacement of the shock absorber is measured by wire displacement sensor while the vibration of the middle plate is measured by accelerometer. All the input signal from the sensor than were processed to get the output data.

3.3.2. Experimental Procedure

The procedures for the laboratory experiment are as follows:

i. Select testing capacity.
ii. If the test rig is set-up with different range of testing capacity, remove away the upper and lower block and replace them with the actual block testing.
iii. Tighten and secure all the blocks.
iv. Switch ON the power supply for computer and data acquisition system.
v. Turn ON the air supply and adjust the air regulator to testing criteria.
vi. At this moment, the data acquisition system must be turn ON, because it can make record the sensor signals before activating the cylinder to move up position.
vii. Press the push button to activate the cylinder to move up.
viii. Press another push button to activate the cylinder to down position.
ix. Push button up and down for 5 times.
x. Turn OFF the recording signal from the software.

3.4. DATA INTERPRETATION

Spectrum of frequency analyzers can be used for signal analysis. A spectrum or frequency analyzers is a device that analyzes a signal in frequency domain by separating the energy of the signal into various frequency band. The separation of signal energy into frequency bands is accomplished through high-pass then low-pass filters. The conversions of data signal into PSD were done by FFT analysis.

3.4.1. Fast Fourier Transform (FFT) Data Signal

One method of finding PSD is the usage of Fast Fourier transform. Fast Fourier transform is the simplest form known as the decimation in time algorithm. The central insight which leads to this algorithm is the realization that a discrete Fourier transform of a sequence of N points can be written in terms of two discrete Fourier transforms of length

$N=2$. Thus if N is a power of two, it is possible to recursively apply this decomposition until we are left with discrete Fourier transforms of single points.

3.4.2. Power Density Spectrum

Computation of PSD is done directly by the method called FFT or computing autocorrelation function and then transforming it. Power spectral density function (PSD) shows the strength of the variations (energy) as a function of frequency therefore shows at which frequencies variations are strong and at which frequencies variations are weak. The unit of PSD is energy per frequency (width) and you can obtain energy within a specific frequency range by integrating PSD within that frequency range.

PSD tells us at which frequency ranges variations are strong and that might be quite useful for further analysis. PSD is use to identify oscillatory signals in your time series data and want to know their amplitude thus able to analyze unwanted noise and from the PSD able to find the optimum condition for noise reduction.

3.4.3. DASYLab® Software

DASYLab® is a popular Easy-To-Use Software in Data Acquisition System with all kinds of interface connected to the hardware, such as RS232, RS485, RS422, USB, Parallel port, ISA bus and PCI bus, etc. Furthermore, various function modules for measurement and control are supplied by DASYLab®. For DAQ card (in PCI and ISA bus) solution, user only needs to install corresponding drivers from ICP DAS. Thus DAQ interface cards can be applied to the environment of measurement and control using DASYLab®. The following is a part of application interface by DasyLab® and the bundled driver

Use DASYLab® to interactively create an acquisition, control, simulation or analysis task. Simply select a function module and place it on the worksheet. Configure your task by connecting the modules. DASYLab® supports many data acquisition and control devices, as well as different interfaces that communicate with external instruments.

DASYLab® includes acquisition, control and analysis modules, including analog and digital inputs and outputs, mathematics, statistics, digital filters, FFT analysis, and switches. Logical operations, switches and sliders or the Sequence Generator provide exact, time-based waveforms for complex control signals. Store the results of the data acquisition and analysis for off-line analysis by other programs.

The Chart Recorder, *Y/t* Chart and the *X/Y* Chart will plot the data as curves. The List Display and Digital Meter display the data numerically. Freely scalable Analog Meter, Bar Graph and Status Lamp displays are useful for Process and Measurement controls.

3.4.4. Defining Experiments in the Worksheet Window

Firstly after start DASYLab® software, the DASYLab® Worksheet window appears with the menu, Function Bar, Module Bar, Browser and the Info Area. Click on the Function bar Worksheet button to open the Worksheet view from one of the other DASYLab® views. Use the worksheet window to create your measurement worksheet. Select a function module from the Browser or Module bar and place it on the work area.

The Modules symbolize the program functions, starting from the acquisition and generation of the data up to output and display of that data. A worksheet can contain up to 256 modules. Use the Black Box module to put an additional 256 modules in a "sub-worksheet". The multiple Black Box modules, and each Black Box can contain Black Boxes. Open a Black Box by double-clicking on the module, close it by clicking on the Close Black Box button on the Function bar. Refer to Figure 3.6 shows the DASYLab® Worksheet Window with the work area, the Module Bar and Browser.

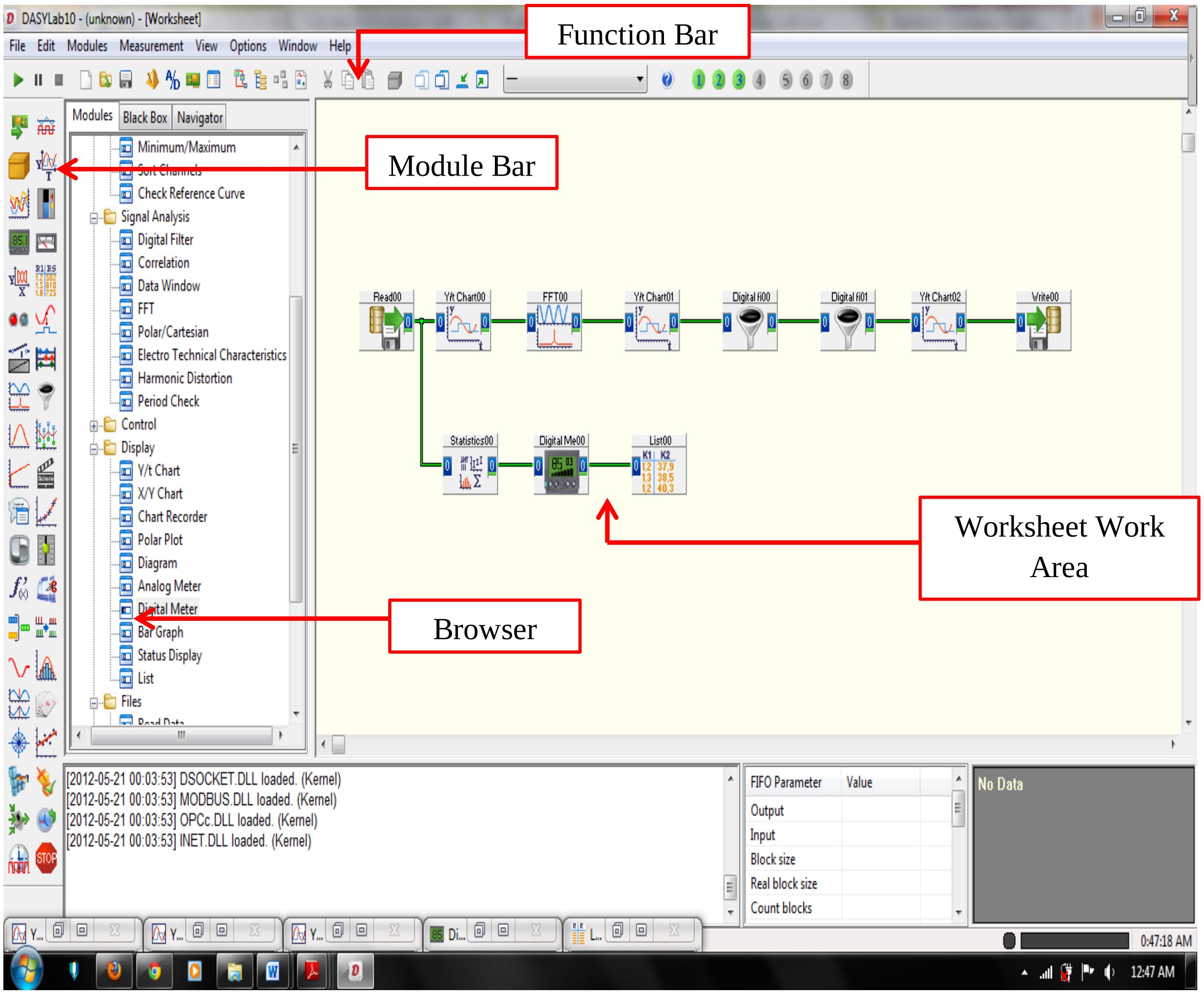


Figure 3.6: Worksheet view

Use the Browser to select a module and drag and drop it onto the work area. The Browser shows all modules that were installed with DASYLab® and the selected drivers and options. The Browser uses a tree structure, organized by a module group. The Module bar contains a selection of modules that are used frequently. Click on the module icon in the Module bar, move the mouse to the work area, and click to place the module. You can choose to display or remove the Module bar by selecting View and then Module Bar.

CHAPTER 4

RESULT AND DISCUSSION

4.1 INTRODUCTION

This section will present the signals obtained from laboratory experiments that the vibration signals at different frequencies. The types of signals are presented in graphical form the time domain and frequency domain. The data obtained were studied through a statistical analysis of behavioral statistics. The main purpose of all this analysis is to extract noise from the vibration signals, as described in Section 4.3.3 to 4.3.2.

4.2 SIGNAL PRESENTATION

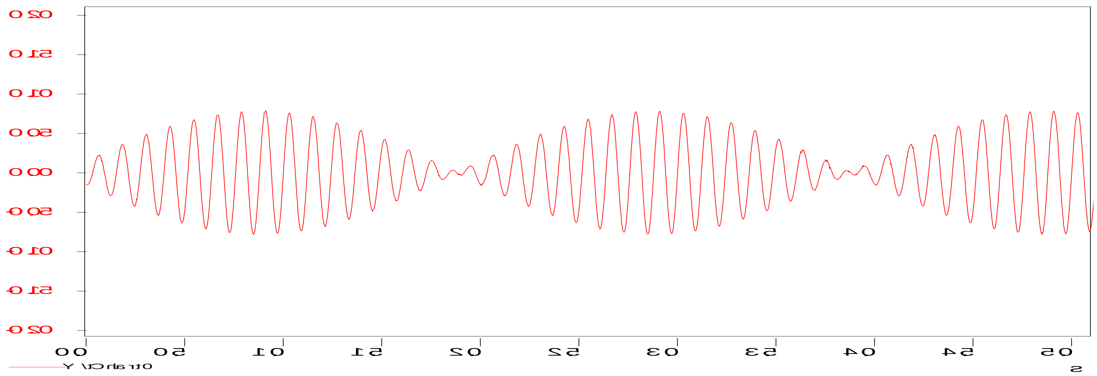
4.2.1 Vibration Signal

Vibration signals were observed to get the value of each point of observation of the vibration signal. Figure 4.1 and 4.2 shows the vibration signal has been observed during laboratory experiments conducted. Vibration signal collection also varies according to the different frequencies.

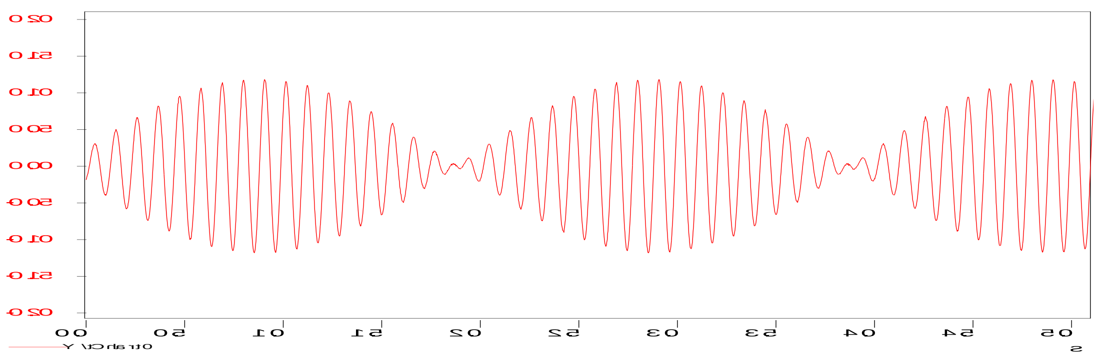
Refer to Figure 4.1, the vibration signals collected at 10 Hz gives a total amplitude of the vibration of the highest compared with other frequencies of the value 0.075 volt (V).

Followed by 9 Hz, and 8 Hz, are respectively, with the amplitude of the vibration of 0.055 V and 0.035 V. It should be noted that the value of the amplitude amount is smaller when the frequency decreases. Justification the amplitude of the vibration pattern of the frequency value discussed in the next paragraph.

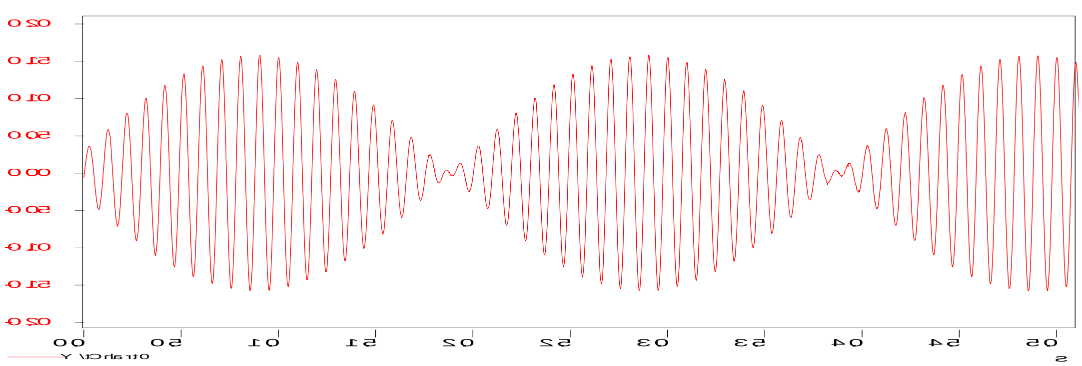
The pure vibrations signals contains a combinations of vibrations signals which shown that vibrations signals are not-smooth and there are still disturbance present in the vibrations signals.



(a)



(b)



(c)

Figure 4.1: Plots of vibration signal in time domain to frequency 8-10 Hz: (a) 8 Hz, (b) 9 Hz and (c) 10 Hz

According to the vibration signal, the total amplitude vibration is highest when applied loading at a frequency of 10 Hz. These results indicate a high frequency value represents the value of the force applied to the component. The increase of the force applied will increase the displacement of the spring coils and increase the vibration response and the strain on the components.

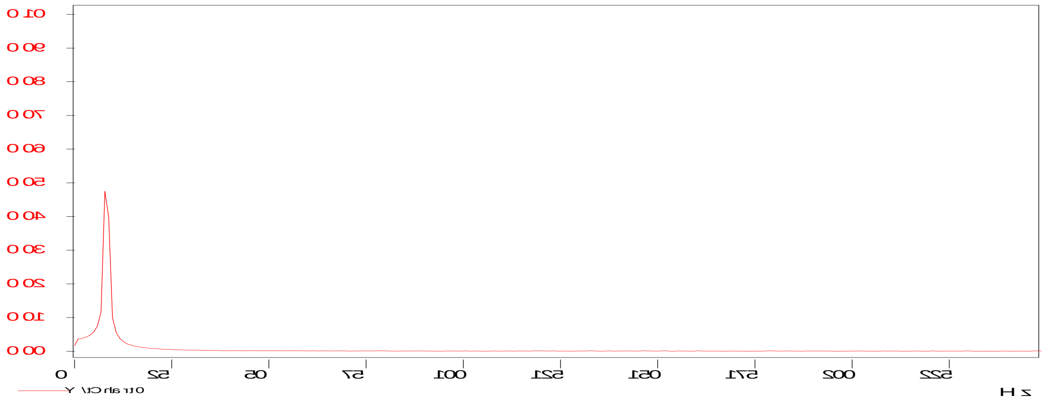
Table 4.1: Total strain amplitude vibration for each frequency

Frequency (Hz)	**Total Amplitude of Vibration (V)**
8	0.035
9	0.055
10	0.075

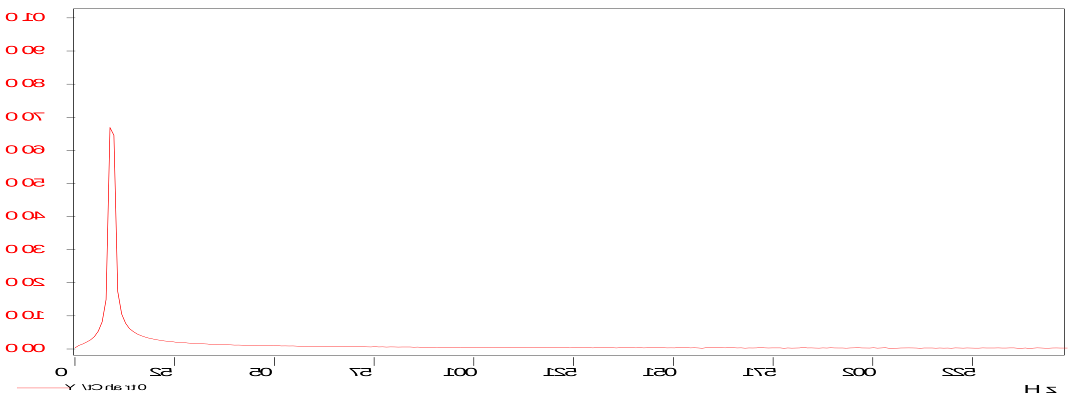
Below are the graph that show the PSD when loading are applied at a frequency of 8Hz, 9Hz and 10Hz.and the highest peak shows Total Amplitude of Vibration (V) for the signals which are 0.035V, 0.055V and 0.075V. From Figure 4.2 show the PSD graphs which generations of a pure vibrations signals before a filter conditions are applied.

The PSD graph indicates that the graph is not-smooth and there is a disturbance which is present in the pure vibrations signals. From the PSD graphs the vibrations signals are not smooth which indicates there are still noise present in the vibrations signals and therefore a filter need to be introduce in order to filter the noise present in the vibrations signals. Filtrations are done by setting a high-pass and a low pass-filter according to certain frequency which makes a certain vibrations signals frequency to pass within these ranges.

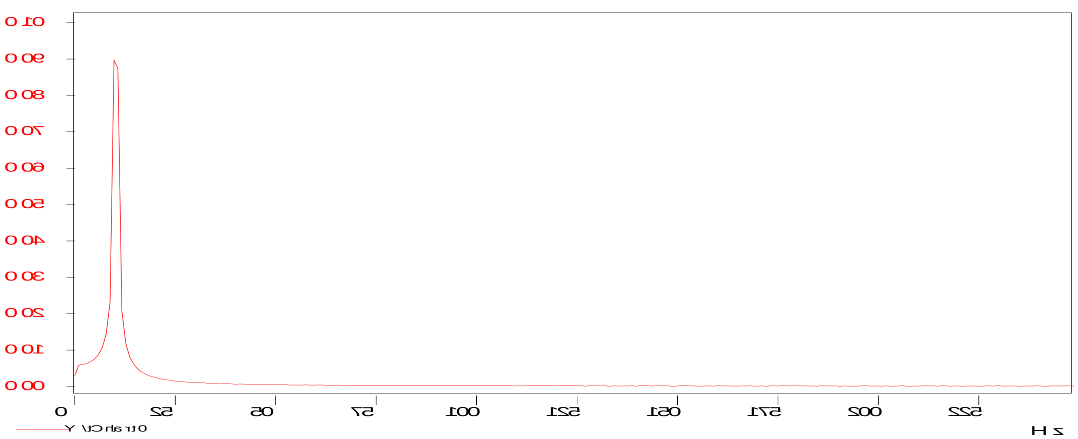
Therefore when making the PSD graphs, the PSD graphs are smooth and thus creating a free-noise vibrations signals which is the noise are suppress or in other words remove from the vibrations signals.



(a)



(b)



(c)

Figure 4.2: Plots of vibration signal in frequency domain (FFT) to frequency 8-10 Hz : (a) 8 Hz, (b) 9 Hz and (c) 10 Hz

4.3 DATA ANALISIS

4.3.1 Statistical Analysis

Statistical analysis is one of the analyses performed to classify the vibration signal collected through laboratory experiments. There are four statistical parameters used in this study the mean, standard deviation, root mean square (r.m.s) and kurtosis.

An average or mean is select as a statistical parameter in this analysis is to identify the types of forces experienced by a coil spring that is whether the tension or compression force. Standard deviation measures the average value of data distribution. Standard deviation indicates that every point in a data set is far from the average value of the data. The parameters of the root mean square (r.m.s) are used to measure the total energy of the observed signal. The last parameter is the statistical kurtosis and these values are sensitive to the shape and amplitude of high data transient.

Refer to Table 4.2 shows the values of statistical parameters for signal vibration. For the vibration signal, the average value for all values of vibration frequencies is 0 V. This situation can be seen clearly in Figure 4.2 Standard deviation increasing with the increase in the frequency. This situation can be seen at 8 Hz to 10 Hz is the standard deviation increased from 2.5 x 10^{-2} V, 3.9 x 10^{-2} V, to 5.3 x 10^{-2} V. Based on these values, 8 Hz give the standard deviation of lowest and 10 Hz provide the highest standard deviation. This condition shows the data in each data set on the rise as the frequency increases.

Based on Table 4.2, is the r.m.s value is equal to the standard deviation. The r.m.s value is equal to the standard deviation signal with zero mean value. Thus, the results of statistical study are valid as the statement above. Kurtosis values also showed an increase for each frequency value. These values can be seen clearly in Table 4.2.

Table 4.2: Statistical value of the vibration signal at each frequency

Frequency (Hz)	**Mean**	**Standard Deviation ($x\ 10^{-2}$, V)**	**r.m.s ($x\ 10^{-2}$, V)**	**Kurtosis**
8	0	3.9	3.9	2.2571
9	0	5.9	5.9	2.2580
10	0	7.8	7.8	2.2581

4.3.2 Noise

The analysis of rotating machinery is central to refinement activities in automotive and general industry. It also allows engineers to trace faults in gearboxes, transmission systems and bearings. Every rotating part in a machine generates vibration, and hence noise, as a result of small imperfections in the balance or smoothness of the components of the machine.

In every case, we can relate the frequency of the vibration to the speed of the rotating machine. For this experiment is based on literature review. This experiment is important to observe the vibration signal to be used as input to determine the resulting noise. This situation was the same analyzes conducted based on the literature review.

Refers to the Figure 4.3 and Figure 4.4 below, it shows that there is noise in each graph of the vibration signal is taken from experiment. Based on the vibration signal is obtained shows that there is noise, it does meet the objective of which is to extract the noise using the frequency domain.

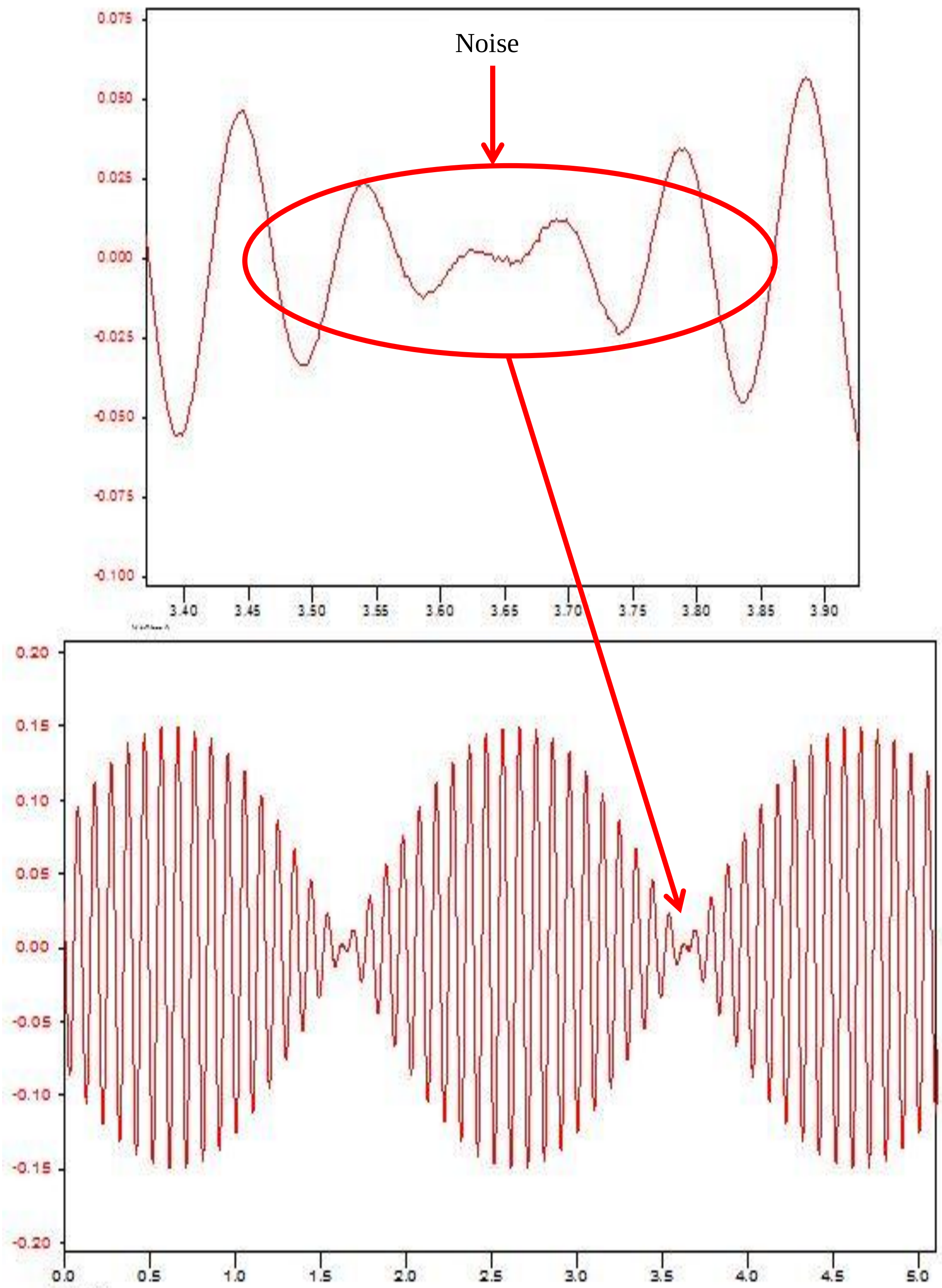


Figure 4.3: Noise signal at one of the time domain signal

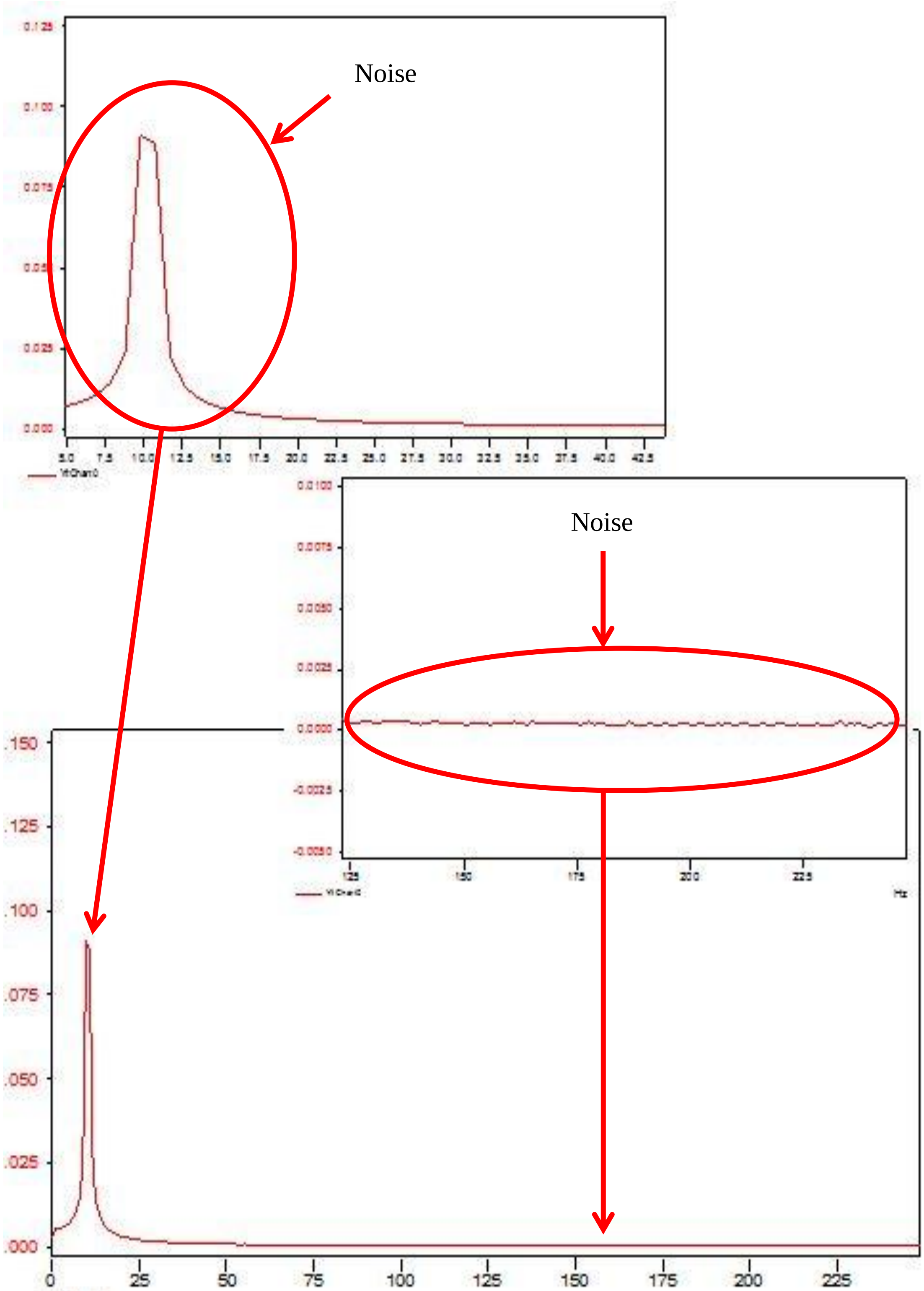


Figure 4.4: Noise signal at one of the frequency domain signal

4.3.3 Filter

Vibration signal from a transducer requires signal processing to produce the data that we need. The process usually involves filtering, setting sample rates and resolution, windowing, etc. It is important to understand what filters do and how they are used in the field of vibration. Filtering is a process that removes some frequencies from a signal in order to suppress interfering frequencies and reduce noise.

In signal processing, a filter is a device or process that removes from a signal some unwanted component or feature. Filtering is a class of signal processing, the defining feature of filters being the complete or partial suppression of some aspect of the signal. Most often, this means removing some frequencies and not others in order to suppress interfering signals and reduce background noise.

Refer to Figure 4.5 until Figure 4.8, by using DASYLAB® software. Firstly read the data signals from signal source which are from excel data. Second, the data were plotted in the *y/t* chart then converted to FFT chart to get the power spectrum density. Third, from the FFT chart the data goes to the digital signal to undergo low pass filter which is 8 Hz, 9 Hz and 10 Hz. Fourth, it goes to another digital filter which is the high pass filter which is 8 Hz, 9 Hz and 10 Hz, these values can be seen clearly in Table 4.3. Both of the filtration process is called band pass filter. Fifth, the filtered signals were plotted in the *y/t* chart. Finally, from the *y/t* chart, the after filtered data were analyzed to get the optimum condition for free noise vibration signal. The statistical analysis values which are the, r.m.s, means, standard deviation and kurtosis were calculated and the values are compared.

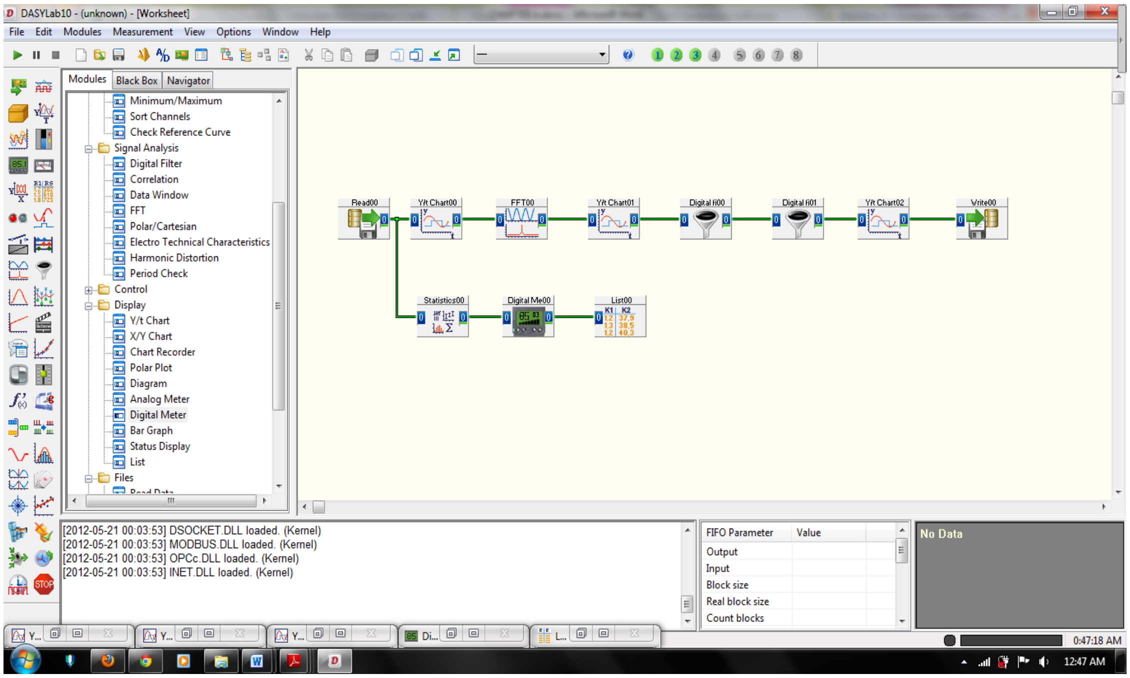


Figure 4.5: DASYLab® worksheet

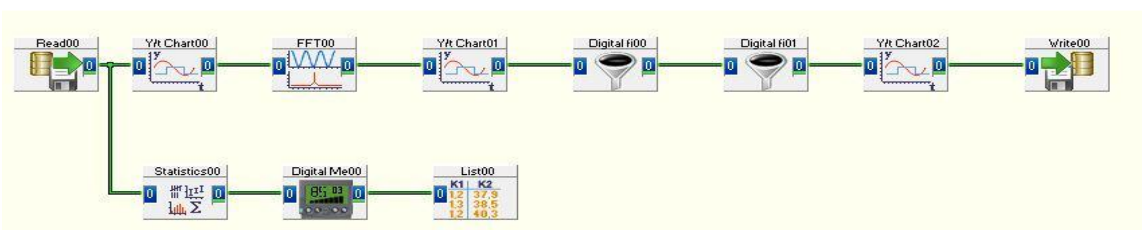
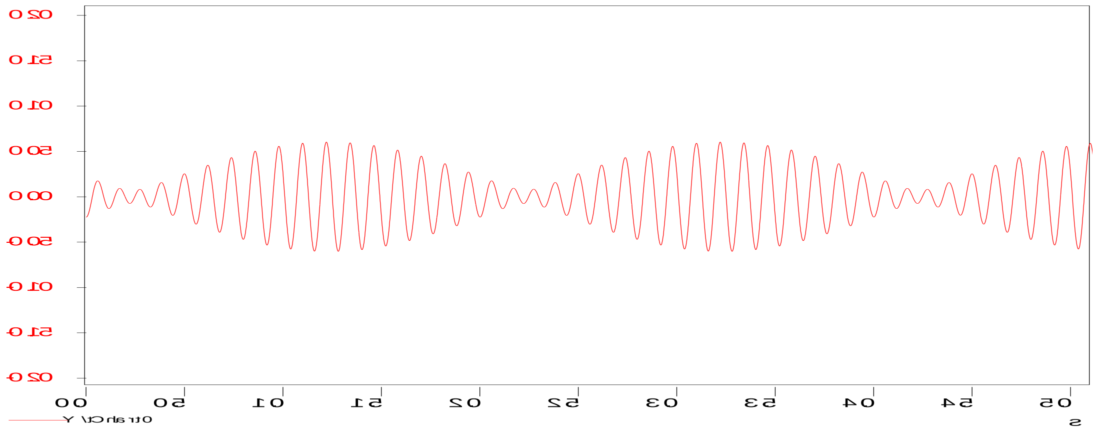


Figure 4.6: DASYLab® procedure process

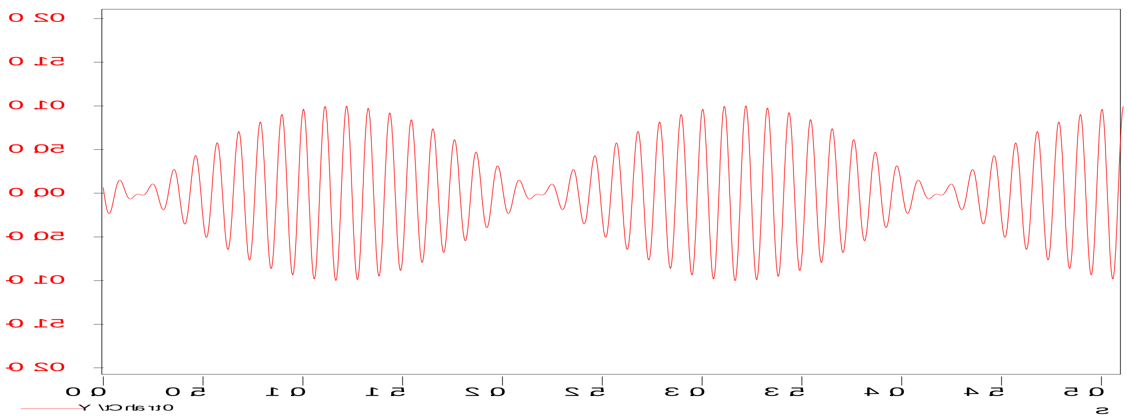
Table 4.3: Data filtering testing

Low-Pass Filter	**High-Pass Filter**
8 Hz	9 Hz
8 Hz	10 Hz
9 Hz	8 Hz
9 Hz	10 Hz
10 Hz	8 Hz
10 Hz	10 Hz

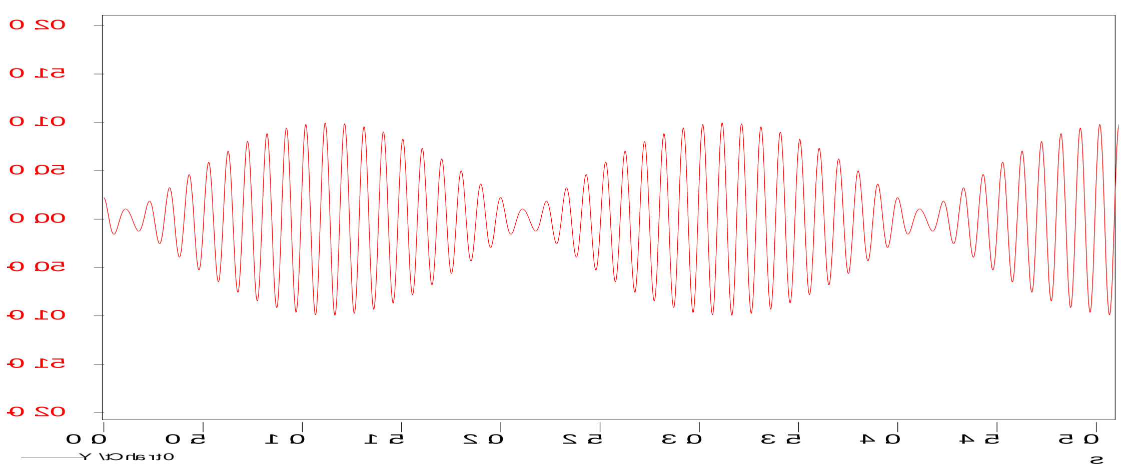
The removal of noise in the pure vibrations signals creating a free noise vibrations signals. A free noise vibrations signals only contains the vibrations signals only without the present of noise. The noise is already filtered when the pure vibrations signal undergoes filtering process which involves the setting of a high-pass filter and a low-pass filter.



(a)



(b)

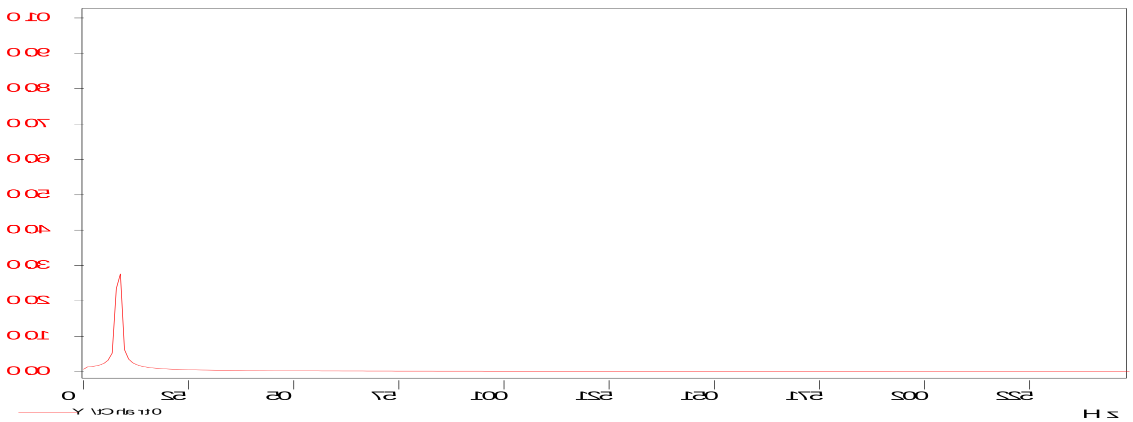


(c)

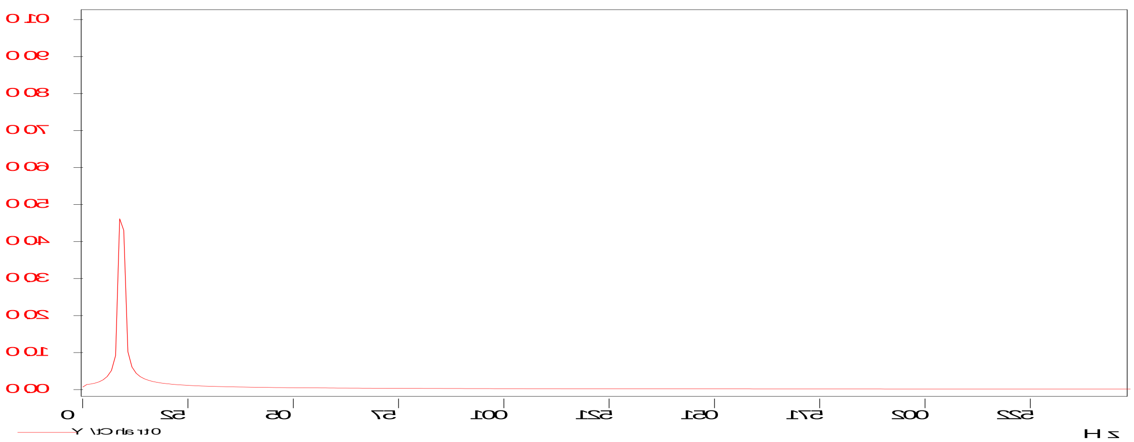
Figure 4.7: Plots of free noise vibration signal in time domain to frequency 8 -10 Hz: (a) 8 Hz, (b) 9 Hz and (c) 10 Hz

The graph shown are after the signals undergone a high-pass filter of 8 Hz and a low-pass filter of 10 Hz. All the vibrations signals are repeated with the same filters-pass conditions. From Figure 4.8 show the PSD graphs, there is a change in the signals vibrations which can be seen in the PSD graphs the vibrations signals output are smooth and noise reduces drastically.

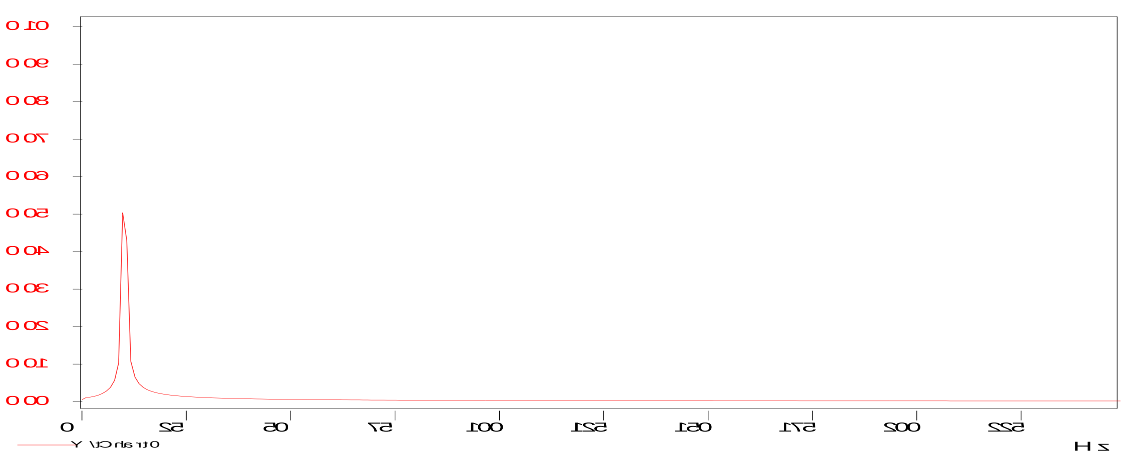
Therefore the experiment is a success, which indicates a decrease in the noise present in the vibration signals or the noise were completely suppress. In other words the noises in the vibrations signals that which were initially present before the filters conditions are applied were successfully filtered.



(a)



(b)



(c)

Figure 4.8: Plots of free noise vibration signal in frequency domain to frequency 8 -10 Hz: (a) 8 Hz, (b) 9 Hz and (c) 10 Hz

4.4 OPTIMASATION OF NOISE EXTRACTION

The generation of power spectrum density, low-pass filter and high-pass filter is generated and it is use to extract noise from the data signal. There are 2 ways to extract noise from a vibration signal which is by using a low pass filter and a low pass filter which is the band-pass filter.

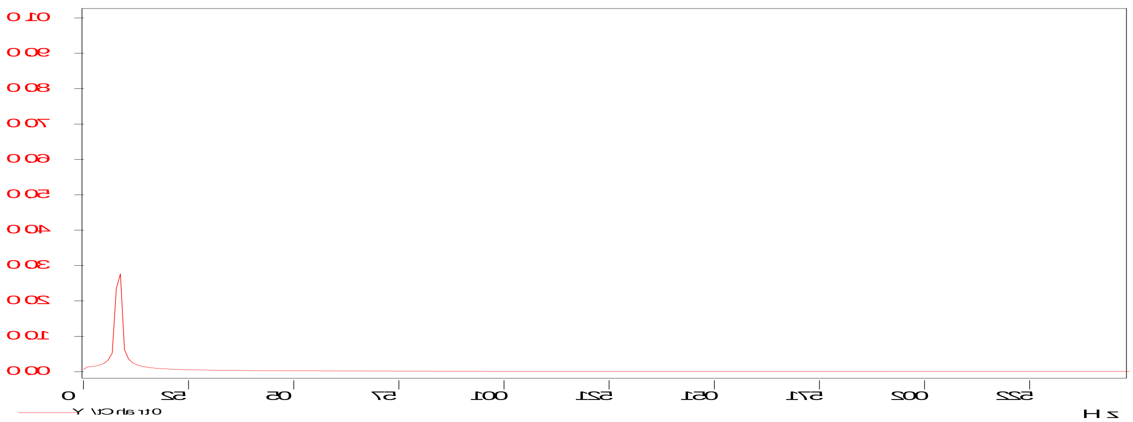
A filter is use to extract noise from the vibration signal, then it is compare to the original vibration signal. By using power spectrum density, it can be seen that the spectrum is in optimum condition compare to the original power density spectrum. A low-pass filter frequency only allows low frequency to pass through, but block the high frequency, the frequency that the filter begins to attenuate the content. A high-pass filter frequency only allows low frequency to pass through, but block the low frequency, the frequency that the filter begins to attenuate the content. Band-pass filter only allow frequency between at any 2 given frequency to be pass through.

The optimization of the low-pass filter is optimum when the frequency of the filter is set up at 10 Hz. Which the frequency only allows to pass is a below than 10 Hz frequency. When the low-pass filter is set at frequency of 8 Hz or 9 Hz, the noise cannot be extracted and the power spectrum density is not optimum condition. When the low-pass filter is set at frequency of 10 Hz the noise cannot be extracted and the power spectrum density is not in optimum condition.

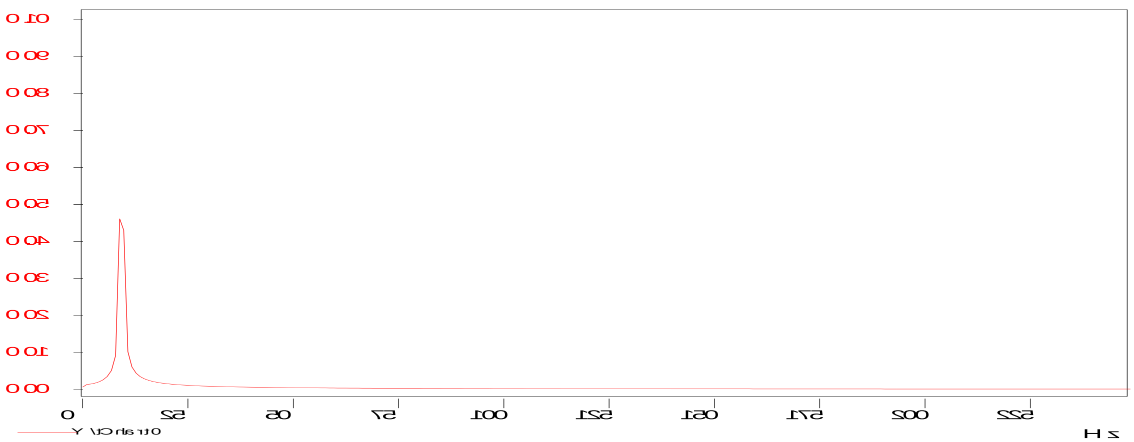
The optimization of the high pass filter is optimum when the frequency is of the filter is set up at 8 Hz. Which the frequency only allows to pass is a higher than 8 Hz frequency. When the high-pass filter is set at frequency of 9 Hz and 10 Hz, the noise cannot be extracted.

When the high-pass filter is set at frequency of 8 Hz the noise cannot be extracted and the power spectrum density is not optimum condition. When the high-pass filter is set at frequency of 10 Hz the noise cannot be extracted and the power spectrum density is not in optimum condition. When low-pass filter and the high-pass filter are set to the frequency of 9 Hz, the power spectrum density is not optimize thus the noise cannot be extracted when the low-pass filter and high-pass filter is set to 9 Hz frequency.

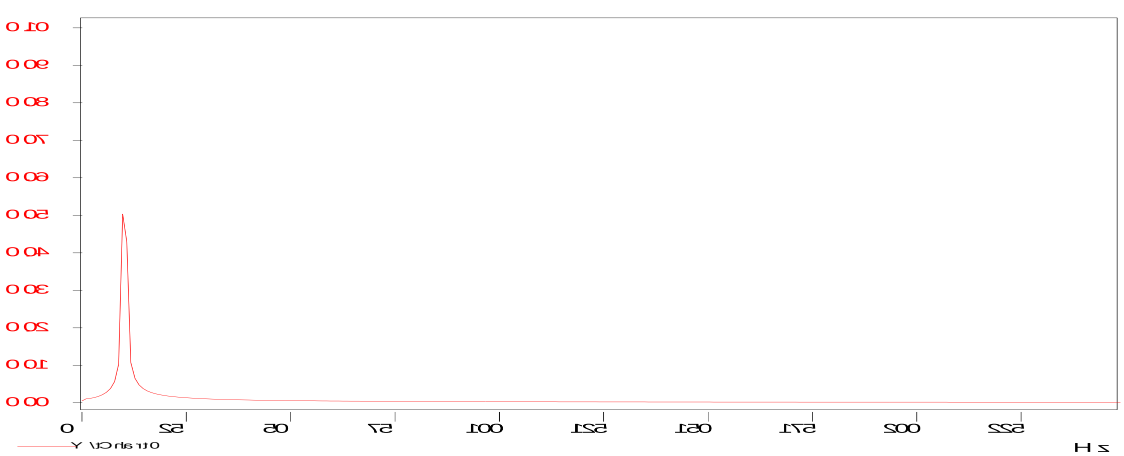
The power spectrum density graphs for optimization of noise were shown in Figure 4.9 and the values of set of frequency that were used in noise optimization were shown in Table 4.4.



(a)



(b)



(c)

Figure 4.9: The optimization of free noise vibration signal in frequency domain with frequency 8-10 Hz: (a) 8 Hz, (b) 9 Hz and (c) 10 Hz

Table 4.4: Data optimization testing

Low-Pass Filter	**High-Pass Filter**	**Noise**
8Hz	9Hz	Present
8Hz	10Hz	Optimize
9Hz	8Hz	Present
9Hz	10Hz	Present
10Hz	8Hz	Present
10Hz	10Hz	Present

4.5 RESULT SUMMARY

The objective of this study is to extract the noise by using frequency domain analysis is achieved. The noise extraction using frequency domain analyses were analyzed.

From the data generated from DASYLab® software, the data signals data were process into FFT and PSD graph were plotted. From the PSD graph generated, the graphs were analyzed to get which is the most suitable band pass filter for noise filtration. As a conclusion from the analysis of PSD graph the most suitable condition for band pass filter is when the low pass filter is set to 8 Hz and the high pass filter is set to 10 Hz.

The analysis were further based on finding an optimization frequency for a high-pass filter and a low-pass filter thus finding the most optimum power density spectrum by using frequency domain analysis.

CHAPTER 5

CONCLUSION AND RECOMMENDATION

5.1 INTRODUCTION

In this chapter, the findings and result of this study were concluded. A recommendation for improvement of experiment were suggested which base on the overall aspect of the experimental study for further studies.

5.2 CONCLUSION

In the present study, finally the objective of this study is to extract the noise by using frequency domain analysis is achieved. The noise extraction using frequency domain analysis was analyzed. The analysis were based on finding an optimization frequency for a high-pass filter and a low-pass filter thus finding the most optimum power density spectrum by using frequency domain analysis.

The optimization of the low-pass filter is optimum when the frequency of the filter is set up at 10 Hz. The optimization of the high pass filter is optimum when the frequency is of the filter is set up at 8 Hz. Therefore at these conditions of frequency the noise was extracted and the power spectrum density was optimized.

5.3 RECOMMENDATION

This topic discusses the recommendation of the study which involve proposing an improvement of the study. It is important for future use and for proposing a guideline for future study. Noise extraction involves in vibrations signals therefore the more vibrations signals use for the study the more the accurate the results, even better if it involving a vibrations signals that are generated from a real application in real conditions and not from a test-rig conditions.

Damper functions are to lower machine vibrations. If the vibration of machine is lowered thus the damper will also have effect on the noise. Therefore introducing a new test rig consist of a combination of a damper spring system in automotive spring will also have effect in the noise reductions procedure.

Using a real model in the experiment will also give accurate data and result therefore it is better to use a real model such as getting the data of an automotive spring while a car is moving to give an accurate results.

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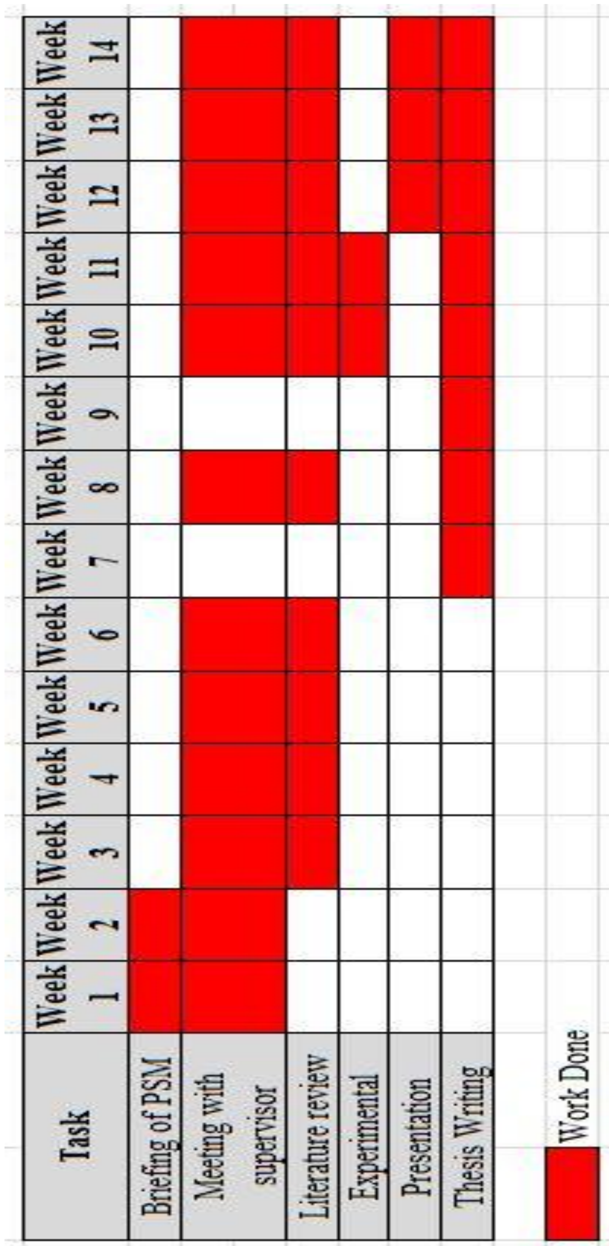
APPENDIX A1

GANTT CHART FOR FINAL YEAR PROJECT 1



APPENDIX A2

GANTT CHART FOR FINAL YEAR PROJECT 2