Development of a Prototype System for Milling Tool Condition Monitoring

Master of Science Thesis [in the Programme Software Engineering and Technology]

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Thesis Report

Preface

This is a master thesis in the master program named “Master in Software Engineering and Technology” at Chalmers University of Technology. The supervisor for this thesis work at Chalmers University of Technology is Per Zaring.
We are very thankful to our supervisor who guided us in a right way and provided us all the necessary help for better understanding of the thesis work and writing a comprehensive final report for the thesis work. We are also grateful to Ella Olsson and Peter Funk who helped us very much in analyzing the recordings and guided us very much in for the implementations.
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ABSTRACT

In this thesis we shall present a new technique for signal processing. The main aim of this thesis is to monitoring cutting process of milling machine. We shall provide some results and then try to implement a prototype system based on analyzed result. We shall also describe some very famous and old techniques for cutting process which are already used in current professional industry. Then we shall describe why these techniques are not so good.

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1. Introduction

1.1. Background

The machine building industry is growing very quickly and becoming large and large day by day in the current era. As new technologies are introduced in the machine building industry therefore fast and precise but also quite expensive machines are coming into use during the production processes. By using the latest and valuable technologies for the production; the need of automation got more importance to minimize the risks and damages. It also reduces the work burden of continuous monitoring on people at working place. The most important part for development of the prototype system in order to guarantee the smooth production steps during the large-scale producing process is to monitor the condition of the cutting tool, whether it is worn out or not. If the condition of the cutting tool is worn out, an immediate alarm should be arisen and the machine should be stopped automatically. It is not only helpful for the improvement in the work efficiency and quality of the product but also can help to avoid the damage to the machine itself and save large amount of money and time caused by the damage of the deeply worn out tool.

The development of the current world is very much due to the progress of new and emerging technologies, and manufacturing industry is the basic industry which plays a very important role in good progress. It has very deep and important relationships with all aspect of our life. We take the example of Berg Propulsion a well known and reputed industry near the beautiful city of Gothenburg Sweden, As the Gothenburg city is surrounded by the sea water therefore need for water transportation is much important in the city. The main and important component for the ships is propellers which are produced by milling machines. The Berg Propulsion produces the propellers and hence plays an important role in water transportation industry.

There are very high manufacturing nowadays, machine cutting process becomes important too. But there is need of high accuracy in automation process. It has many benefits to the companies such as reduction of loss of money and interference of human. With manual process system, there will possibility of making human error a lot, so it can affect the quality of cutting process.
In Berg Propulsion, they are using very expensive machines to produce hub body of many kinds of boats. Hub body consists of bronze material. While cutting material, it produce immense heat which make cutting tool warm too. After some time, cutting tool will wear out. If cutting tool has worn out and still cutting the hub, it damage the surface of hub body and degraded hub body will be the result. So, it will waste a lot of money and time. Of course, there is cold water coming out to make cooler the cutting tool but it cannot prevent to wear out tool. It is impossible to eliminate of human error, no operator can guess status of cutting tool as it is worn out or not.

In the company, they use manual system; they are relying heavily on operator experience. Operator will listen sound of cutting tool. All the workers are in working condition now and analyze sound waves carefully which are coming from machine cabin. As, cutting sound become abnormal, operator stop the machine, open the cabin and walk inside the cabin carefully. He changes the cutting tool with the new one. It is implied that company has latest model machines which is high-tech and well equipped. Still, they use a lot of man power to operate that machine to work in correct manner. We can imply one more thing, to analyze sound worker should be a very experienced person. He should be really expert in listening sound and figure out the sound when cutting tool becomes worn out. As, making hub body is very long time process, so in night shifts we need another experienced operator which can analyze sound. As company has milling machine which is very costly and it require extra care of continuous observation and maintenance. So, it requires company to utilize machine and human resource as efficiently as possible to minimize the risk of wasting money. Also, little damage in cutting tool may cause damage the surface of the hub body. The damage can be so significant that it can be rejected. Also, worn out cutting tool may damage the accessories of machine which make thing really worst. So, we need a stable method and quality algorithm to monitor the whole process. So, we can check whether cutting tool is worn out or not, so we can take decision easily to change the cutting tool or not.

In our thesis, we shall analyze sound data coming from microphone. We try to observe sound wave of worn out cutting tool automatically from computer. We shall develop pilot system which can signal to the operator that the cutting tool has worn out, which immediately will change the cutting tool. Up till now, we have not decided how we signal to the operator.

As a whole, monitoring cutting tool present us following benefits:

Firstly, it enhances the efficiency of the milling machine.

Secondly, it ensures the quality of hub body should be up to the standard.

Thirdly, it saves man power and there will no breakage of equipment of machine or disturb the surface of hub body. As cutting tool is worn out, machine should stop immediately to minimize the risk of damaging of a hub surface.

Last but not the least; it can save money of improper changing o cutting tool. As, human is meant to make mistakes. He can listen sound wrongly and change cutting tool before of worn out. In fact, it can be used longer for cutting. To some extend it is kind of waste of money too.

1.1.1. State of art

Although cutting tool monitoring has progressed very quickly since last decade but there is still a lot of work has to be done in the field. Some methods for monitoring the cutting tool are only suitable in one special area and some are in testing phase, so a lot of work is required to put them into the practice. Hence we can say that there is a very long way to go in the field of monitoring the cutting tool conditions.

Automatic monitoring of cutting tools may be divided into two main parts. One is direct monitoring method and the other is indirect monitor method. In direct monitor method parameters of cutting tools like position of the cutting tools, shape of the cutting tools and so on are monitor directly. The direct monitor method can be performed off line only. The direct method can be achieved by the help of several regular methods like contact resistance, optical method, optical fibre, pick-up of TV signals and so on. In indirect monitor method we measure some of the parameters of the cutting tools during the cutting process and compare them with normal cutting parameters to analysis whether the cutting tool is worn out or not. This is the method which can lead us to achieve our goal. There are a lot of methods to perform indirect monitor method. Cutting resistance, tensional moment, cutting vibration, acoustic emission, cutting temperature, work piece dimension, work piece surface roughness, cutting sound method etc are well known and good examples of the indirect monitor method.

Some famous and important conventional methods for cutting tools are described below [1]

- **Ray Measurement**
  In this method some radical material is mixed in the cutting tools and then the image is taken with the wearing and tearing of the tool itself. Then the micro particle of the radical material will go through a previously designed ray measurer together with the cutting chip. The micro particle will be closely related to the cutting tool wear extent in ray measurement technique. The amount of the ray reflects the abrasion loss of the cutting tool. Although it is a good technique for monitoring the cutting tool but still there are some drawbacks in this technique. First of all it is not an easy process to make all radical micro particles into the ray meter. The other and important disadvantage of the ray measurement method is that it pollutes the atmosphere a lot and is also bad for the health of the people. So, the ray measurement method is not widely used due to the reasons described above.
- **Optical Fibre Measurement**
  This method is good to use when the diameter of the cutting tool is big. In this method the reflection capacity changes of the edges after the cutting tool is worn out considered for analysis. As it involves the consideration of the edges of the cutting tool therefore it is not a good idea to use this technique in the projects which uses the small cutting tools for monitoring the condition.
- **Computer Image Processing**
  In this method a camera is used to receive the images of the edges of the cutting tool. These images are digitalized and processed by the computer system and as a result the shape and dimension of the cutting tool will be displayed on the computer screen. This method seems to be good but still it is not very useful for the real world problems and used in only automatic monitoring in laboratory. It is not very useful in real world because it demands very high requirements. In actual production process, the working environment is very horrible. There are hot chips of cutting material scattered all around and cold water is sprinkled on the material to cool down the temperature of the cutting material therefore it is impossible to use this technique in the real world in these conditions.
- **Cutting Resistance Measurement**
  Cutting resistance is an important and most related physical feature to the worn extent during the process of cutting. There exists a lot of parameters which should be considered and take care while measuring the cutting resistance. Cutting force and its differential coefficient, the ratio of the cutting resistance, spectrum analysis, time-series analysis and correlation function of the dynamic cutting resistance are some of the parameters which should be considered. As it involves

a lot of factors to be considered so it is very complicated method during the cutting process to discover whether the cutting tool is worn out or not. Hence this method is also not very useful to use in monitoring the condition of the cutting tool.

- **Work piece Dimension Measurement**
  The size of the work piece changes while the tool is worn out during the cutting process. Hence it is possible to determine the condition of cutting tool indirectly by measuring the changes in the dimensions of the article surface. These measurements can be carried out through two different ways. These are contact based work piece measurement and contactless measurement in which we measure the distance meter age between the cutting tool and work piece. The main advantage of using this technique is that we can give attrition value of the cutting tool directly and we can also make sure the quality of work piece and can make sure that the cutting tool monitoring in the finish machine is come true by combining it with the online and real time compensation of processing precision. This technique has also some drawbacks. For example environment can easily disturb the real time measurement. Also cutting compound and chips may affect the measured results. Some other aspects like thermal expansion and run out and vibration may affect the precision in the measurement during the process cycle. Also when cross section work pieces are processed it requires the exact positioning tracking of the sensor which may also cause errors and hence increase difficulty in implementation.
- **Cutting Temperature Measurement**
  Cutting Temperature is an important and crucial phenomenon. The temperature could ascend promptly with the increase in the abrasion of the cutting tool. There are three different methods to measure the cutting temperature. In first method which is known as cutting tool and work piece thermocouple we measure the average temperature in the cutting area with different standards among different work pieces and cutting tools. The second method is thermocouple of using two types of metal filaments fixed on a particular point inside the cutting tool. The cutting temperature can be measured on this particular point. There are also some problems in using this method like when the temperature is changed the response time is very slow. Also a lot of time is wasted while preparing this at start. The third and last method is to set an infrared camera operating system to determine the cutting temperature field distributions. It has very short response time and high sensitivity. The drawback of this method is that instrument is very complicated and hard to focus which makes to diagnose the cutting tool temperature in the cutting field difficult.
- **Vibration Frequency Measurement**

Due to the existence of the friction between the work piece and cutting tool edge different vibration frequencies can be emerged during the process of cutting. There are two basic techniques to monitor the vibration during the cutting process. The first technique is to divide the amplitude into high and low parts and comparing them during the cutting process. The second technique is to separate the amplitude into many independent spectrum bands and then record and analyze these spectrum bands during the cutting process by the help of a computer system. By using this we can also monitor the wear extent of the cutting tool.

- **Work pieces Surface Roughness Measurement**

  The surface roughness of the work piece bumps up rapidly while attrition rate of the cutting tool increases. Hence we can evaluate the wear extent of the cutting tool indirectly by using this point of view. This is an efficient technique. We need sample demarcation in advance in this technique. This method still could not put into practice because it is affected by cutting emulsion, chips, article texture, vibration and other cutting factors.

These techniques have both advantages and disadvantages according to the different conditions and situations. Different methods have different precision rate according to the situation. Although some techniques have good precision rate but these techniques are still not really good methods as a whole for monitoring the cutting tool. Many of them have just used in laboratories and have a very long time to go into the industrial use.

1.1.2. Development trend of cutting tool monitor

There are still some issues present in current existed cutting tool monitoring methods. That is why these methods could not satisfy all the practical needs of quick response, reliability, robustness and some other performance issues. Therefore it is very important to find out some other useful techniques to improve the performance and to get the improved and good results of other issues. Some important trends are presented below:

- **First Trend**

  Techniques which are based on fusion of cutting tool monitor. For example sensor integration, multi sensor fusion and decision making on signal processing.

- **Second Trend**

  Techniques which are based on signal processing of cutting tool monitor. The main purpose of using the signal processing is to abstract the important and useful information related to status of the cutting tool. More commonly wavelet technique is used to monitor the condition or status of the cutting tool.

- **Third Trend**

  These techniques are based on intellectual technology of cutting tool monitor. The conventional methods in these techniques are fuzzy inference, neural network, generic algorithm and expert systems.

- **Fourth Trend**

  These techniques are based on the integration of the cutting tool monitoring systems, cutting tool management system, numerical control machining transmission system and ERP systems. The main requirement for these techniques is that the cutting tool monitor systems should be able to transmit the status report of the cutting tool monitor to the cutting tool information management system just in time and meanwhile share the monitor data and control the cutting process automatically through the NC system.

1.2. Problem Statement

How to implement a prototype system that is able to monitor the working condition of the milling tool using acoustic emission signals from the milling machine passively collected during operation.

1.3. Purpose

The main purpose of this thesis work is to monitor and analyze the sound data collected from the milling machine, and identify the features in the sound signal. This will be beneficial for the company in many ways. We will get maximum utilization of the cutting tool. It is also useful because we have no need of a person who is watching the machine continuously to detect whether the tool is worn-out or not. Hence burden on the workers is very less. The most important purpose of this thesis work is to improve the performance and quality of the work.

1.4. Demarcations

In this thesis work reader can find about signals and its different types. The reader can also know the basic concept of signal processing. We have also tried to give an over view of case base reasoning and filters along with the different types of filters. It is also described in this thesis work that what problems may occur during the process of monitoring the condition of the cutting tool. Also comparison is shown in the graphs about different conditions of the cutting tool. The detailed analysis of the sound signals is also described in this thesis work.

2. Methods

We have adopted different research and development strategies for Development of a prototype system for milling machines. In research area we have consulted different books, read some papers and study the material which is available on the web regarding the collection of data, processing the data and then presenting that data. Also we have consulted some experienced persons who have the good domain knowledge to resolve the issues in analyzing the data.

2.1 Case based reasoning

The Process of solving new problems based on the solutions of similar past problems is called Case-Based Reasoning (CBR). [4]

Case-Based Reasoning (CBR) is a four-step process. [5]

- Retrieve: In this step we give a target problem and retrieve cases from memory relevant to solving it. A case includes a problem, as well its solution, annotations about how the solution was derived.
- Reuse: In this step we map the solution from the previous case to the target problem. This might involve adapting the solution as needed to fit the new case.
- Revise: Having mapped the previous solution to the target case, testing of the new case in the real world or a simulation will be implemented. If necessary, the solution in the new case will be revised.

- Retain: When the solution has been successfully adapted to the target problem, the resulting experience should be stored as a new case in the memory.

These four steps are called a CBR cycle. Therefore we can simplify this mental process to describe CBR typically as a cyclical process comprising the four REs: Retrieve the most similar cases, Reuse the cases to attempt to solve the problem, Revise the proposed solution if necessary and retain the new solution as a part of a new case.

Currently the CBR cycle rarely occurs without human intervention. For example, many CBR tools just play a role in case retrieval and reuse systems. Case revision, namely adaptation, is usually undertaken by managers of the case base. But it is not a weakness of CBR. It encourages human collaboration in decision support. The figure below describes clearly how each step is handled in the CBR cycle. [6]

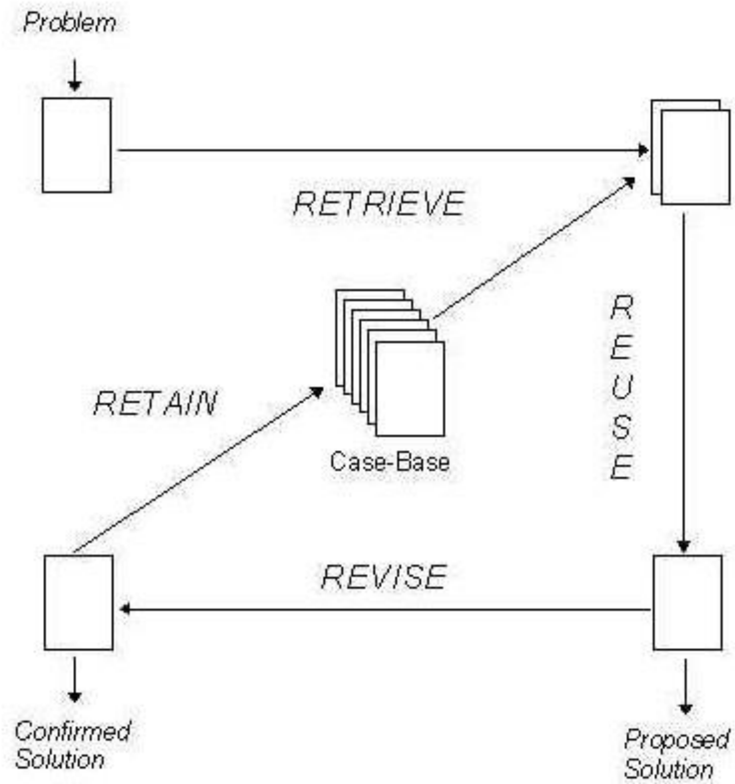


Figure the CBR Cycle

Case-based reasoning (CBR) is a method in the Artificial Intelligence which we will use in the upper-layer in the building of this project. It not only saves the time of working out a problem just looking into the case library and find out the most similar case to the current case and give out the solving methods of the new case, but also release the expert from all the trivial problem to deal with some big and important problems, which means it is very good for companies just hiring some low-cost and inexperienced workers to be in the front desk to solve some problems just searching up in the case library, and it certainly save a lot of money for the company to employ so many experts everywhere.

There are many advantages of CBR over rule-based reasoning through which we can see the perspective of this new kind of method. These advantages are described below: [7]

- CBR systems can be built without passing through the knowledge elicitation bottleneck, since elicitation becomes a simpler task of acquiring past cases.
- CBR systems can be built where a model does not exist.
- Implementation becomes a simpler task of identifying relevant case features.
- A CBR system can be rolled out with only a partial case-base which means using CBR, a system need never be complete, since it will continually growing. It removes one of the troubles of rule-based systems how to tell when a system is complete.
- CBR systems can propose a solution quickly.
- Individual or generalized cases can be used to provide explanations that are sometimes more satisfactory than explanations generated by chains of rules.
- CBR systems can learn by acquiring new cases, making maintenance easier.
- Finally, by acquiring new cases, CBR systems grow to reflect their organization's experience which rule-based systems cannot.

Hence there is no doubt that this kind of expert system can be very successful in the future. And that is why we choose case-based reasoning in the upper-layer of this project.

3. Previous Work

A thesis work was done at Berg Propulsion on hub body. They have used very simple equipments for data collection and analyzing. They used a laptop and a simple microphone for collecting data. They used wave-lab for recording and analyzing for sound. Wave-lab is very powerful tool for sound recording from which you can split big file into small ones. So, we can adjust file size according to the needs. They were not able to put the microphone inside the cabin as there are a lot of risks involved. They cutting tool become hot and to cold down the cutting tool, they use cooling compound to cool down the material. Also, small hot chips of cutting material which were scattered everywhere in the cabin might affect the microphone. That is why, they had placed microphone outside the cabin. The distance of microphone from cabin was 1 to 2 metres to the cutting tool. They used simple microphone which we used for speaking in daily life. They recorded frequencies between 0 Hz to 22 kHz, as most frequencies lies between these ranges. As, all sounds are recorded from all directions from microphone, so it was difficult to distinguish which sounds were noises and which were useful. So, they needed to filter out useful sounds from collected data.

The process of hub making is very long. So, they collected sounds for eight hours. They set sample rate of 44.1 KHz for sound signals. It is normal sample rate which they used to show the entire process of hub body in Berg Propulsion. Normally this sample rate gives good results.

They selected midsized hub body which takes little less time as compared to large sized hub. Wave-lab has good feature which can splits big files into small ones. They needed to stop the recording and then record again as when operator needs to change the cutting tool. During the whole process, they split the whole eight hours recording into twenty nine separate sound files based on different cutting stages.

Milling process of hub body could be divided into two main parts. The first part is the first cut. Because in the beginning, material will be very coarse and it needs to be smoothen a little. So, they do a rough first cut by cutting tool to smooth hub body a little. The second part is last cut. It is used to make hub body surface smoother. It should be noted that the first cut should not be completely done before the start of second cut.

The first cut is also divided into further two parts; the top half part of the hub body and the remaining half part of the hub body. When the first cut of first half part will be finished, the cutting tool will be changed to the final cut of the remaining half part of the hub body.
For analysis of sound recordings they used Fast Fourier Transform to draw frequency domain model. Then, they analyzed different frequencies how they were behaving with sound amplitudes. They have used software Audacity to analyze Fast Fourier Transform.
To analyze sound they used window size of different sizes. It would give us clear picture which windows size was giving us more precise information of sound recordings.
First they have set window size of 100ms. They have chose file2 to do Fast Fourier Transform. They had chunk of a file from thirty seconds to thirty six seconds for analyzing. You can see the output in the file below. Only 30.1 to 30.2 second is shown in the figure. There we can see the sound waves in Fast Fourier Transform. Then, they have changed the window size to 200ms and 500ms and you can see the pictures below.

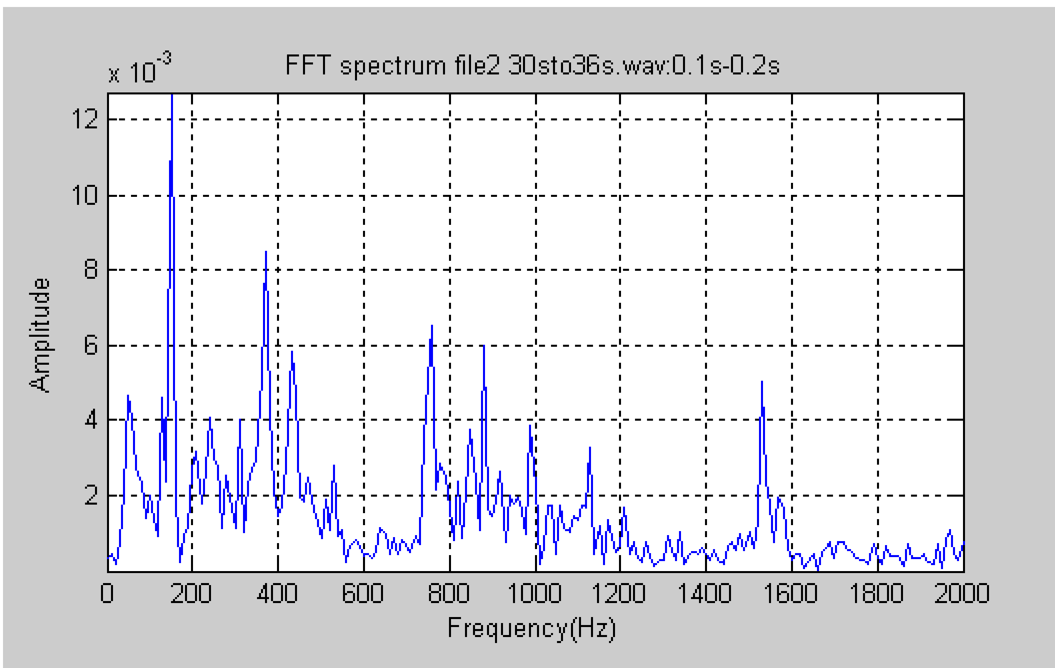


Figure 1 100ms Window Size

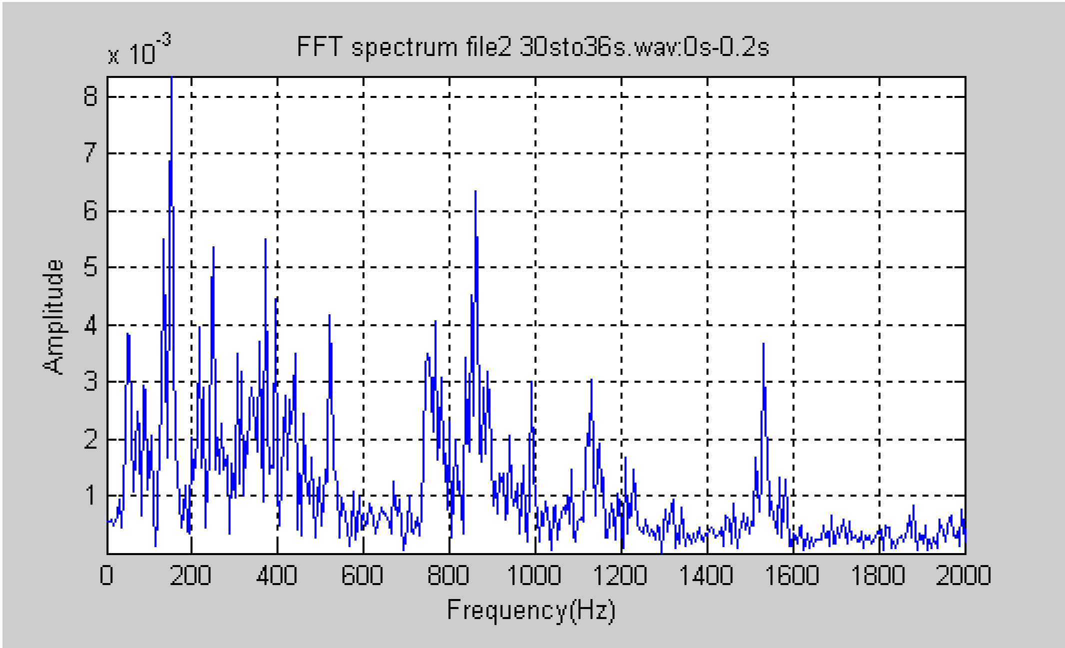


Figure 2 200ms Window Size

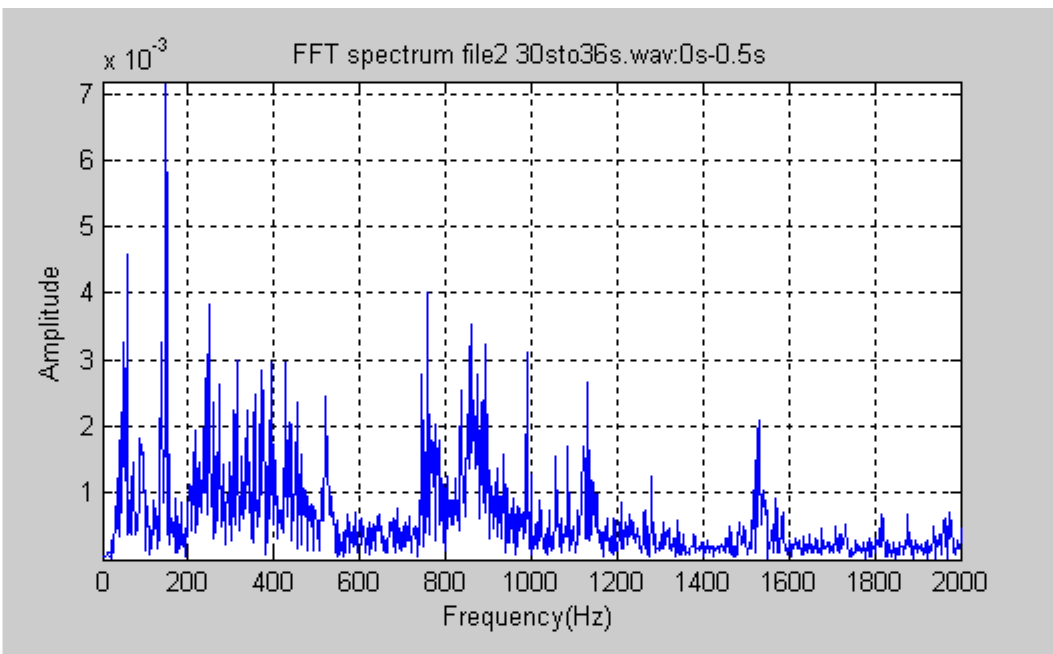


Figure 3 500ms Window Size

Their figures showed us that the main peak points are same in all the figures. If we check frequency around 800 MHz, it is similar in all the pictures among window size of 100ms, 200ms and 500ms. First, they have selected 100ms window size. As, 100ms windows size give them more precise information of sound recordings. There, we can see clearly the amplitude of the sound with respect to frequency. As, window size 200ms and 500ms give them same information, but now they chose the window size 100ms which is not too small as compared to human sound or too large as 500ms window size which might affect the processing speed while dealing with large amount of data.[2]

4. Basic Concept

4.1.1 Signals

In general a signal is any spatial varying or time varying quantity.

In Physical world, any quantity which is measurable through time over the space could be taken as a signal.[3]

There are two main types of signals.

4.1.1.1 Analogue Signal

Any continuous signal for which the time varying feature (variable) of the signal is representation of some other time varying quantity is called an analogue signal.

An image of analogue signal is shown below but all the analogue signals do not vary as smoothly as shown in the figure.

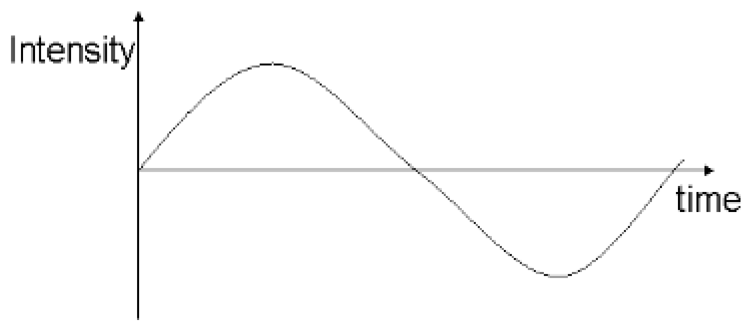


Figure of Analogue Signal

We should consider some advantages and disadvantages of the analogue signals for using them. These advantages and disadvantages are described below:

Advantages:

The fine definition of the analogue signal has the potential for an infinite amount of signal evolution. Analogue signals are of higher density than digital signals.

The processing of analogue signals could be achieved more simply and easily than with the equivalent digital signals.

An analogue signal could be processed directly by analogue components, though some processes are not available except in digital form.

Disadvantages:

Any system of analogue signalling has noise, for example random unwanted variation.

As the signal is copied and re-copied, or transmitted over long distances, these apparently random variations become dominant.

Although the resolution of an analogue signal is higher than a comparable digital signal, the difference can be overshadowed by the noise in the signal. Most of the analogue systems also suffer from generation loss.

4.1.1.2 Digital Signal

The signals which are non continuous are called digital signals. They change in individual steps and consist of digits or pulses with discrete values. The value for each pulse is constant but there is an abrupt change from one pulse to the other next. These signals have two amplitude levels called as nodes and represented by 1 or 0, True or False and High or Low. An image of digital signal is shown in the figure below:

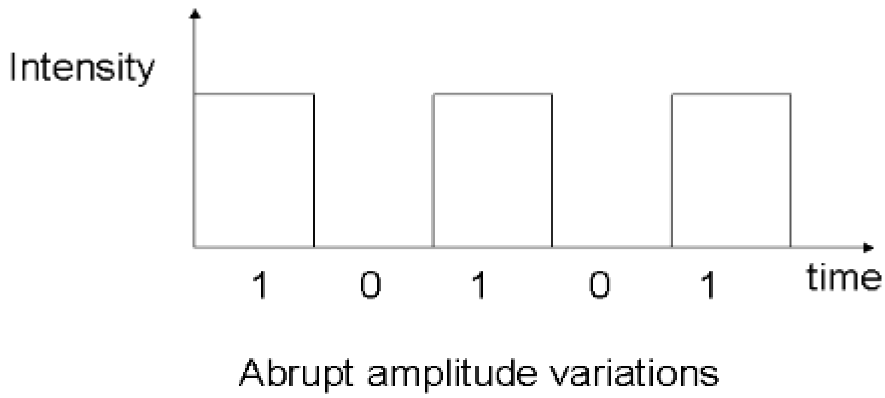


Figure of Digital Signal

Digital signals consist of patterns of bits of information. These patterns could be generated in many different ways, each producing a specific code. Modern digital computers also process and store all kinds of information as binary patterns. All the texts, sounds, pictures and videos are stored in binary values in these computers.

The main advantage of digital signals over analogue signals is that the precise signal level of the digital signal is not vital.

4.2 Fast Fourier Transform

History

Gauss has described about critical factorization step at 1805. FFTs (Fast Fourier Transform) were first discussed by Cooley and Tukey (1965). A discrete Fourier transform can be computed using an FFT by means of the Danielson-Lanczos lemma if the number of points N is a power of two. If the number of points N is not a power of two, a transform can be performed on sets of points corresponding to the prime factors of N which is slightly degraded in speed. An efficient real Fourier transform algorithm or a fast Hartley transform (Bracewell 1999) gives a further increase in speed by approximately a factor of two. Base-4 and base-8 fast Fourier transforms use optimized code, and can be 20-30% faster than base-2 fast Fourier transforms. Prime factorization is slow when the factors are large, but discrete Fourier transforms can be made fast for N = 2, 3, 4, 5, 7, 8, 11, 13, and 16 using the Winograd transform algorithm. [8]

Definition

A fast fourier transform is an efficient method of estimating and computing frequency spectrum of a signal. [9]

Another definition of FFT

A Fast Fourier Transform is an efficient algorithm to compute the discrete Fourier transform which is used as DFT for short and its inverse. [10]

The DFT transforms N discrete time samples to the same number of discrete frequency samples, and is defined as: [11]

$X(k)=\sum n=0N-1x(n)e-(i2\pi nkN)$

An FFT computes the DFT and produces exactly the same result as evaluating the DFT definition directly; the only difference or advantage is that an FFT is much faster. In the presence of round-off error, many FFT algorithms are also much more accurate than evaluating the DFT definition directly, as discussed below.

Let x_0, ..., x_{N-1} be complex numbers. The DFT is defined by the formula

$$X_k = \sum_{n=0}^{N-1} x_n e^{-i2\pi k \frac{n}{N}} \qquad k = 0, \dots, N-1.$$

Evaluating this definition directly requires $O(N^2)$ operations: there are N outputs X_k, and each output requires a sum of N terms. An FFT is any method to compute the same results in $O(N \log N)$ operations. So we could say that the fast Fourier transform (FFT) is a discrete Fourier transform algorithm which reduces the number of computations needed for N points from $2N^2$ to $2N \lg N$, where lg is the base-2 logarithm.

There are many type of algorithms used in FFT. But far the most common algorithm for FFT is Cooley-Tukey Algorithm.

4.3 Signal Processing

The process of extraction of information from complex signals in the presence of noise. Generally by the conversion of the signal into the digital form followed by analysis using different algorithms is called signal processing. It is also called digital signal processing (DSP).

We can also define the signal processing as:

Signal processing is an area which deals with operation on or analysis of signals, in either discrete or continuous time to perform useful operations on those signals. [12]

4.3.1 Filters

In signal processing, the device or process which removes some unwanted features or components from a signal is called filter.

Filtering is a class of signal processing, the defining feature of filters being the complete or partial suppression of some aspects of the signal. Most often, this means removing some frequencies and not others in order to suppress interfering signals and reduce background noise. However, filters do not exclusively act in the frequency domain; especially in the field of image processing many other targets for filtering exist.

One of the most important functions of filters is to allow signals in some specific part of the frequencies getting through while signals in other part of the frequency periods are being limited. Essentially the filter is a frequency selection circuit.

We could classify filters on many different bases and these overlap in many different ways. There is no simple hierarchical classification. Filters could be: [13]

- Analogue or digital
- Linear or non-linear
- Time-invariant or time-variant
- Discrete-time(sampled) or continuous-time
- Passive or active type of continuous-time filter

- Infinite impulse response (IIR) or finite impulse response (FIR) type of discrete-time or digital filter.

Some definitions of useful filters for this project are described below:

4.3.1.1 Analogue Filters

Analogue filters are a basic building block of signal processing which are much used in electronics. Amongst their many applications are the separation of an audio signal before application to bass, mid-range and tweeter loudspeakers; the combining and later separation of multiple telephone conversations onto a signal channel; the selection of a chosen radio station in a radio receiver and rejection of others. [14]

4.3.1.2 Digital Filters

In different fields like electronics, computer science and mathematics, a digital filter is a system which performs mathematical operations on a sampled, discrete-time signal to enhance or reduce certain aspects of that signal. This is in contrast to the other major type of electronic filter, the analogue filter, which is an electronic circuit operating on continuous-time analogue signals. An analogue signal may be processed by a digital filter by first being digitized and represented as a sequence of numbers, then manipulated mathematically, and then reconstructed as a new analogue signal. In an analogue filter, the input signal is "directly" manipulated by the circuit.

A digital filter system usually consists of an analogue-to-digital converter and a microprocessor. Digital filters may be more expensive than an equivalent analogue filter due to their increased complexity, but they make practical many designs that are impractical or impossible as analogue filters. [15]

4.3.1.3 Linear Filters

Linear filters in the time domain process time-varying input signals to produce output signals, subject to the constraint of linearity. This result from systems composed solely of components (or digital algorithms) classified as having a linear response. [16]

4.3.1.4 Non-Linear Filters

A signal processing device whose output is not a linear function of its input is called a non-liner filter. [17]

4.3.1.5 Time-Invariant Filters

A device which performs the same operation at all the time is called a time-invariant filter. It is also called shift-invariant filter. [18]

4.3.1.6 Time-Variant Filters

A system which is not a time-invariant is called a time variant system. Roughly speaking, characteristics of its output explicitly depend upon time. [19]

4.3.1.7 Active Filters

Actually active filters are a type of analogue electronic filters. These are distinguished by the use of one or more active components i.e. voltage amplifiers or buffer amplifiers. Typically this will be a vacuum tube, or solid-state (transistor or operational amplifier).

If we compare active filters with passive filters we can see that there are three major advantages of active filters over passive filters.

- Inductors can be avoided. Passive filters without inductors cannot obtain a high Q (low damping), but with them are often large and expensive (at low frequencies), may have significant internal resistance, and may pick up surrounding electromagnetic signals.

- The shape of the response, the Q (Quality factor), and the tuned frequency can often be set easily by varying resistors, in some filters one parameter can be adjusted without affecting the others. Variable inductances for low frequency filters are not practical.
- The amplifier powering the filter can be used to buffer the filter from the electronic components it drives or is fed from, variations in which could otherwise significantly affect the shape of the frequency response. [20]

4.3.1.8 Passive Filters

Passive filters are a kind of electronic filters which are made by only passive components in contrast to an active filter. Passive filters do not require an external power source (beyond the signal). Because in most cases filters are linear therefore passive filters are composed of just four basic linear elements. These elements are capacitors, indicators, transformers and resistors. Also some more complex passive filters may involve nonlinear elements or more complex linear elements like transmission lines.

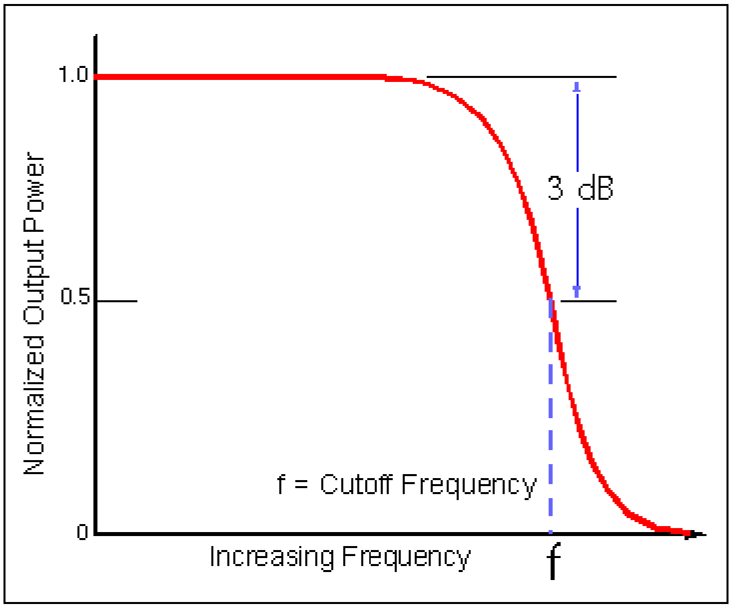
If we compare passive filters with active filters we may found several advantages which are described below:

- Guaranteed stability
- Passive filters scale better to large signals (tens of amperes, hundreds of volts), where active devices are often impractical
- No power consumption, but the desired signal is invariably attenuated. If no resistors are used, the amount of signal loss is directly related to the quality (and the price) of the components used.
- Inexpensive (unless large coils are required)
- For linear filters, generally, more linear than filters including active (and therefore non-linear) elements. [21]

4.3.1.9 Low Pass Filters

A filter which allows passing the low frequency signals and reduces the amplitude (attenuate) of the signals with the frequencies higher than the cut-off frequency is called a low pass filter. The actual attenuation amount for each frequency varies from filter to filter. Sometimes it is also called a high-cut filter or treble-cut filter while using in audio applications. The low pass filter is opposite to the high pass filter. The combination of both low and high pass filters is called a band pass filter.

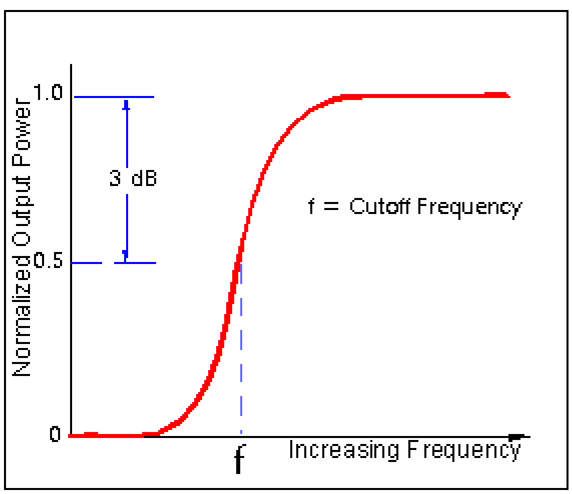
There are many different forms in which the concept of the low pass filter exists. It includes digital algorithms for smoothing sets of data, electronic circuit, blurring of images, acoustic barriers and so on. Low pass filters have the same role in the signal processing which moving averages play in some other field, like finance; both provide a smoother form of a signal that removes the short term oscillations, leaving only long term trends. A low pass filter is shown in the figure below: [22]



Low-pass Filter

4.3.1.10 High pass Filter

A filter which allows passing the high frequencies and attenuates the frequencies which are lower than cut-off frequency is called High Pass Filter. The actual attenuation amount for each frequency is a design parameter of the filter. It is also called a low-cut filter. It is called terms bass cut filter or rumble filter if used for audio application. A figure of high pass filter (LTI) is shown below: [23]



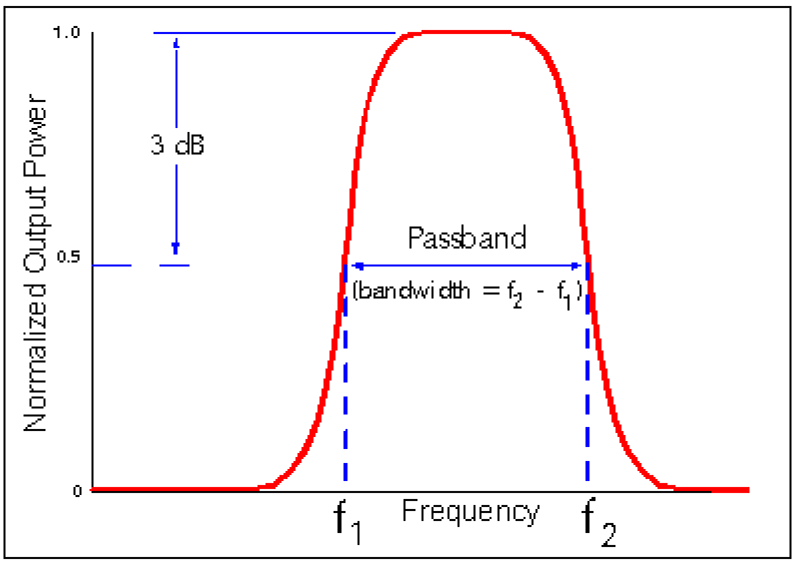
High-pass Filter

4.3.1.11 Band-pass Filter

A device which allows passing the frequencies within a certain range and rejects the frequencies outside that range is called Band-pass filter. RLC circuit (a resistor-inductor-capacitor circuit) is an example of an analogue electronic band-pass filter. These filters could also be created by combining the high pass filters with low pass filters.

An ideal band-pass filter is that which has a completely flat band-pass, e.g. with no gain/attenuation throughout, and will completely attenuate all frequencies outside the band-pass. Additionally, the transition out of the band-pass would be instantaneous in frequency. However, in practice, there is no ideal band-pass filter. Hence the filter does not attenuate all frequencies outside the desired frequency range completely; in particular, there is a region just outside the intended band-pass where frequencies are attenuated, but not rejected. This is known as the filter roll-off, and it is usually expressed in dB of attenuation per octave or decade of frequency. Generally, the design of a filter seeks to make the roll-off as narrow as possible, thus allowing the filter to perform as close as possible to its intended design. Normally, this is achieved at the expense of pass-band or stop-band ripple. [24]

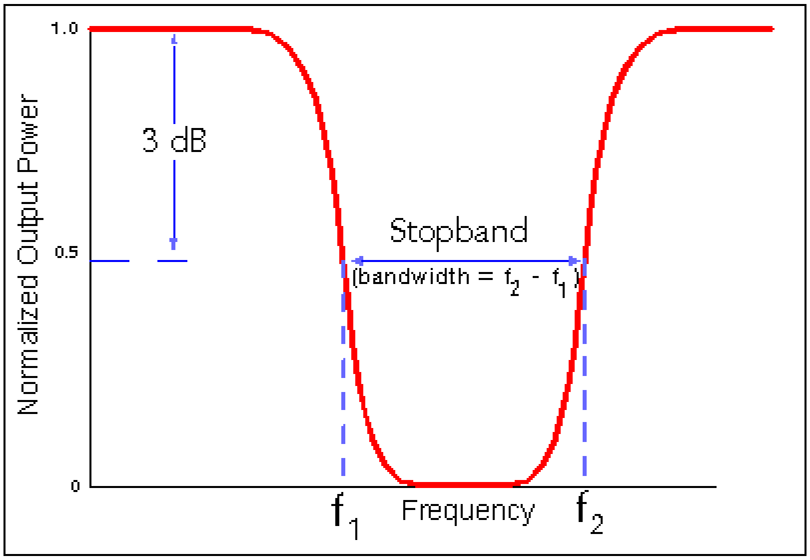
A band-pass filter is shown in the figure below:



Band-pass Filter

4.3.1.12 Band-stop Filter

A device which allows passing most of the frequencies unaltered and attenuates only those which are in specific range to very low level is called a Band-stop filter. The Band-stop filter is opposite to the Band-pass filter. This filter could be designed to stop the specified band of the frequencies but it usually attenuates them below some specified level. A Band-stop filter is shown in the figure below: [25]



Band-stop Filter

4.3.1.13 All Pass Filter

A device which allows passing all the frequencies equally and changes the phase relationship between various frequencies is called an All Pass filter. It performs this task by varying its propagation delay with frequency. Generally it is described by frequency at which phase shift crosses 90°. [26]

5. Empirical Work

In section 1.1.1, we have discussed different techniques for monitoring the condition of cutting tool for milling machines. In this section we will discuss a new technique which is based upon the sound signal analysis for monitoring the condition of the cutting tool.

5.1 Setup equipment and data collection

Previously, there was a thesis conducted on this topic. So, one of our task is to analyze sound signal data with previous data. In our thesis, we used two types of material for sound analysis, Bronze and steel. First task is to compare data between two types of recordings; bronze and steel. Second, we compare current sound signal data with previous conducted thesis sound signal data.

5.1.1 Equipment

Equipments used for sound signal collection and processing are very simple. We need a laptop which should have good processor (Core 2 Duo), RAM (1 GB) and hard disk (200 GB), an Omni directional microphone and USB sound card.

5.1.2 Software

Audacity and Wave-lab 6 are installed on the laptop. Audacity is open source software which has a good reputation for sound recording and signal analysis. We used Audacity for recording sounds and Wave-lab 6 used for sound signal analysis. Wave-lab 6 is very well known sound editing tool.

5.1.3 Microphone Location Adjustment

Before recording, we have to set appropriate location for microphone for recording. Microphone is place at distance of 245 mm from cutting tool. When cutting tool cut the material small particles will scattered around. So, we put metal plate over microphone to protect it.

5.1.4 Material

We have experimented with two types of materials.

1. **Bronze**
2. **Steel**

These materials are not usable for the company. Remember these recordings are not any process of cutting. We just took it because company representative asked us to do. First, we experiment on bronze material then same process is repeated on steel material. The dimensions of materials are given below:

Bronze

Height: 85mm

Breadth: 97mm

Steel

Height: 105mm

Breadth: 104mm

We have not set recording sample rate frequencies in Audacity. Because Wavelab 6 allow setting frequencies ranges. We set the sample rate 44.1 KHz for entire recordings. Keep a thing in mind, microphone is Omni directional, this will record all surrounding sounds.

We have collected small recording for this time. We took recordings on four state of cutting tool.

1. New
2. Little Used
3. Bad
4. Worn out

Note: These four states of cutting tool are not of one cutting tool.

5.2 Experiment One

The purpose of taking four type of cutting material is that when cutting is done on big propellers, cutting tool becomes wear out about four hours. Also, the machine used to recording is ***Delta HG-660 X 2200.*** You can see picture of the machine below:

Figure 4

All recordings consist of not more than two minutes. It is very small process as compared to hub body process. In this process, we record the upper cut of the surface of the bronze material with different states of cutting tools as mentioned above. Operator change the cutting tool after cutting process is over and we immediately stop the recording. We saved different recordings of different states of cutting tools. So, it will be relatively easy to compare data with previous thesis data. All recording is saved in appropriate folder. Also, same process is repeated with steel material.

5.2.1.1 Investigation of window size

The sound file collected from Berg Propulsion, we have analyzed it with Time to Frequency Graph and Fast Fourier Transform. We checked how frequencies ranges changed during the cutting process. We can also see amplitude of the sound wave with respect to time.

Window size is very important to figure out how what is amplitude of wave. For detail information we shall extract information from Fast Fourier Transform. So, deciding appropriate window size is critical.

As we know, one of the features for sound signal is chronotropic character. But for human voices, it could be stable relatively in a short period of time. In general, we could say 10ms to 30ms. Thus it could be considered as a quasi-stationary process during this short period, namely human sound signal has the short-time stationary, which also illustrated that any kind of sound signals should be analyzed and disposed within a short period of time. So the proper window size for human voices is 10ms to 30ms. We could choose the any window size in Fast Fourier Transform according to the speed or other parameters we need.

As one of the characteristics of sound signal is chronotropic. But human sound signal is stable in very short period of time. We can say it is about 10ms to 30ms. So, human voice is analyzed and disposed in very short period of time. This process is known as quasi-stationary process.

However, there are differences in the human voice compared to the machine voice. Machine sounds are always much stable than human voices. So, we can analyze which window size is more appropriate and give us more information. We choose window 100ms, 200ms and 500ms. As we describe in above section, we have recordings of two types of materials, bronze and steel. Each type has four types of recording of different cutting tool. We take sound fragment from each file as an example. We choose sound from 10 second to 10.5 second which may be varying in different window size. From below you can see all of the pictures. You can see files of window size 100ms, 200ms and 500ms.

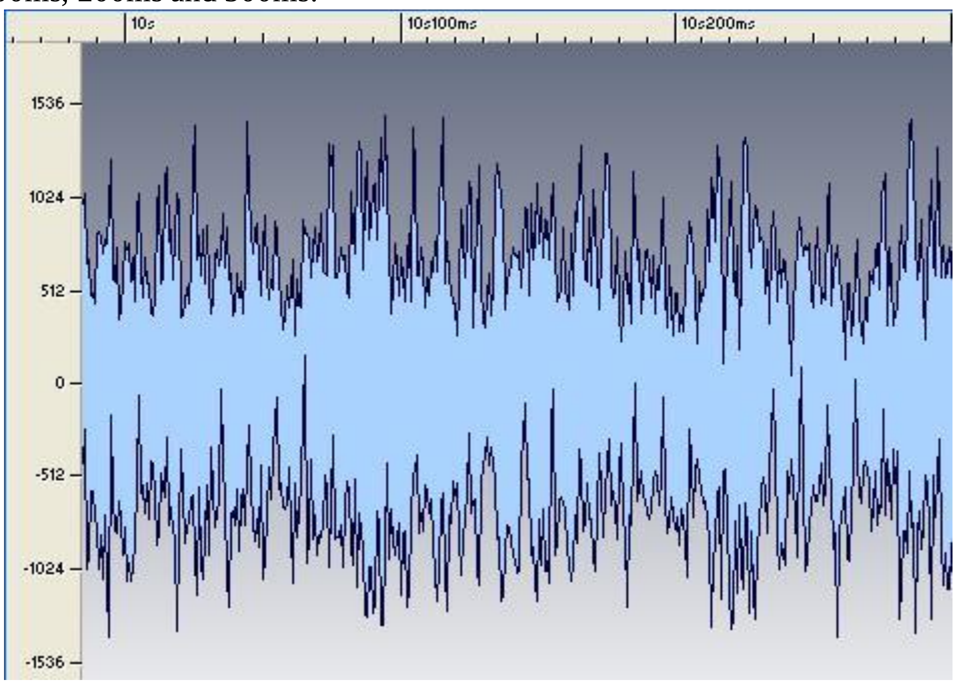


Figure 5 (a) New Bronze 100ms

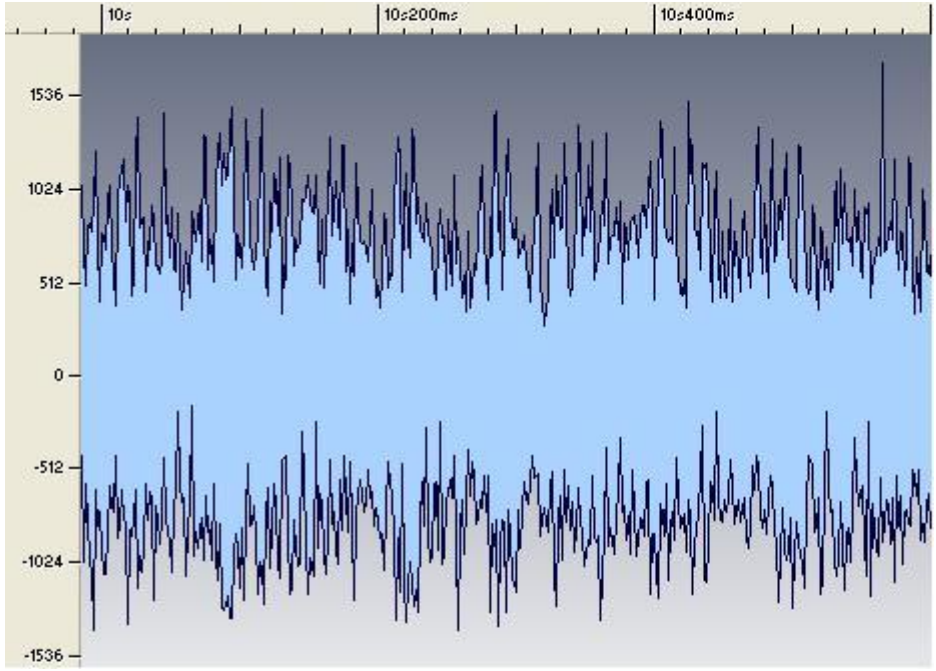


Figure 5 (b) New Bronze 200ms

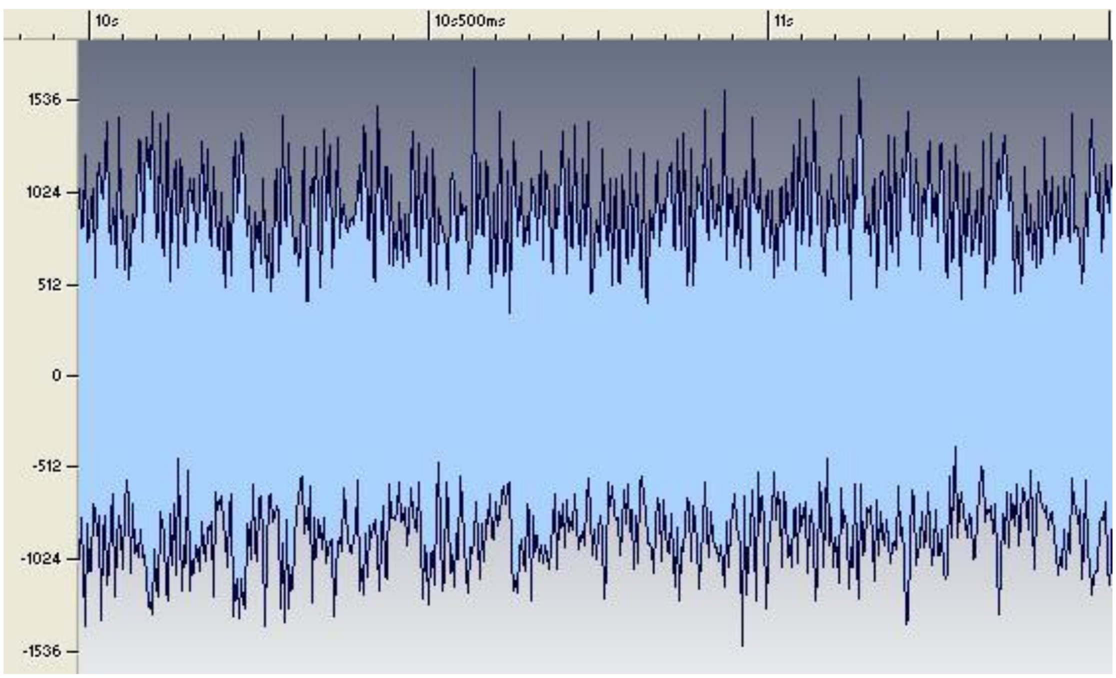


Figure 5 (c) New Bronze 500ms

Above pictures are clearly showing there is not much difference in patterns of waves if we change window size. All looks pretty same in amplitude. The important thing is to consider the amplitude value. When cutting tool is new and tool is cutting bronze material sound amplitude highest value is in between of 1024 – 1536 for positive Y axis and amplitude lowest value is in between of -1024 – -1536 for negative Y axis.

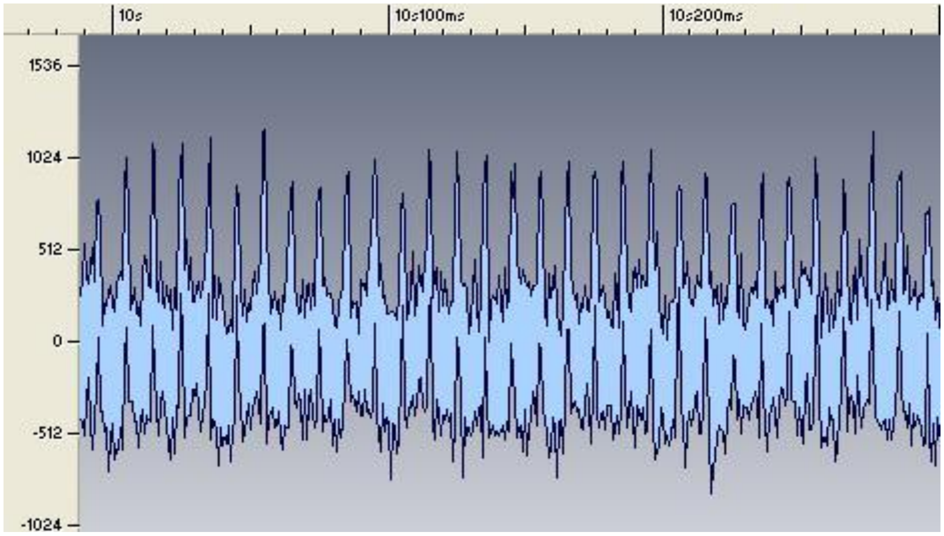


Figure 6 (a) little used Bronze 100ms

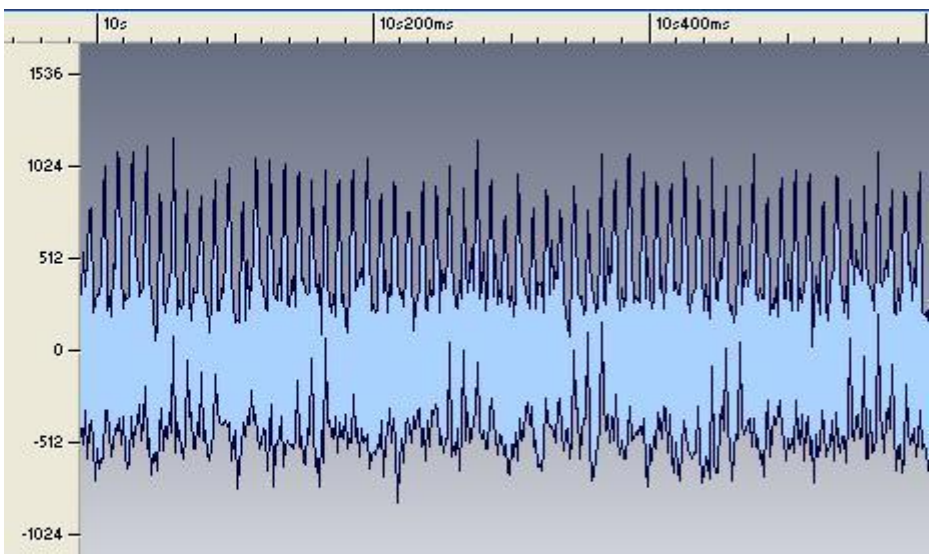


Figure 6 (b) little used Bronze 200ms

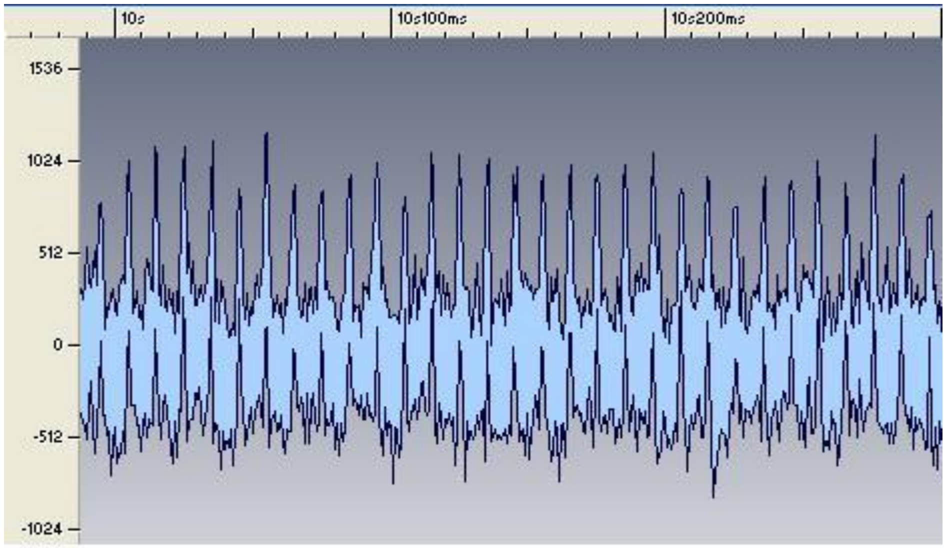


Figure 6 (c) little used Bronze 500ms

Again we can see main peak points of amplitudes are identical. The cutting tool is little used now and tool is cutting bronze material. The sound amplitude highest value is about 1024 for positive Y axis and amplitude lowest value is in between of -512 – -1024 for negative Y axis.

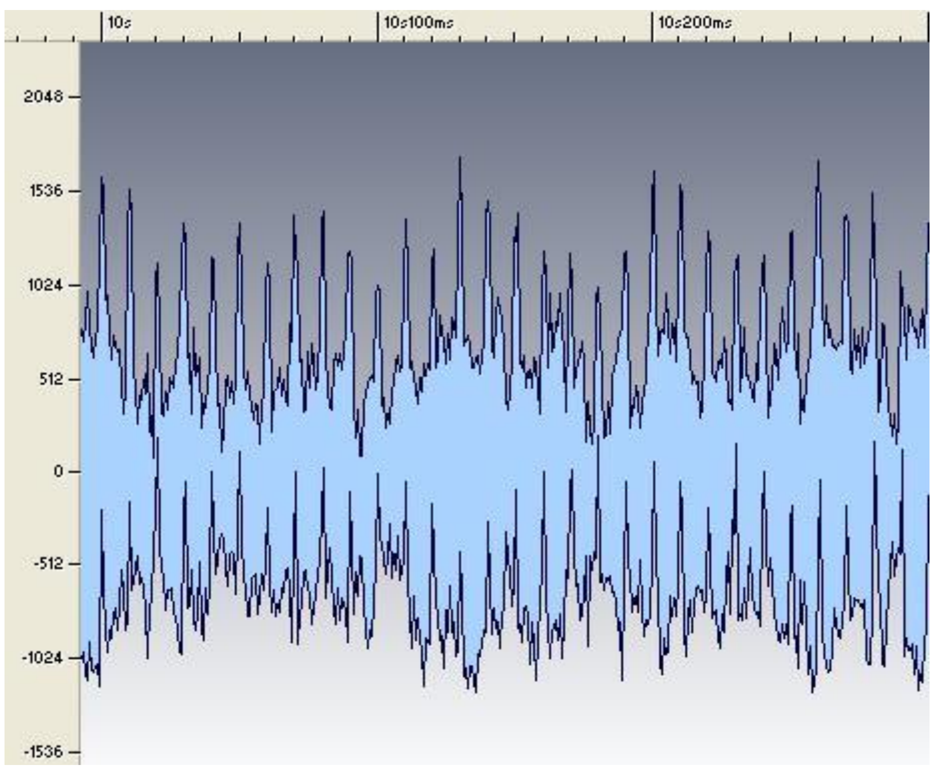


Figure 7 (a) Bad Bronze 100ms

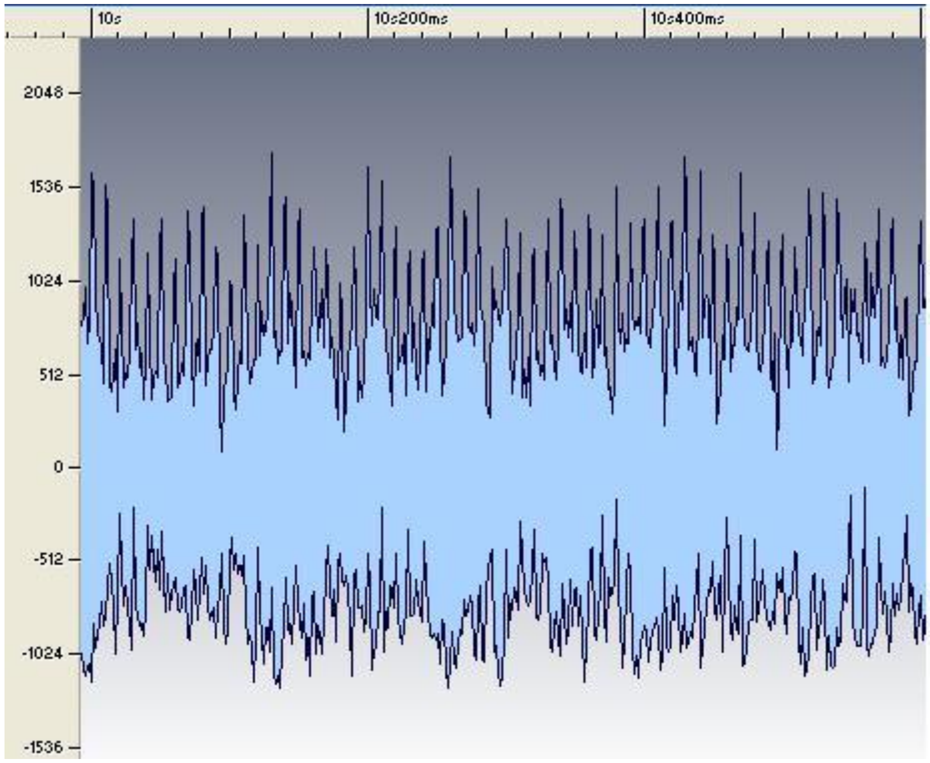


Figure 7 (b) Bad Bronze 200ms

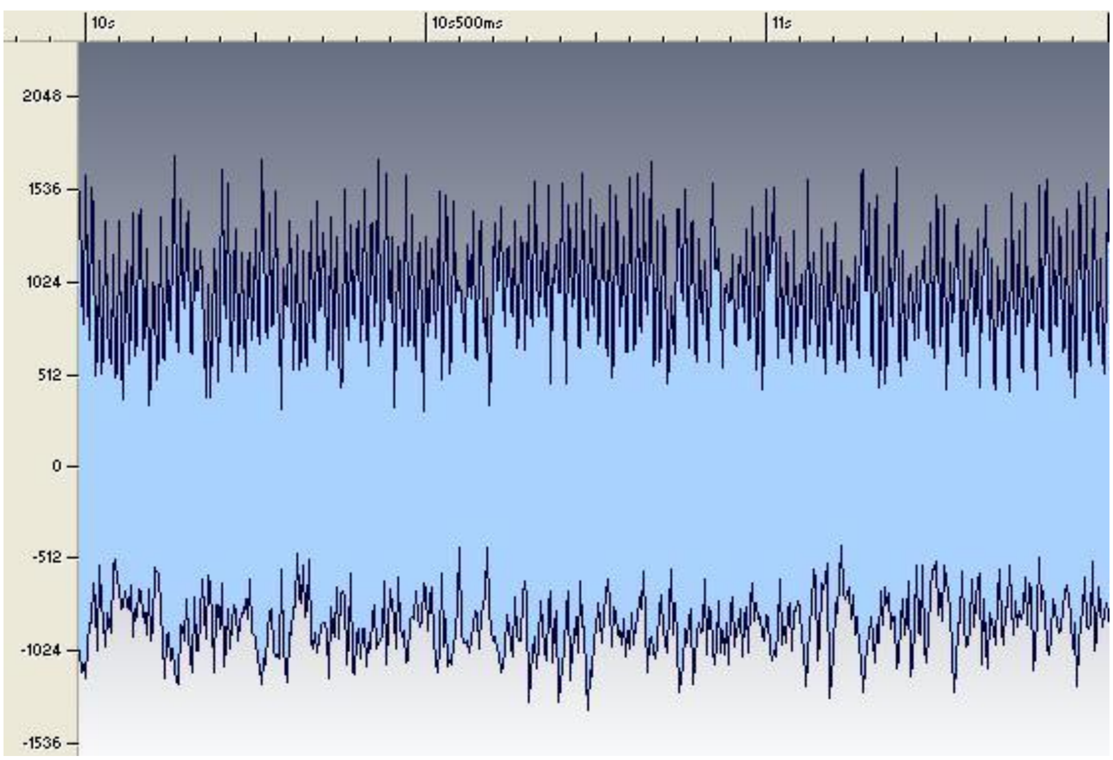


Figure 7 (c) Bad Bronze 500ms

We can see main peak points of amplitudes are identical. The cutting tool is in bad condition now and tool is cutting bronze material. The sound amplitude highest value is about 1536 for positive Y axis and amplitude lowest value is in between of -1024 – -1536 for negative Y axis.

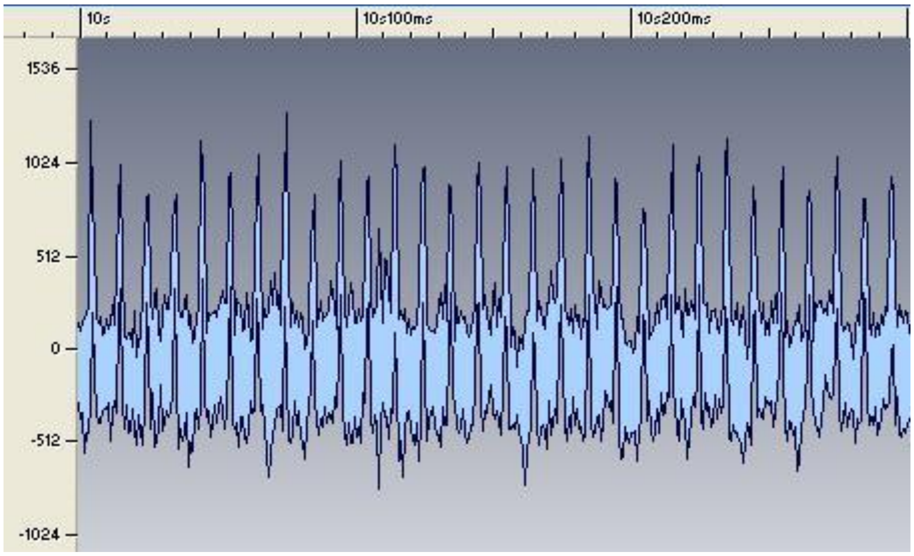


Figure 8 (a) Worn out Bronze 100ms

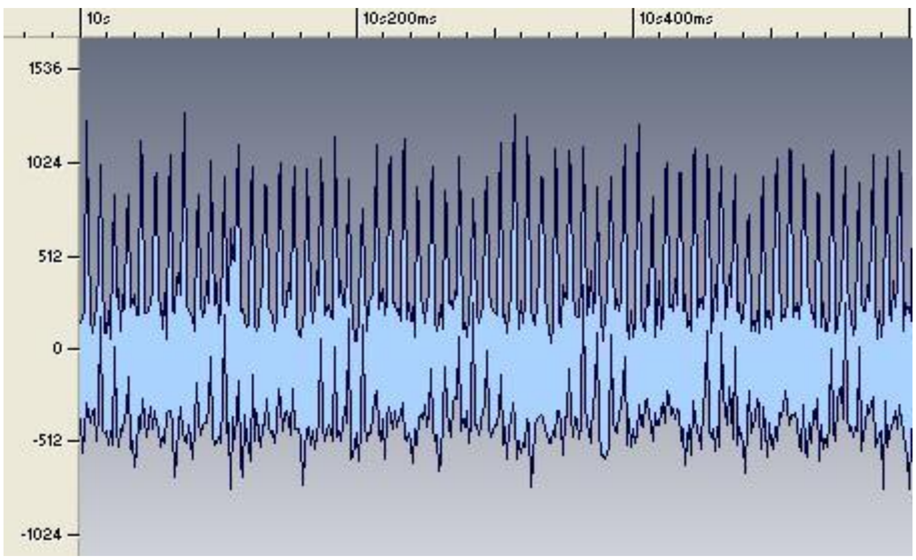


Figure 8 (b) Worn out Bronze 200ms

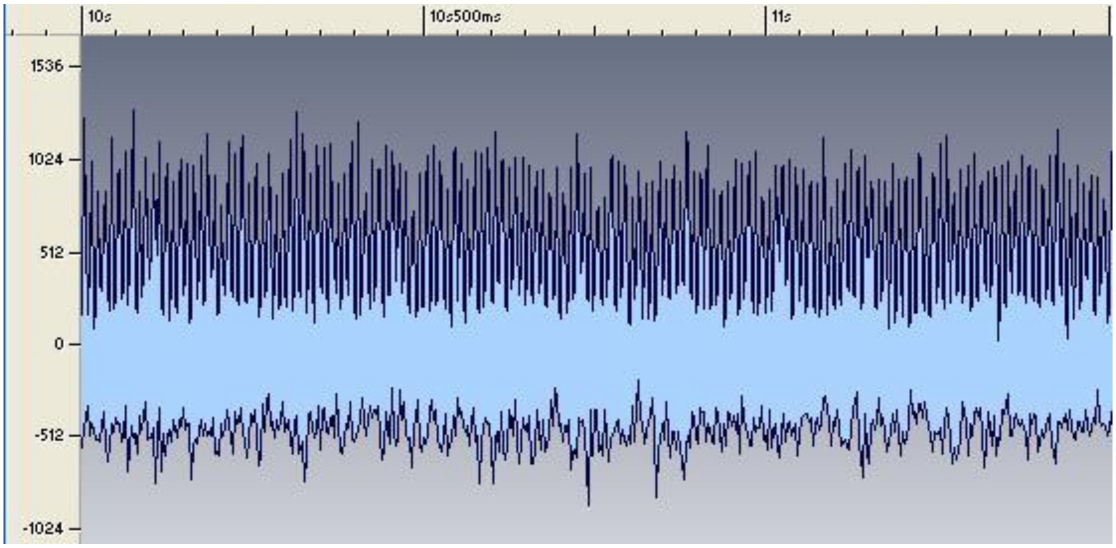


Figure 8 (c) Worn out Bronze 500ms

We can see main peak points of amplitudes are identical. The cutting tool is worn out now and tool is cutting bronze material. The sound amplitude highest value is in between 1024 – 1536 for positive Y axis and amplitude lowest value is in between of -1024 – -1536 for negative Y axis.
After examining windows size for bronze material, we can imply that it will be same with steel material. As it is shown in pictures below:

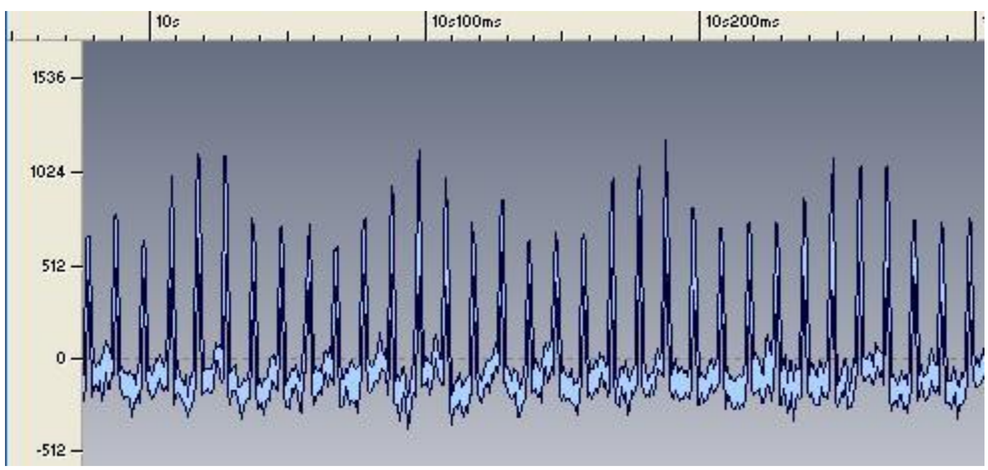


Figure 9 (a) New Steel 100ms

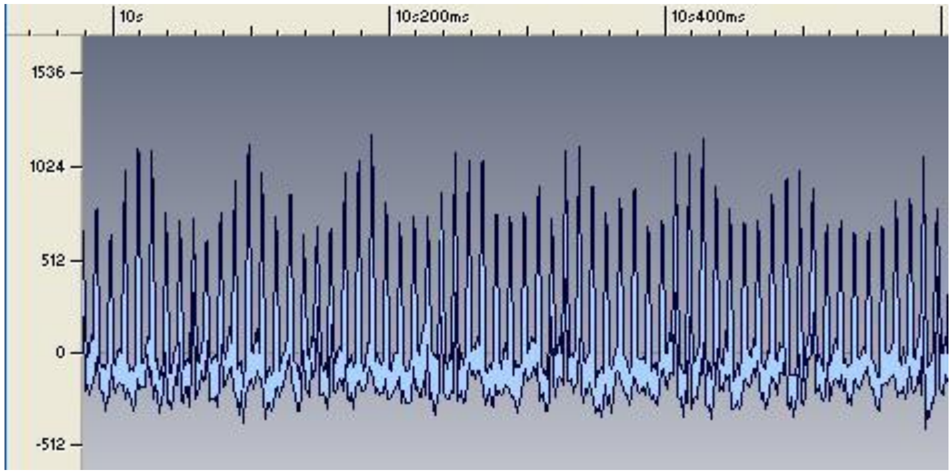


Figure 9 (b) New Steel 200ms

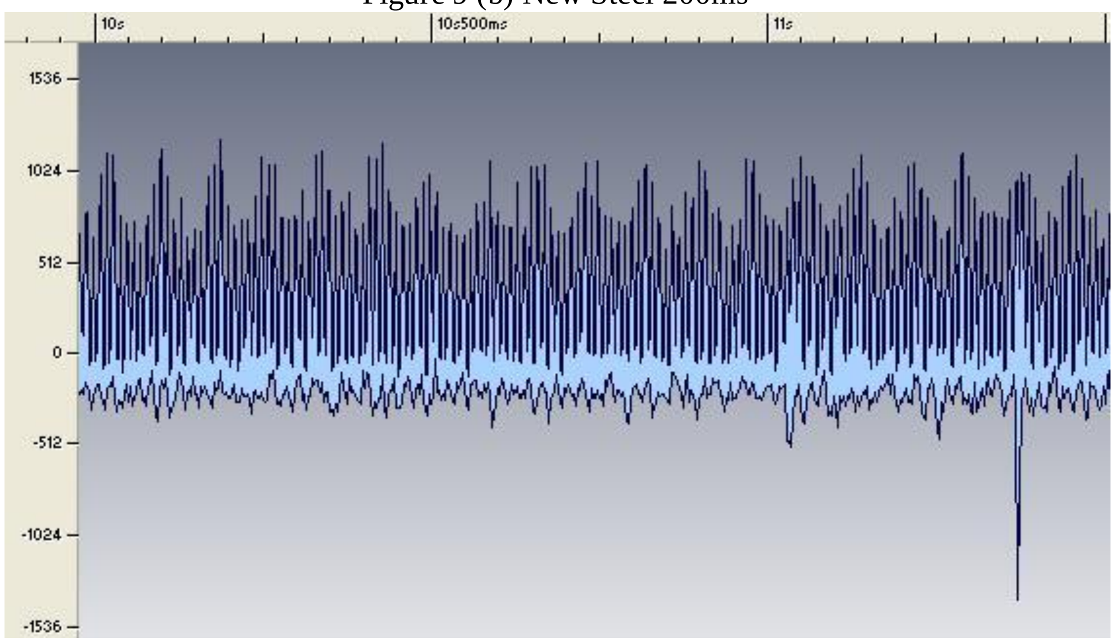


Figure 9 (c) New Steel 500ms

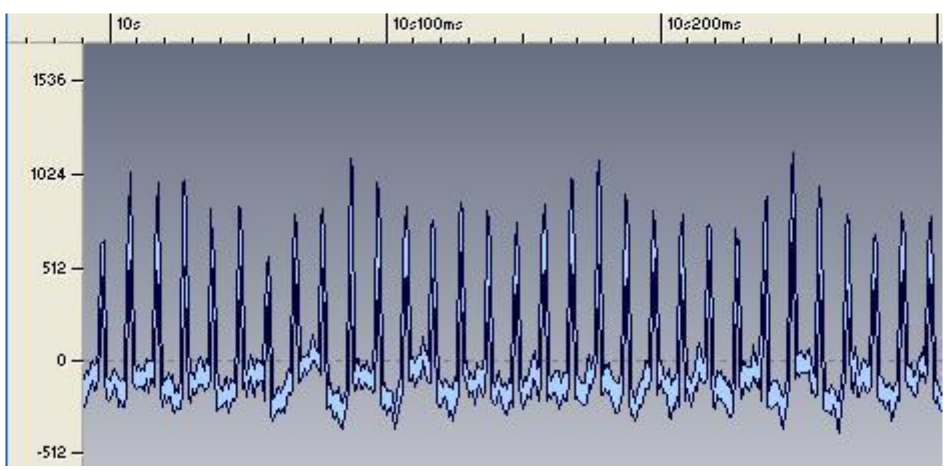


Figure 10 (a) little used Steel 100ms

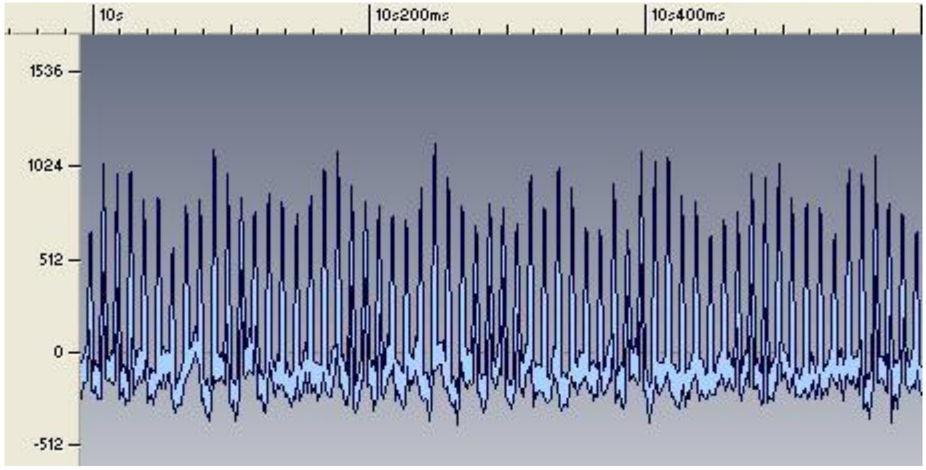


Figure 10 (b) little used Steel 200ms

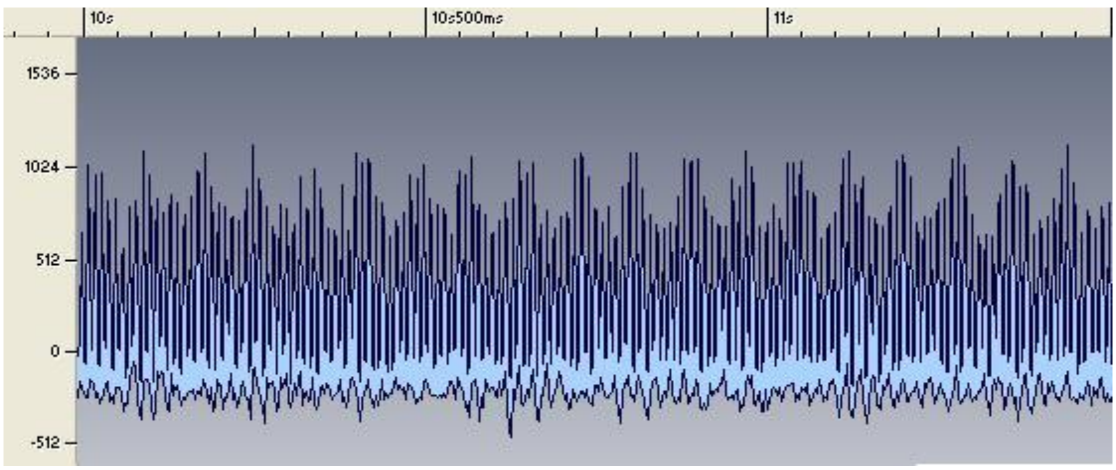


Figure 10 (c) little used Steel 500ms

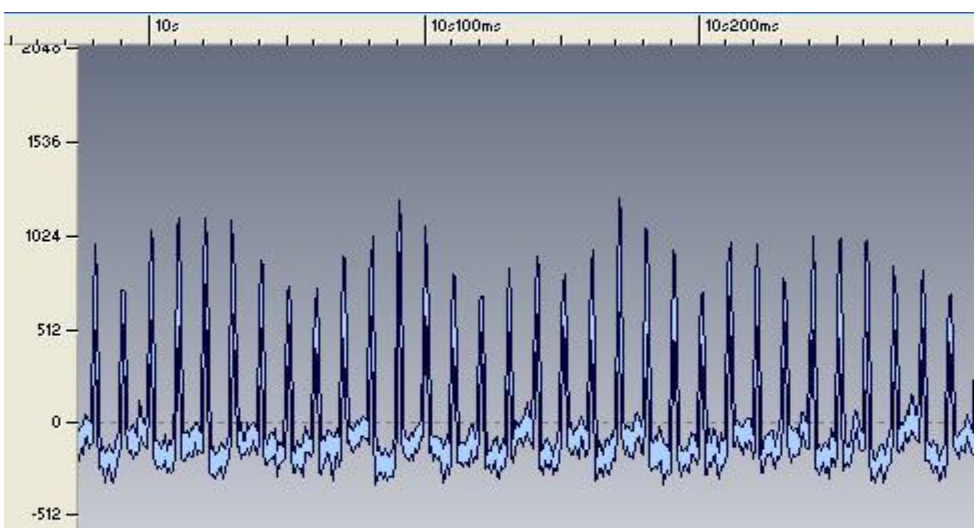


Figure 11 (a) Bad Steel 100ms

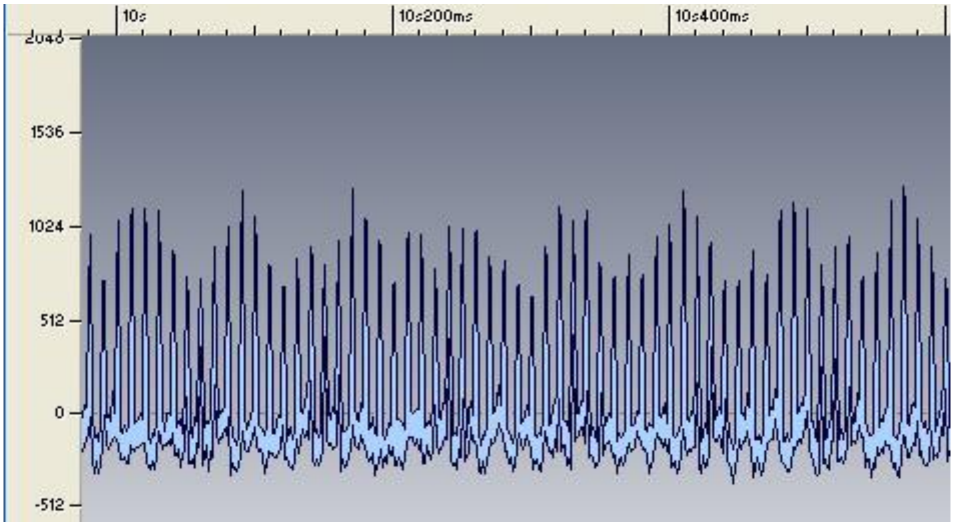


Figure 11(b) Bad Steel 200ms

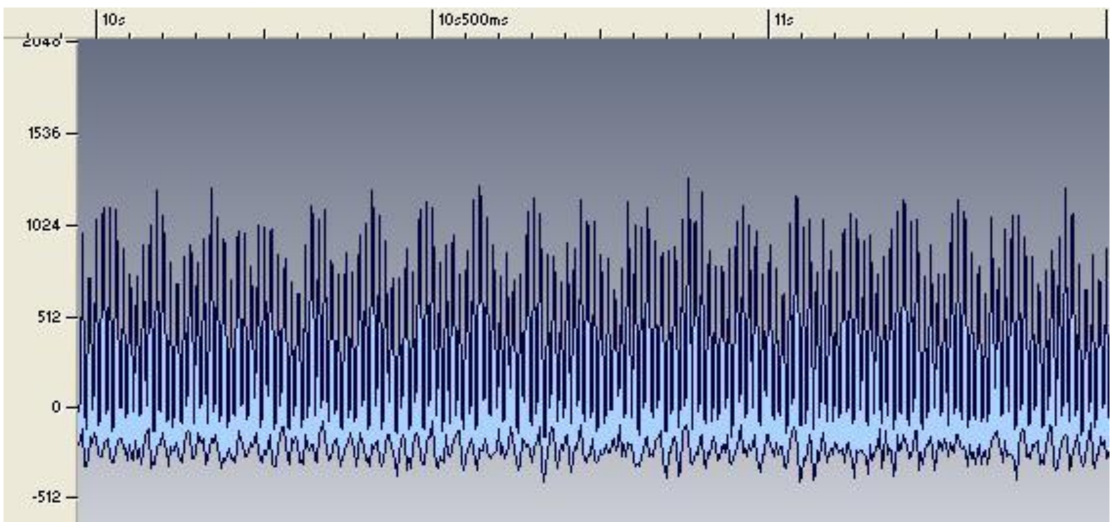


Figure 11 (c) Bad Steel 500ms

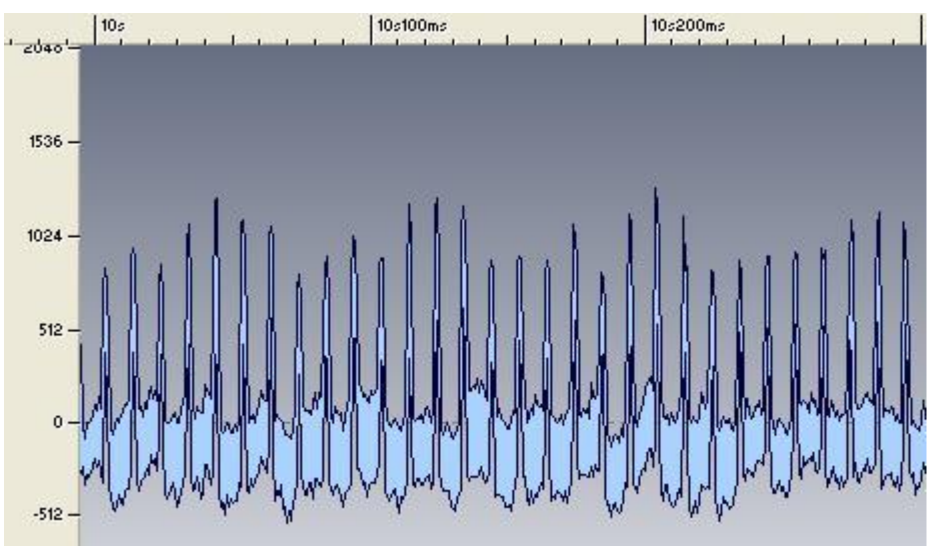


Figure 12 (a) Worn Out Steel 100ms

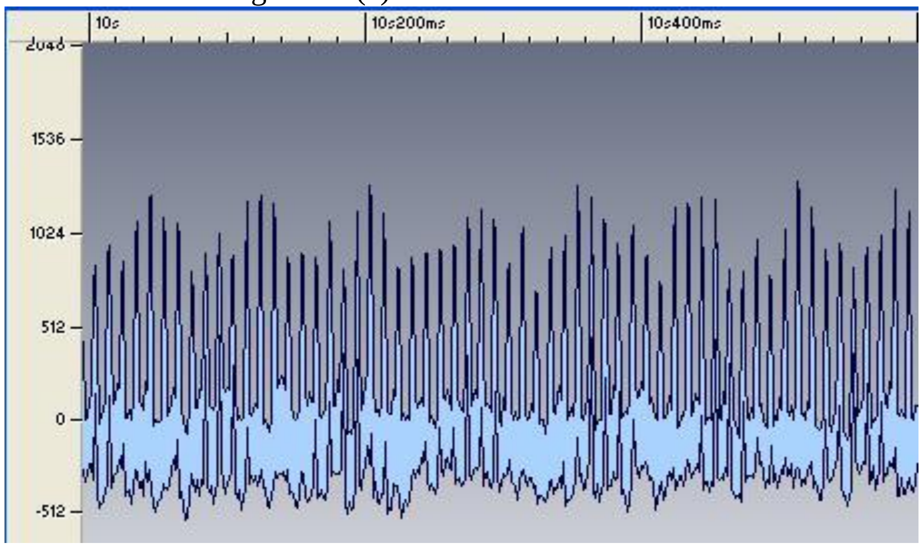


Figure 12 (b) Worn Out Steel 200ms

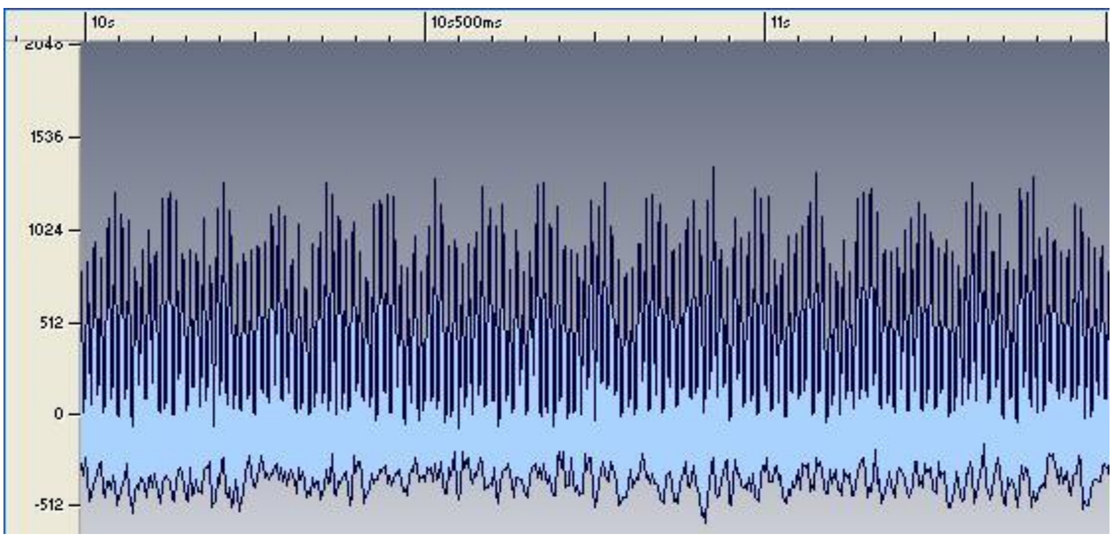


Figure 12 (c) Worn Out Steel 500ms

Here we have seen windows size of 100ms, 200ms and 500ms. We have seen every window size very thoroughly and closely. But there is not much different if we change the window size. I show some part of a sound file from 10 sec to 10.5 sec. The pattern remains the same for all window size. So, I think we can choose window size of 100ms. But it is not guaranteed that we shall use 100ms window size for application. We can change window size which we feel suit best for the application.

5.2.1.2 Fast Fourier Transform Analysis

Fast Fourier Transform is a very good way to analyze spectrum of a signal. We can identify amplitude of sound (db) over frequencies. Below you can see pictures of FFT of sound files which we collected for machine ***Delta HG-660 X 2200*** (see section 4.1). First we shall analyze Bronze material. You can see analyzed values of sound amplitude (db) over frequency below. We have taken third sound file for all states of cutting tools (see section 4.1.4 for states of cutting tools). See below table 1

State of cutting Tool	*Frequency (Hz)*	*Amplitude (db)*	*Frequency (Hz)*	*Amplitude (db)*	*State of cutting Tool*
New	5811	-44.7	13370	-56.2	New
Little Used	5931	-56.2	13361	-63.1	Little Used
Bad	6170	-55.9	13349	-59.4	Bad
Worn out	6122	-55.0	13394	-67.0	Worn out

Table No. 1

The above table shows us analyzed values for spectrum as shown below. We can see that when state of cutting is ***New*** then on frequency ***5811 Hz*** the amplitude of sound is ***-44.7***.

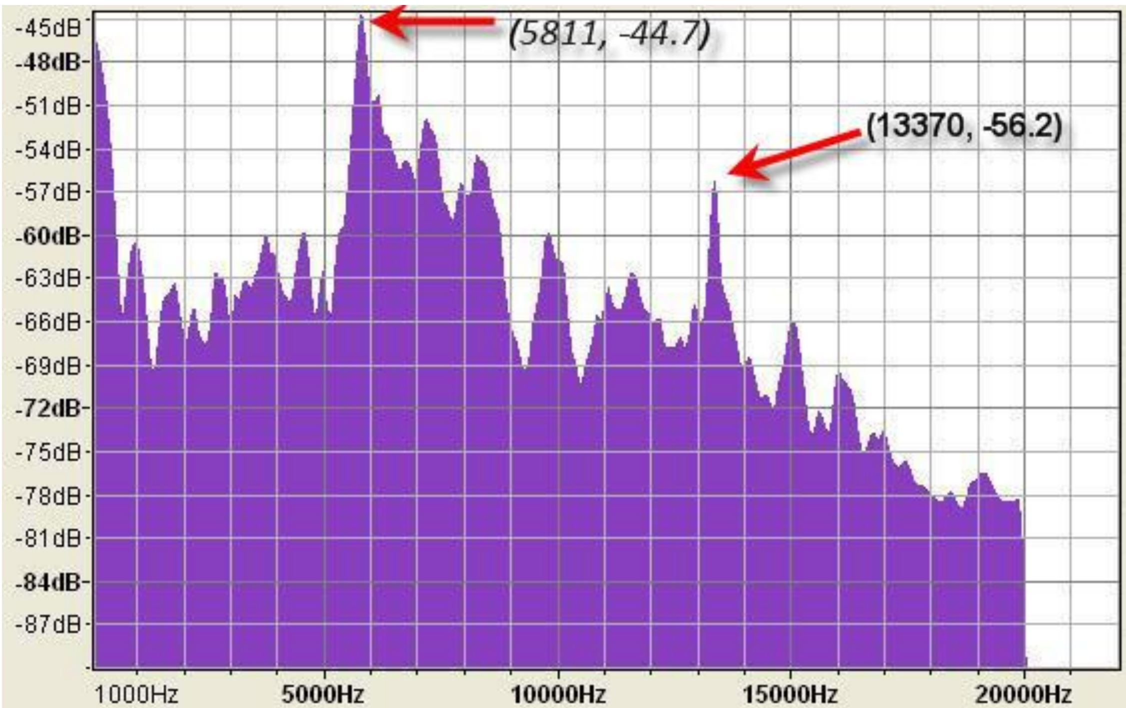


Figure 13 (a) FFT New Bronze

Now as cutting tool is used a little then we can see sound amplitude has decreased. We can see on when state of cutting tool is ***Little Used*** then on frequency ***5931 Hz*** the amplitude of sound is ***-56.2***.

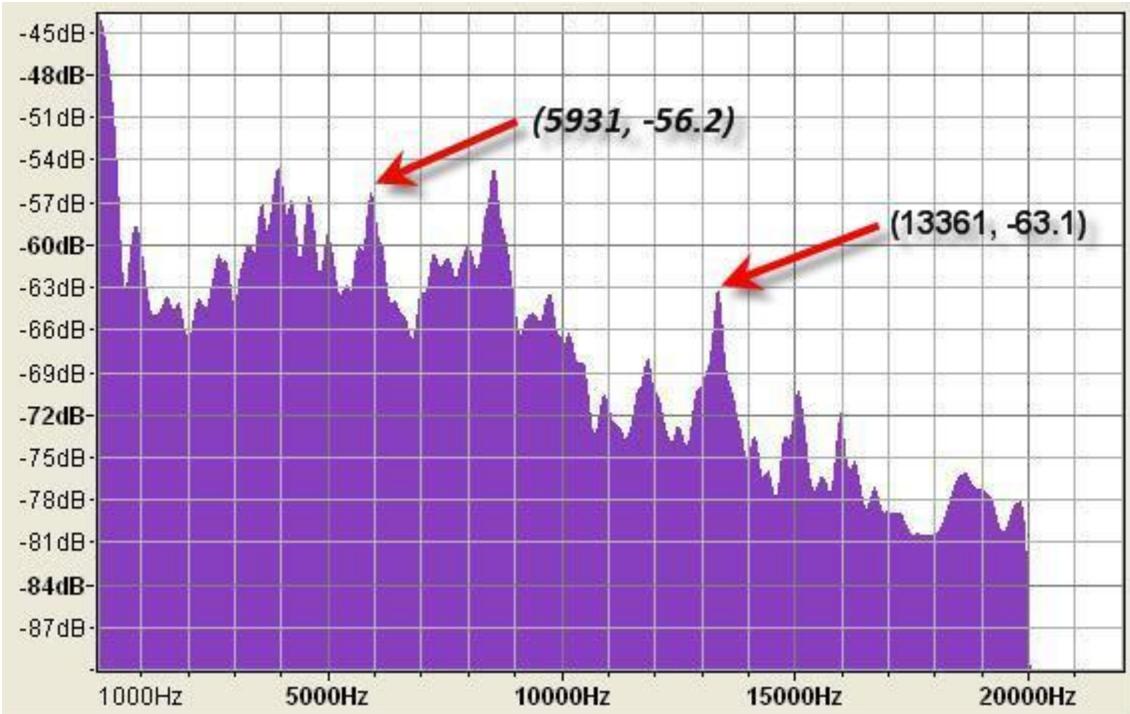


Figure 13 (b) FFT Little Used Bronze

Now, now we cut from a Bad state cutting tool. We can see a small variation in amplitude of sound. We can see on when state of cutting tool is ***bad*** then on frequency ***6170 Hz*** the amplitude of sound is ***-55.9***.

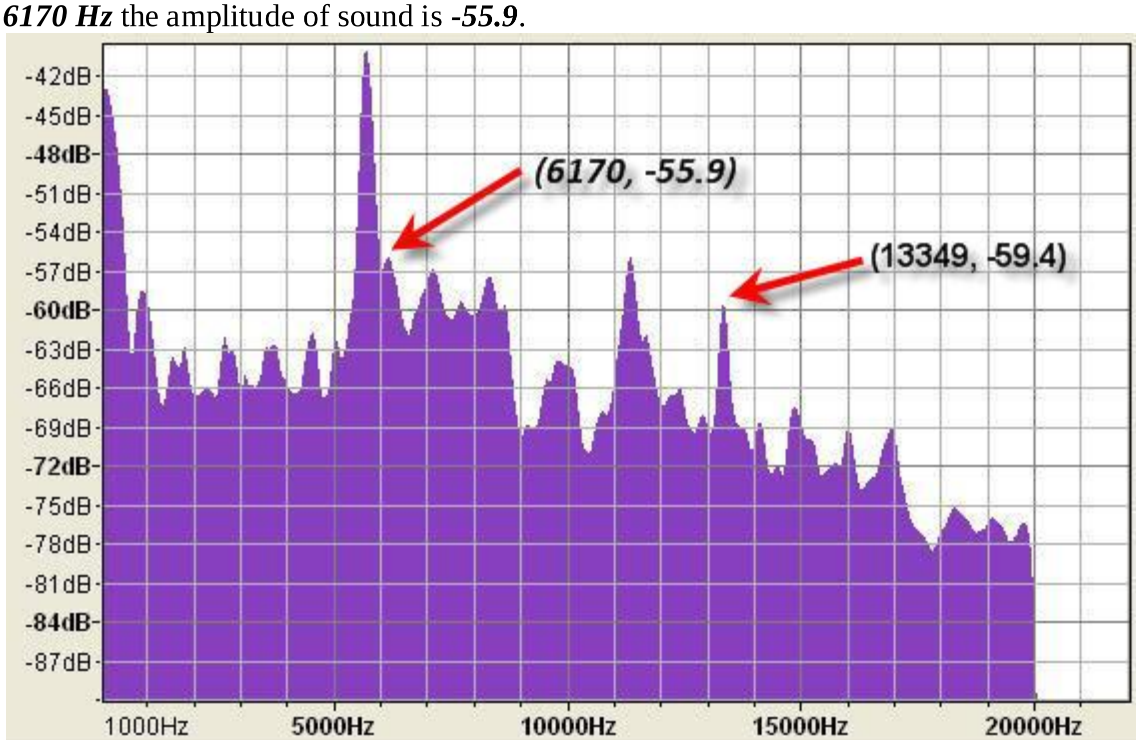


Figure 13 (c) FFT Bad Bronze

Now last but not the least and most important, we cut material from a cutting tool which has worn out state. Again we can analyze a small deviation in amplitude of a sound on certain frequency. We can see on when state of cutting tool is ***worn out*** then on frequency ***6122 Hz*** the amplitude of sound is ***-55.0***.

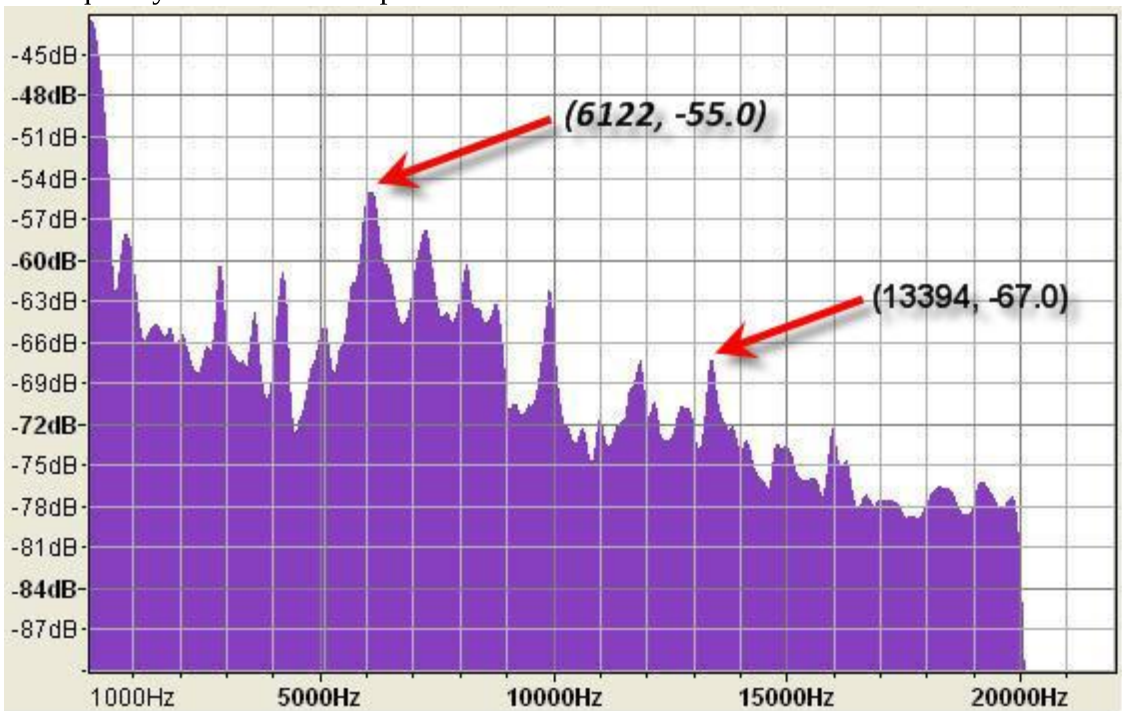


Figure 13 (d) FFT Worn out Bronze

We can see on frequency range between 5811 to 6150 Hz, we can see clearly there are some patterns. Like on frequencies, amplitude of sound is decreasing from top to bottom. An important thing, we can see which will be very helpful when we develop application, the difference between high and low amplitude value is ***-10.3 db***. The more difference it will be better for developing application. Next thing which is better after analyzing is, we have a clear pattern on amplitude of sound. It is also very good for developing application.

Now, we should also discuss other frequencies also. As you can see above ***Table 1***, I have also analyzed frequencies ranges between ***13370 to 13394 Hz***. We can see that when state of cutting is ***New*** then on frequency ***13370 Hz*** the amplitude of sound is ***-56.2***. Now as cutting tool is used a little then we can see sound amplitude has decreased. We can see on when state of cutting tool is ***Little Used*** then on frequency ***13361 Hz*** the amplitude of sound is ***-63.1***. Now, now we cut from a Bad state cutting tool. We can see a small variation in amplitude of sound and this variation is little odd. We can see an increase in sound amplitude which is not good for us. We can see on when state of cutting tool is ***bad*** then on frequency ***13349 Hz*** the amplitude of sound is ***-59.4***. Now last but not the least and most important, we cut material from a cutting tool which has worn out state. Again we can analyze a small deviation in amplitude of a sound which is again little odd as it is decreased. We can see on when state of cutting tool is ***worn out*** then on frequency ***13394 Hz*** the amplitude of sound is ***-67.0***.

Now, first thing is to consider difference between high and low sound amplitude. The difference is ***-10.8***. It is good for us for making application. But, the really worry for us is deviation of sound amplitude when state of cutting tool is ***bad***. That is the reason, we shall not consider this frequency ranges.

Now, we analyzed steel materials. See table 2 below

State of cutting Tool	Frequency (Hz)	Amplitude (db)
New	3245	-69.9
Little used	3315	-71.8
Bad	3273	-58.7
Worn Out	3275	-55.1

Table 2

The above table shows us analyzed values for spectrum as shown below. We can see that when state of cutting is ***New*** then on frequency ***3245 Hz*** the amplitude of sound is ***-69.9***.

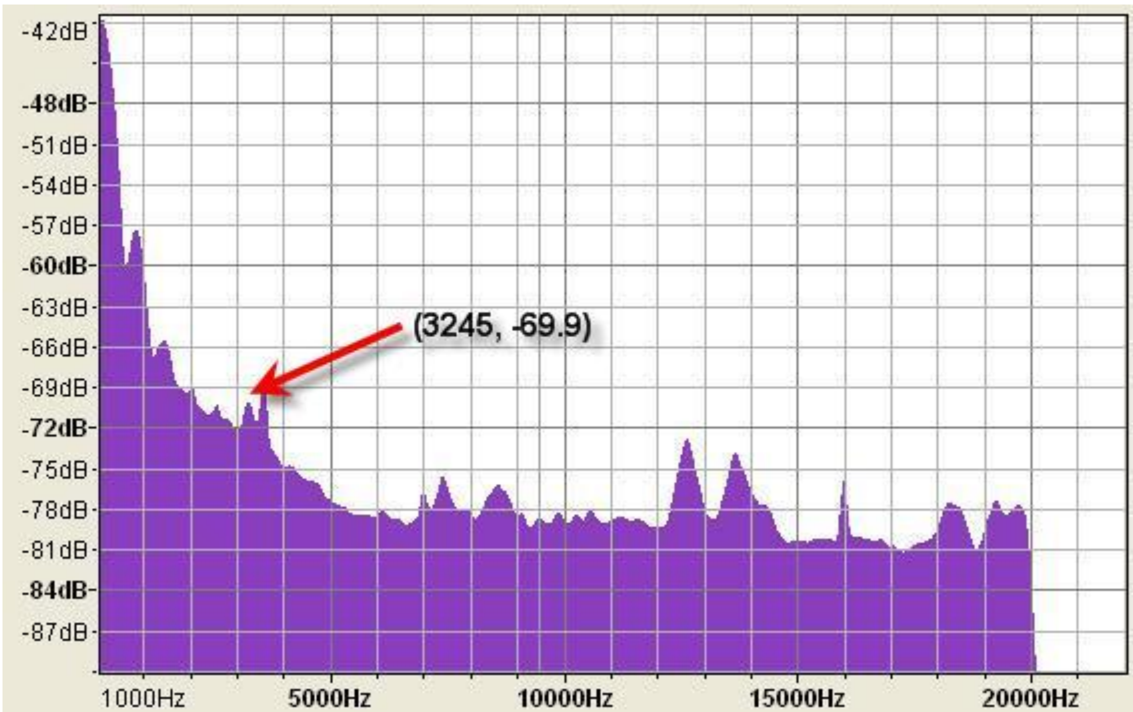


Figure 14 (a) FFT New Steel

Now as cutting tool is used a little then we can see sound amplitude has decreased. We can see on when state of cutting tool is ***Little Used*** then on frequency ***3315 Hz*** the amplitude of sound is ***-71.8***.

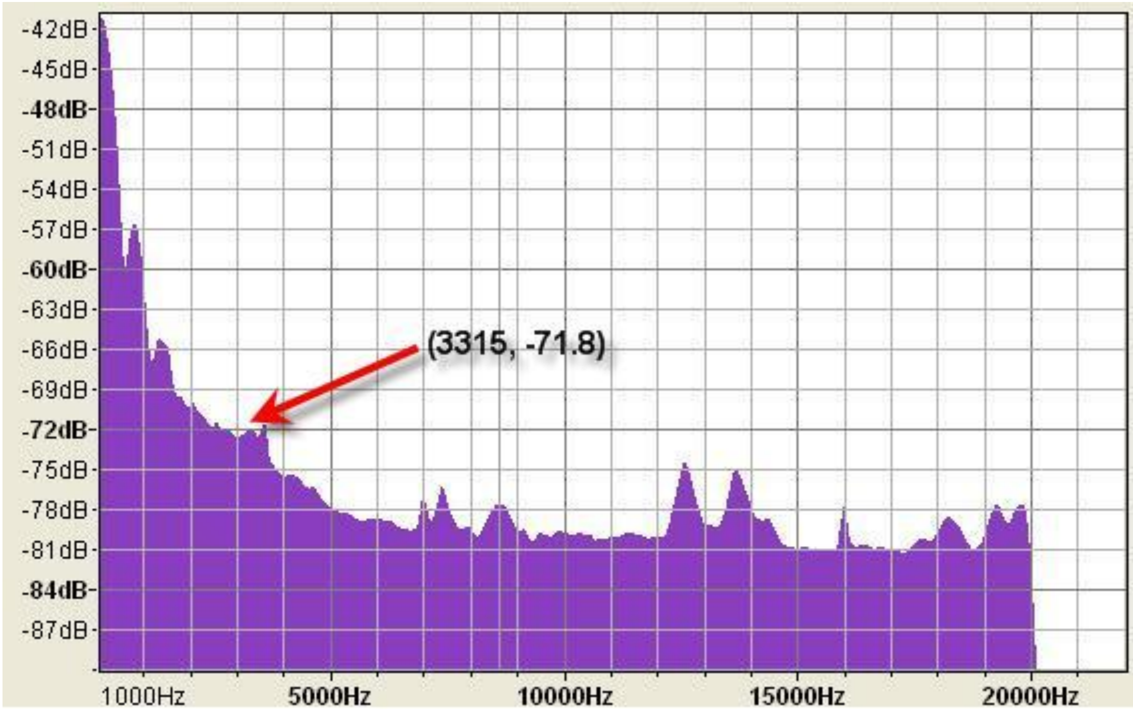


Figure 14 (b) FFT little used Steel

Now, now we cut from a Bad state cutting tool. We can see a small variation in amplitude of sound and it is little odd. We can see amplitude of sound shoot up rapidly. But it can happen with steel material. We can see on when state of cutting tool is ***bad*** then on frequency ***3273 Hz*** the amplitude of sound is ***-58.7***.

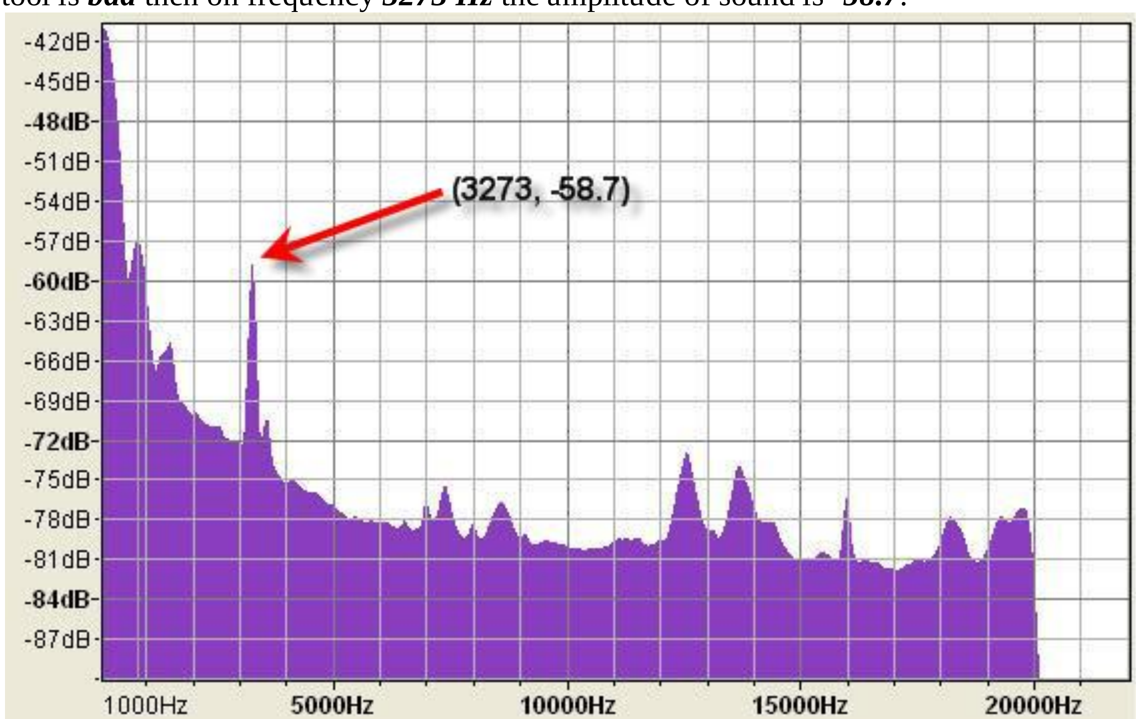


Figure 14 (c) FFT Bad Steel

Now last but not the least and most important, we cut material from a cutting tool which has worn out state. We can see on when state of cutting tool is ***worn out*** then on frequency ***3275 Hz*** the amplitude of sound is ***-55.1***.

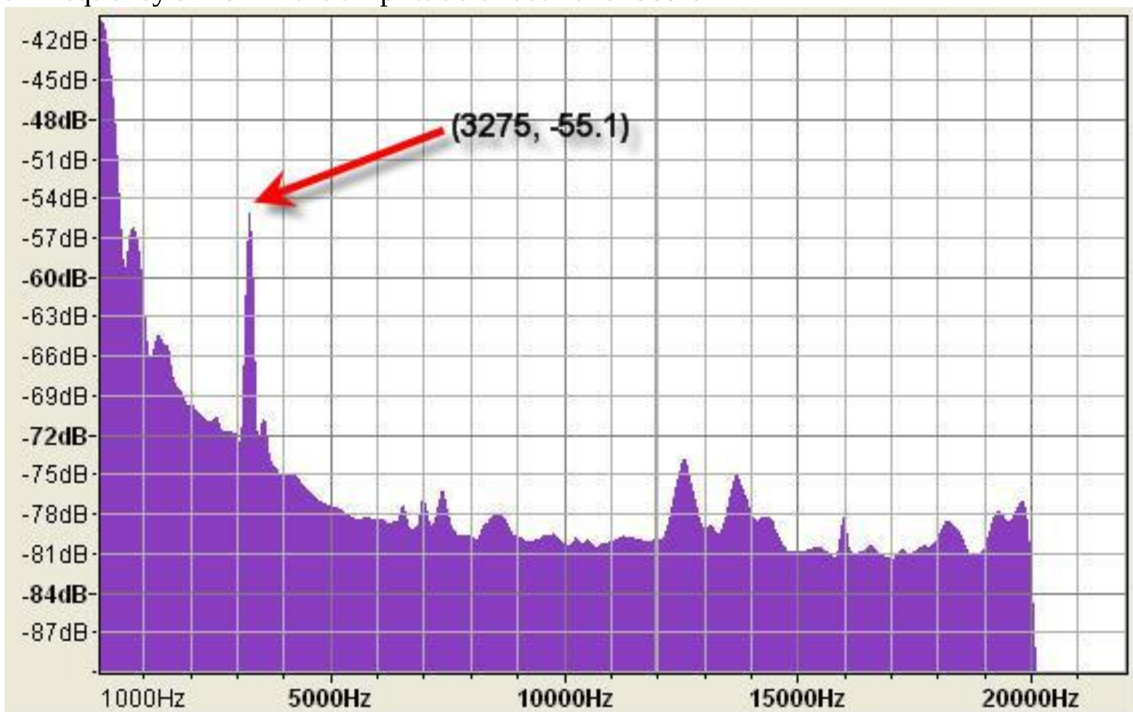


Figure 14 (d) FFT Worn out Steel

We can see on frequency range between 3245 to 3275 Hz, we can see clearly there are some patterns. But there is only one variation on frequency 3315 Hz. This result is a bit odd but if we see all other patterns which are really matter are absolutely fine. Like on frequencies, amplitude of sound is decreasing from top to bottom. An important thing, we can see which will be very helpful when we develop application, the difference between high and low amplitude value is ***-14.8 db***. The more difference it will be better for developing application. Next thing which is better after analyzing is, we have a clear pattern on amplitude of sound. It is also very good for developing application.

We cannot see other patterns in steel material while analyzing. It is really best results which we have figured out sound recordings.

5.2.1.3 Background Noise Analysis

While recording sound, it is impossible we do not face any interruption from outside environment. It is certainly the case with us. We faced many such of background interferences like human voice, machine sound etc. You can see human sound interruption in ***section 5.2***. Here we shall discuss only background sound which we faced from machine.

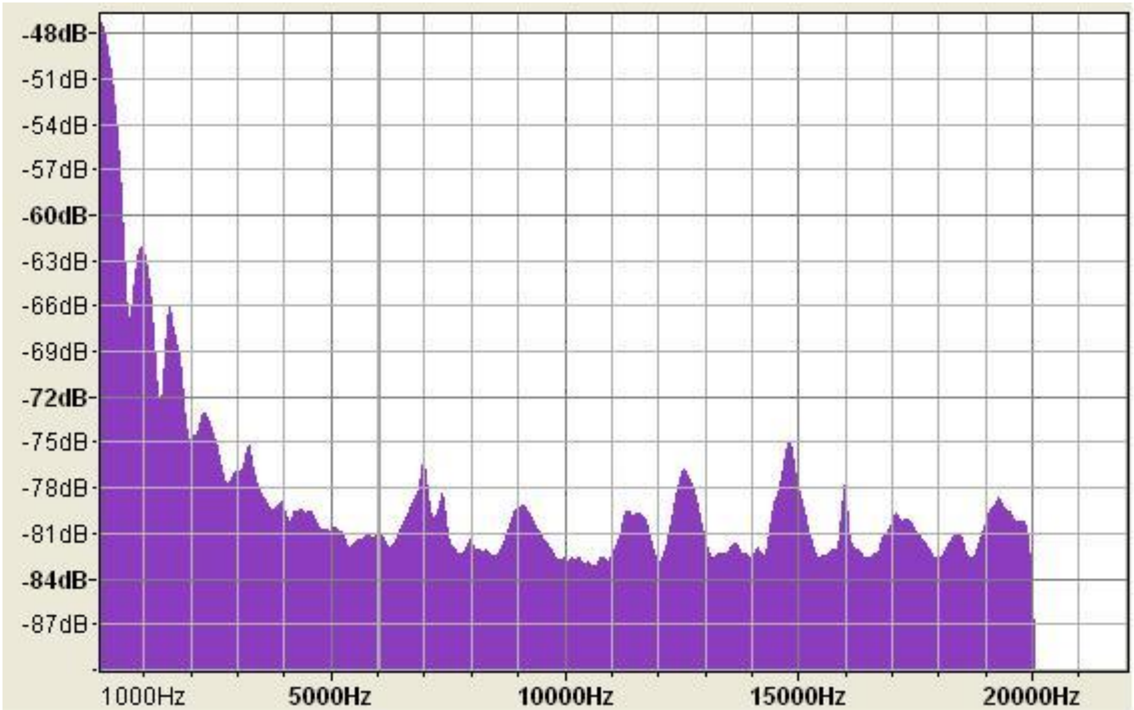


Figure 15 FFT Background

Here we can see clearly sound of running machine has affect cutting process with some interference. So, if we want to get pure cutting sound then we have to deduct this sound amplitude with our original sound file.

5.3 Experiment two

In this experiment, we shall analyze window size and FFT for cutting process for propeller. The machine used in this cutting process is named ***Lathe Machine***, manufactured by ***Torni Tacchi from Italy***. We used Steel made propeller having length of ***3314 mm*** and diameter of ***220-225 mm.***

Figure 16 Lathe Machine

The purpose for this recording, we want to check what happen if a cutting tool cut steel material from new state till worn out. So, we used only one cutting tool for whole process. We started cutting surface when tool is in new state and cut till tool became worn out. While cutting tool cut the surface of material, revolution of machine (rpm) may affect states of cutting tool. If you increase rpm of machine then tool will wear out quickly. We set rpm to ***162.*** We have also set machine cutting depth to ***4mm***.

Also, it is rather large recordings, almost thirty minutes. Audacity is used to record entire process. The location of microphone is very important also. The height from cutting tool and where cutting will start is ***24cm*** width is ***22cm*** and diagonally is ***34cm***.

5.3.1 Investigation of window size

We will check the window size of any sound file. As we have shown in section 4.2. we shall check window size of 500ms, 200ms and 100ms and decide which is appropriate for this process.

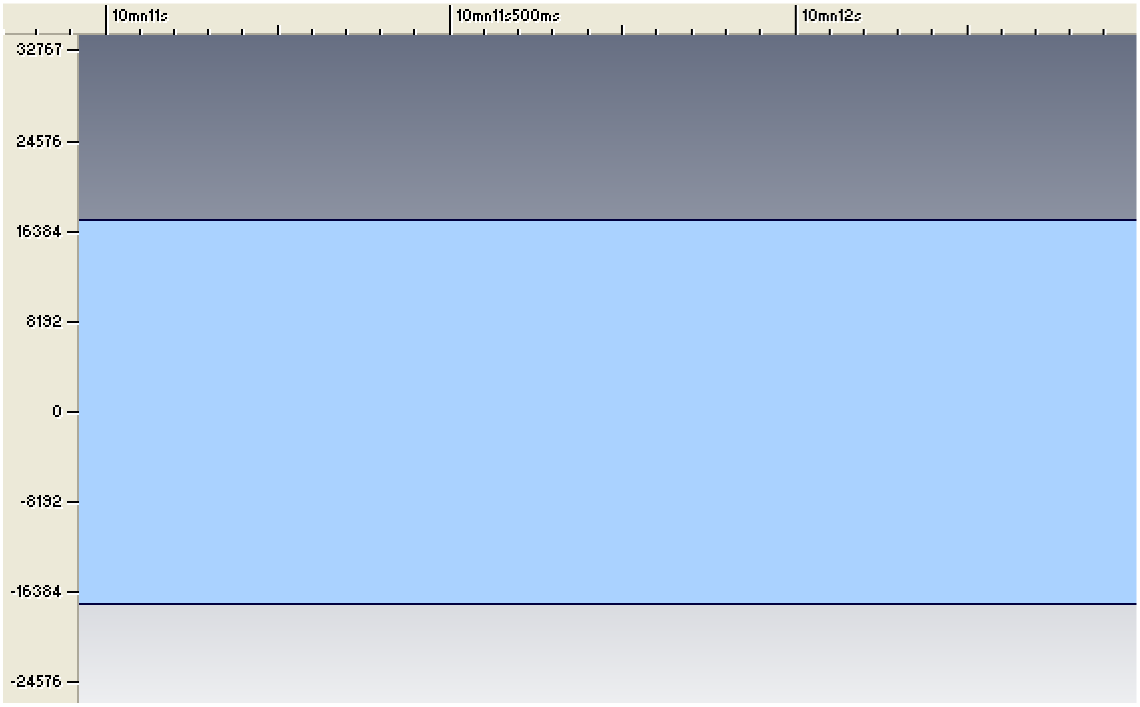


Figure 17 (a) 500ms

Figure 17 (b) 200ms

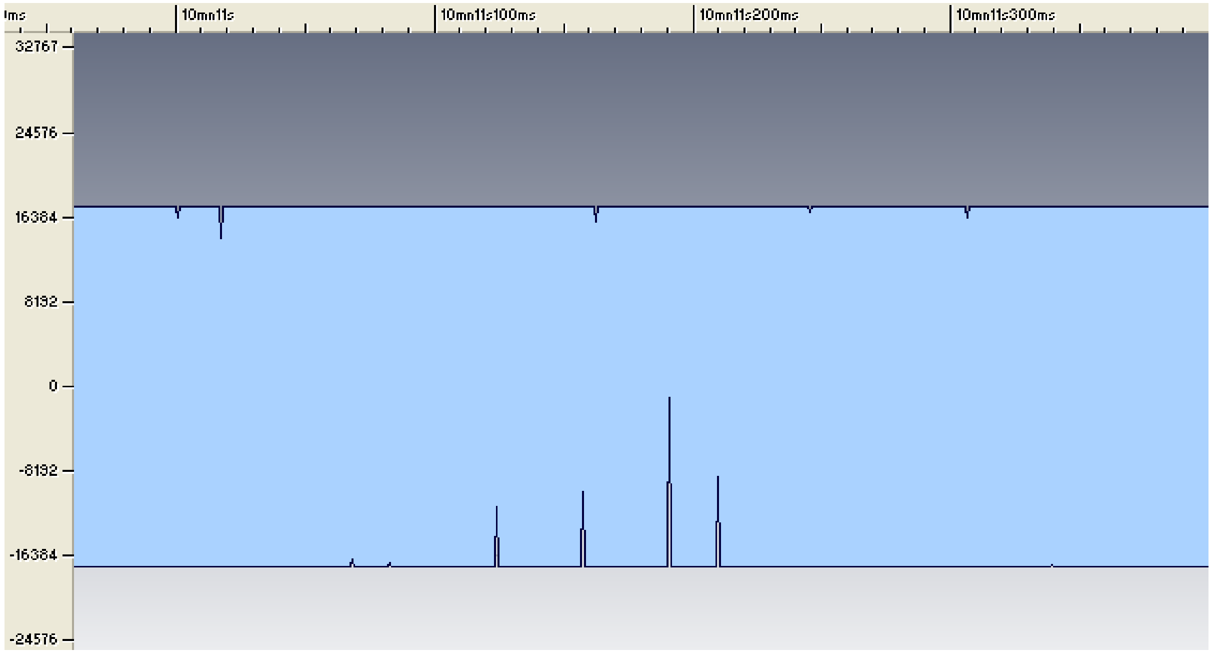


Figure 17 (c) 100ms

Clearly, in all three figures we cannot see any wave patterns. It may be possible that, wav pattern are so well intact so we can see them in large window size. For seeing some patterns we have to zoom in.

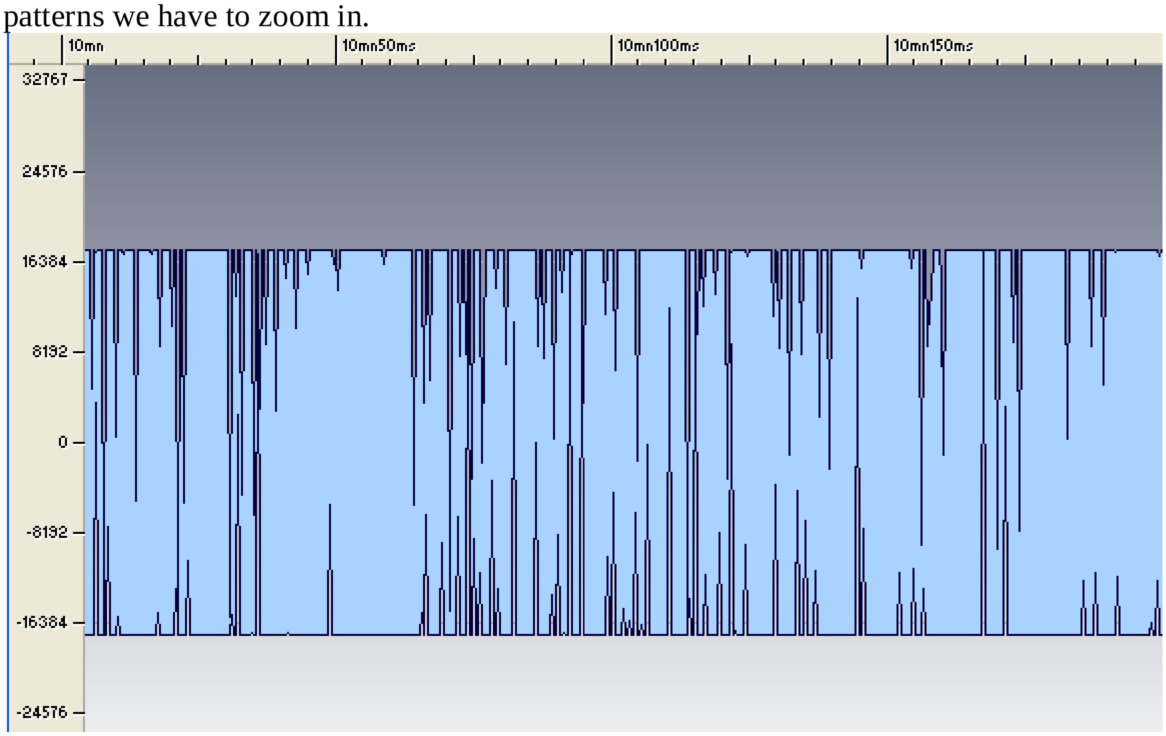


Figure 17 (d) 50ms

In the above picture, we can see some sort of pattern but still it is not clearly visible. So, we have to zoom in more.

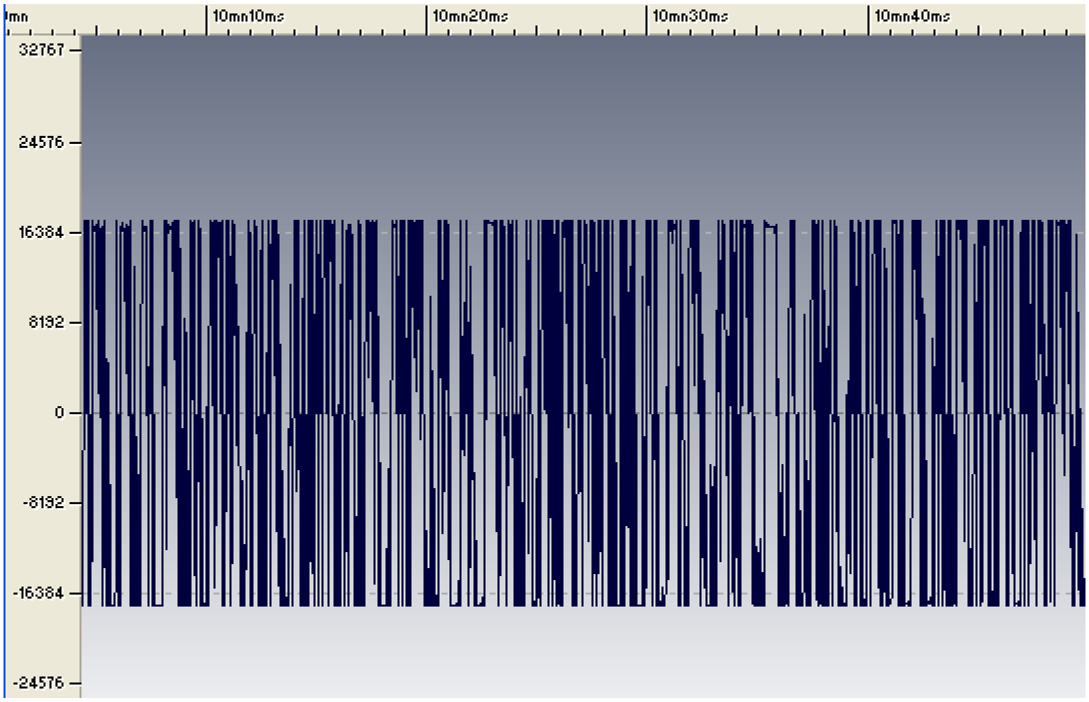


Figure 17 (e) 10ms

Now, here we can clearly see the patterns of waves. And we can see the amplitude of sound is deadly accurate. And it remains accurate in all sound files. It shows us, while cutting process is going on, the amplitude of sound remain almost constant.

5.3.2 Fast Fourier transform analysis

The following table shows us the analyzed values of recording. For easiness, we have divided our recording in small parts. We split it with some split wav tool which can easily be downloaded. Every sound file two minutes and forty seconds long.

Frequency (Hz)	***Amplitude (db)***
5212	-20.5
5220	-19.2
5222	-19.2
5222	-18.2
5211	-22.0
5192	-18.8
5209	-14.8
5187	-13.0
5178	-14.5
5161	-14.1

Table 3

We can see that when state of cutting is ***New*** then on frequency ***5212 Hz*** the amplitude of sound is ***-20.5***.

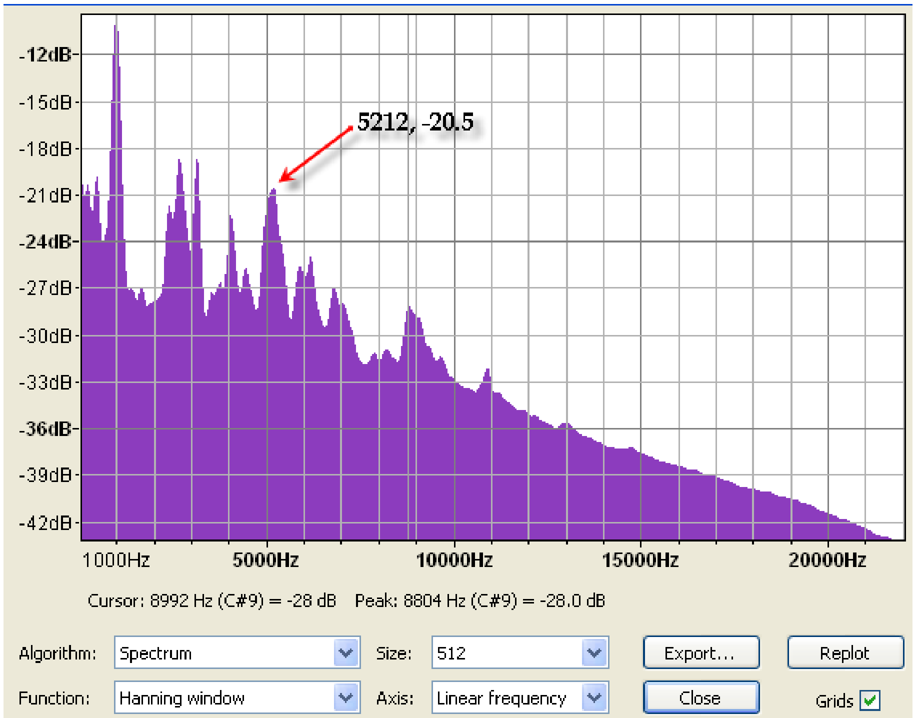


Figure 18 (a) FFT Correct Cut

Now when cutting tool is cutting surface of the steel propeller, it is little used now. So now, we can see little deviation in amplitude of sound. Now, amplitude of sound has increased little. Now, when tool is little used (relatively new) then on frequency ***5220 Hz*** the amplitude of sound is ***-19.2***.

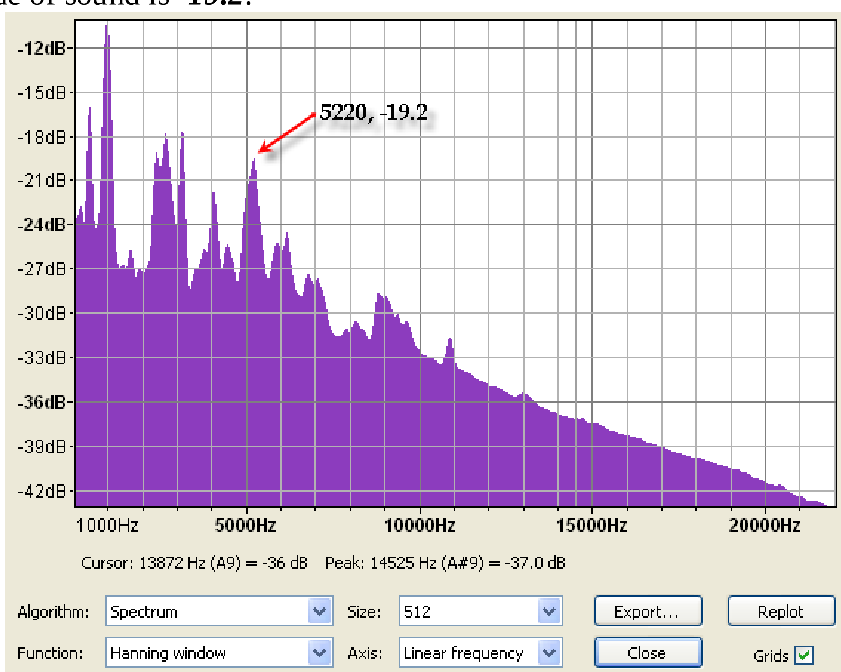


Figure 18 (b) FFT Correct Cut

Now, cutting is more used but it is still in little used state. Now, on frequency ***5222 Hz*** the amplitude of sound remain same (***-19.2***).

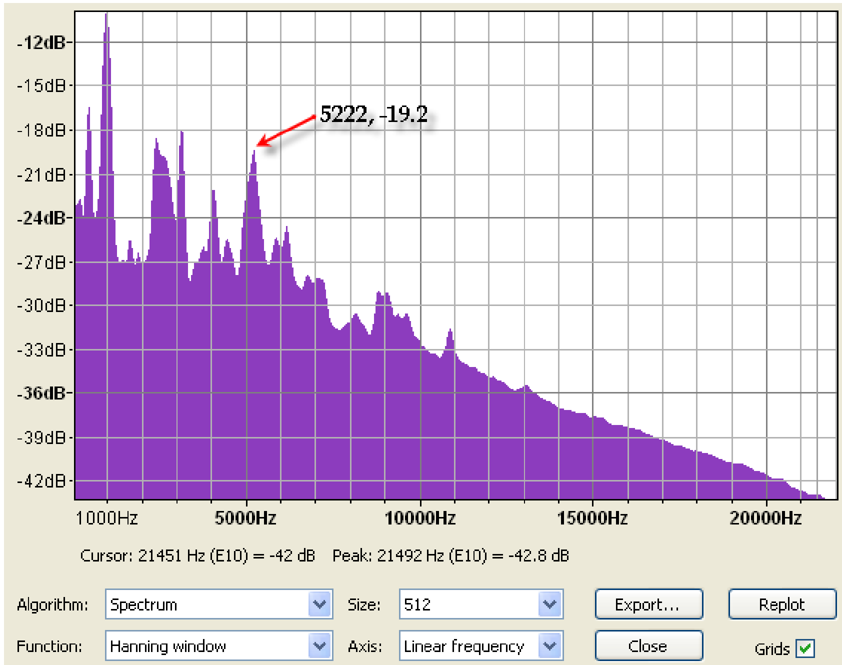


Figure 18 (c) FFT Correct Cut

Cutting tool is cutting surface continuously. So now, we can see little deviation in amplitude of sound. Now, amplitude of sound has increased little. Now, when tool is little used then on frequency ***5222 Hz*** the amplitude of sound is ***-18.2***.

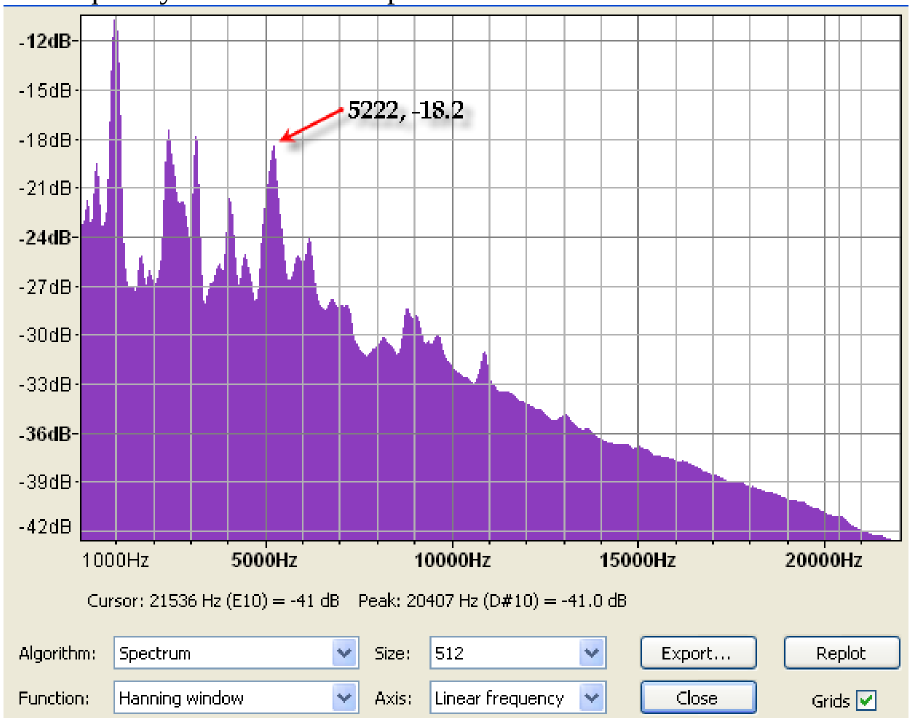


Figure 18 (d) FFT Correct Cut

Now, we cut from a Bad state cutting tool. We can see a small variation in amplitude of sound and it is little odd. We can see amplitude of sound shoot up little. But it can happen with steel material as tool is about to go in more used or bad state. We can see on frequency ***5211 Hz*** the amplitude of sound is ***-22.0***.

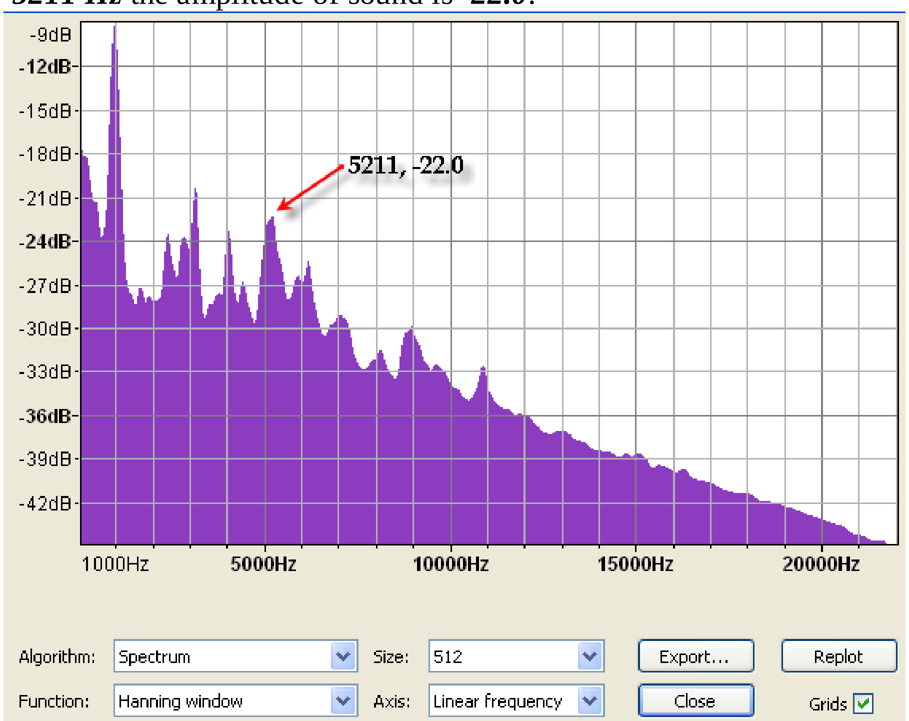


Figure 18 (e) FFT Correct Cut

We can see on frequency ***5192 Hz*** the amplitude of sound is ***-18.8***. It is rather fine.

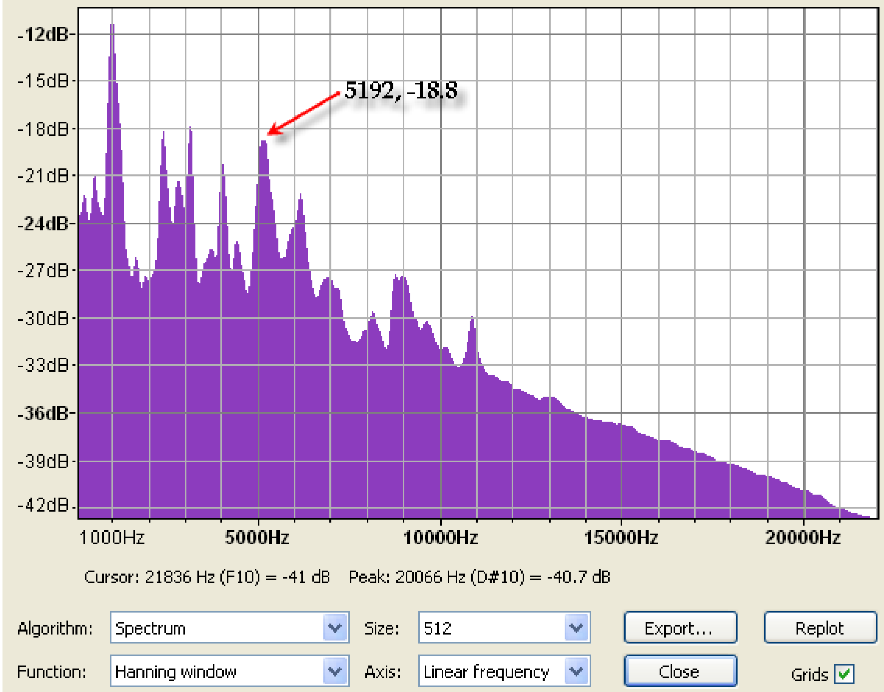


Figure 18 (f) FFT Correct Cut

Now, we cut from a Bad state cutting tool. We can see a small variation in amplitude of sound and it is little odd. We can see amplitude of sound shoot down little. We can see on frequency ***5209 Hz*** the amplitude of sound is ***-14.8***.

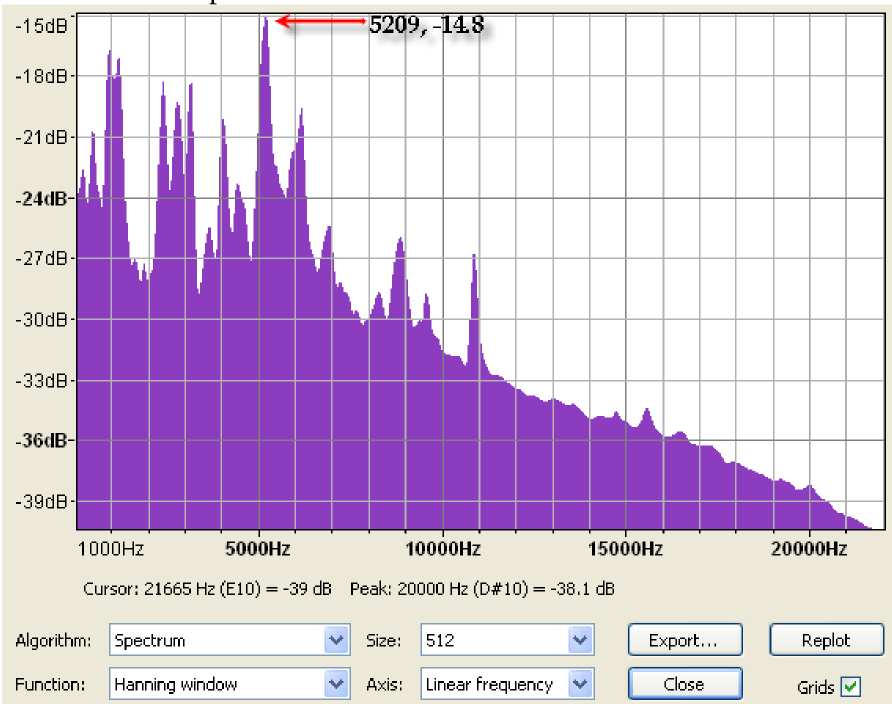


Figure 18 (g) FFT Correct Cut

Now when cutting tool is cutting surface of the steel propeller, it is more used now and it is in bad state now. So now, we can see little deviation in amplitude of sound. Now, amplitude of sound has increased little. Now, when tool is more used (bad state) then on frequency ***5187 Hz*** the amplitude of sound is ***-13.0.***

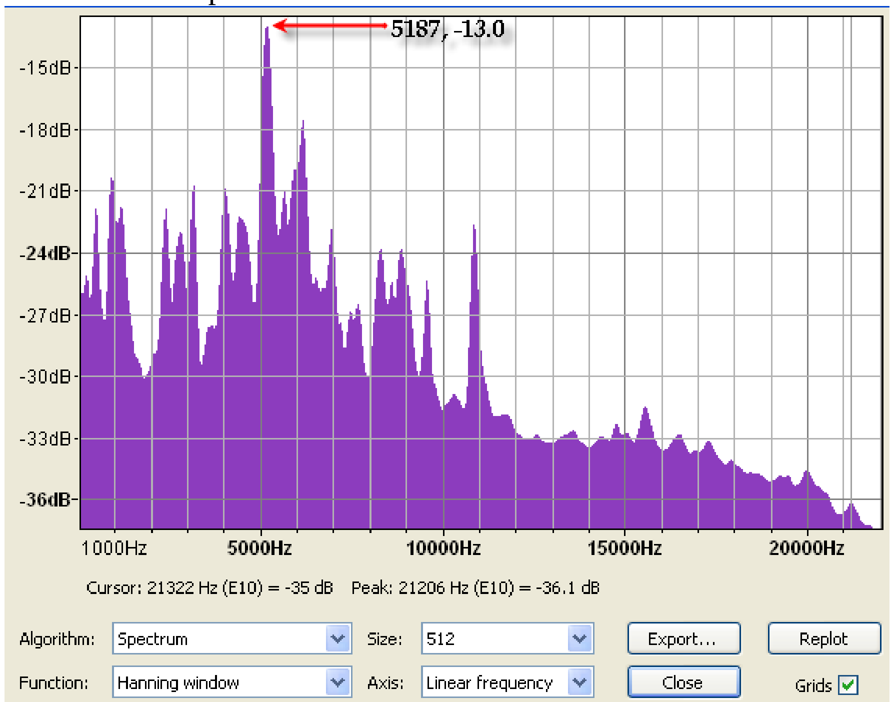


Figure 18 (h) FFT Correct Cut

Now, cutting tool is certainly in bad state or worn out. So on frequency ***5178 Hz*** the amplitude of sound is ***-14.5.***

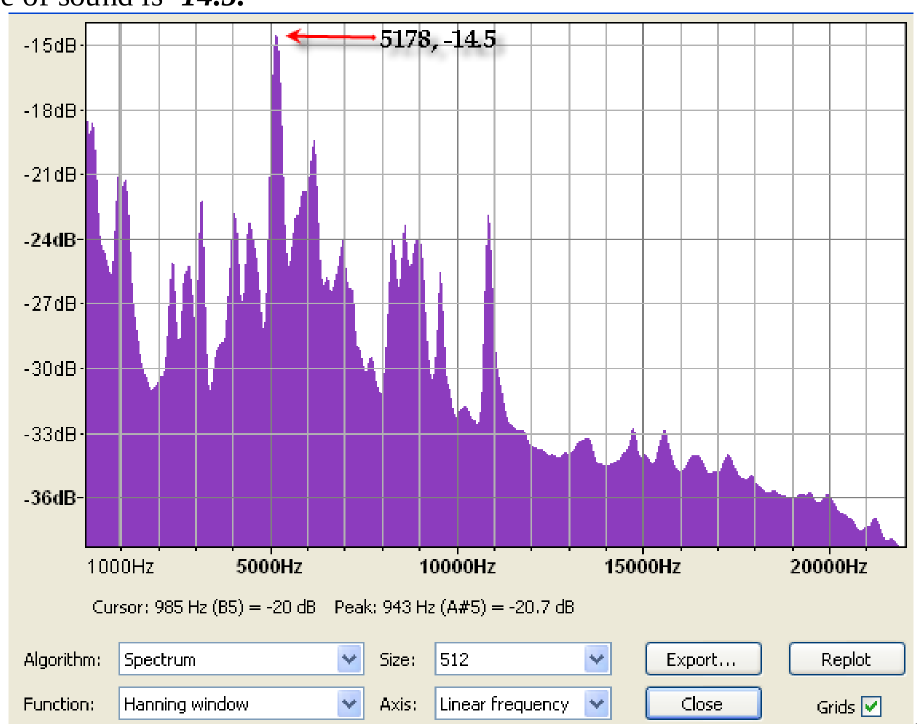


Figure 18 (i) FFT Correct Cut

Now, cutting tool is certainly worn out. So on frequency ***5161 Hz*** the amplitude of sound is ***-14.1.***

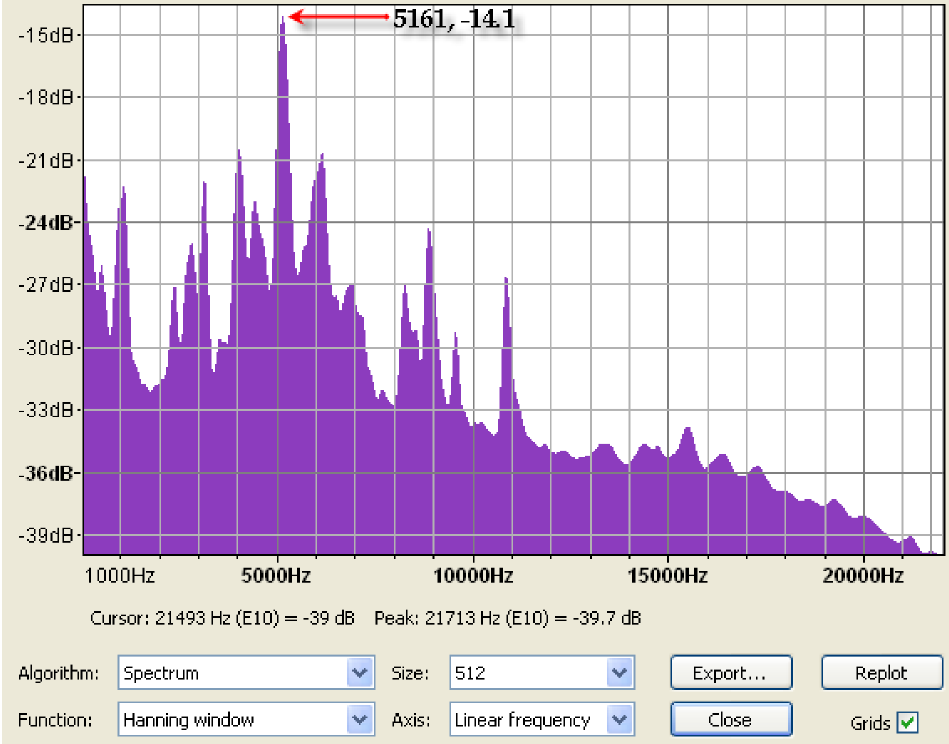


Figure 18 (j) FFT Correct Cut

We can see on frequency range between 5161 to 5222 Hz, we can see clearly there are some patterns. But there is some variation on different frequencies 5211, 5187 Hz. This result is a bit odd but if we see all other patterns which are really matter are absolutely fine. Like on frequencies, amplitude of sound is decreasing from top to bottom. An important thing, we can see which will be very helpful when we develop application, the difference between high and low amplitude value is ***-5.4 db***. The more difference it will be better for developing application. But in this case it is absolutely fine. Because recording sound size is about thirty minutes. So it is large enough duration to identify either cutting tool is worn out or not. Also Next thing which is better after analyzing is, we have a clear pattern on amplitude of sound. It is also very good for developing application.

5.3.3 Background sound analysis

Here we can see clearly sound of running machine has affect cutting process with some interference. So, if we want to get pure cutting sound then we have to deduct this sound amplitude with our original sound file.

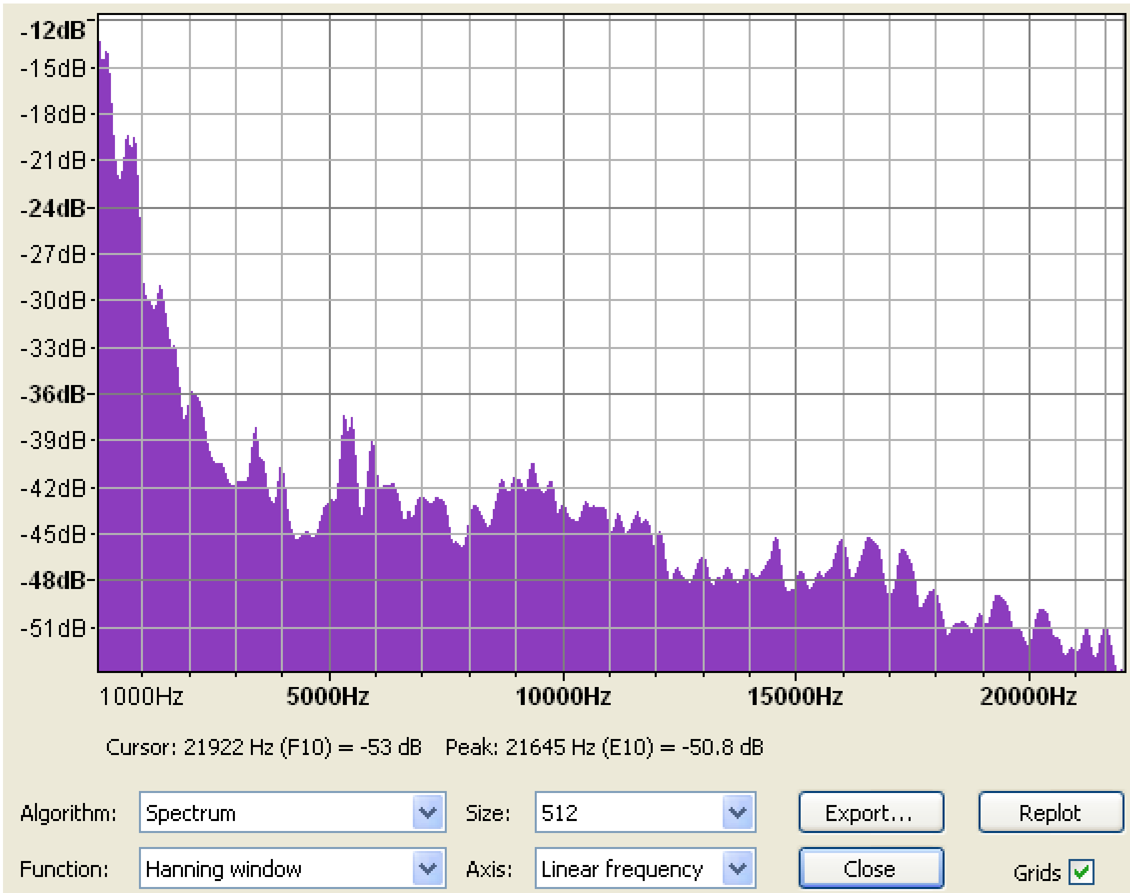


Figure 19 FFT Background Sound

6. Architecture

In development area we have adopted three layered architecture approach due to some advantages of this architecture. It supports modular approach. It is helpful in the condition when there are more than one persons working on the same project. Another important reason to use this architecture is that the bug fixing is relatively easy in this architecture and we have to focus on the respective layer only. These layers are Presentation Layer (PL) Business Logic Layer (BLL) and Data Access Layer (DAL).

6.1 Presentation Layer (PL)

This is the layer through which user interacts with the system. It contains all the graphical images and forms needed for the designing of the Development of a Prototype System for Milling Machines. Presentation layer of the editor is very important for the user's point of view therefore it should be user friendly so that the user can interact with the system easily.

6.2 Business Logic Layer (BLL)

This is the layer where actual code of the program will be written. Hence this is the most important layer. All the functional algorithms will be written in this layer. These algorithms will enable communication between the Presentation Layer and Data Access Layer.

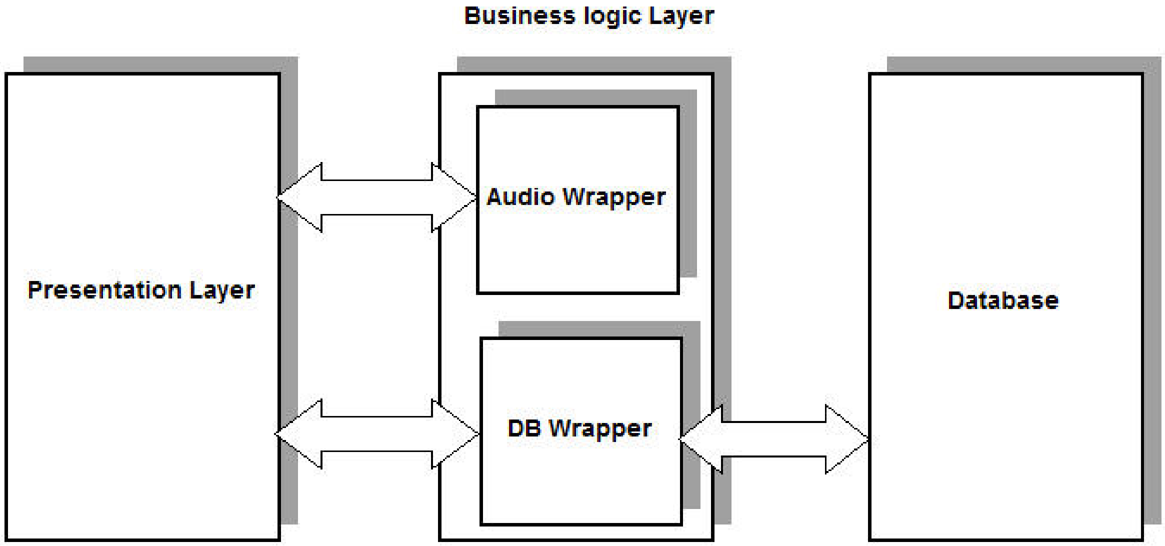
6.2.1 Audio Wrapper

This layer will provide access to microphone data. It is the most important part of the layer. In this layer, coding is done for input sound from microphone. Also, FFT algorithm is implemented in this layer.

6.2.2 DB Wrapper

This is the layer which will provide the access to the database of the system. All the functions related to reading, writing and modifying the data will be the part of this layer.

A figure of these layers interaction is shown below:



7. Implementation

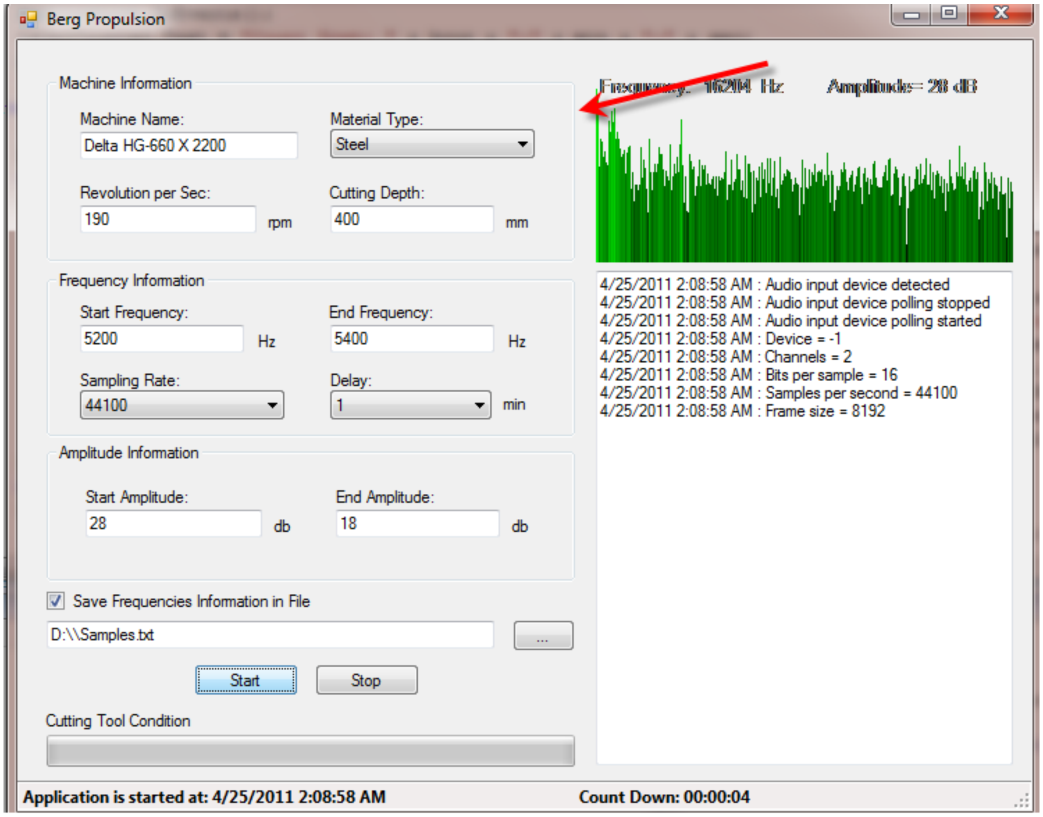
After analyzing all the recordings in both experiments section 4.2 and 4.3, a prototype system is developed.

Now, you'll see little description of prototype system

7.1 Presentation Layer

7.1.1 Machine Information

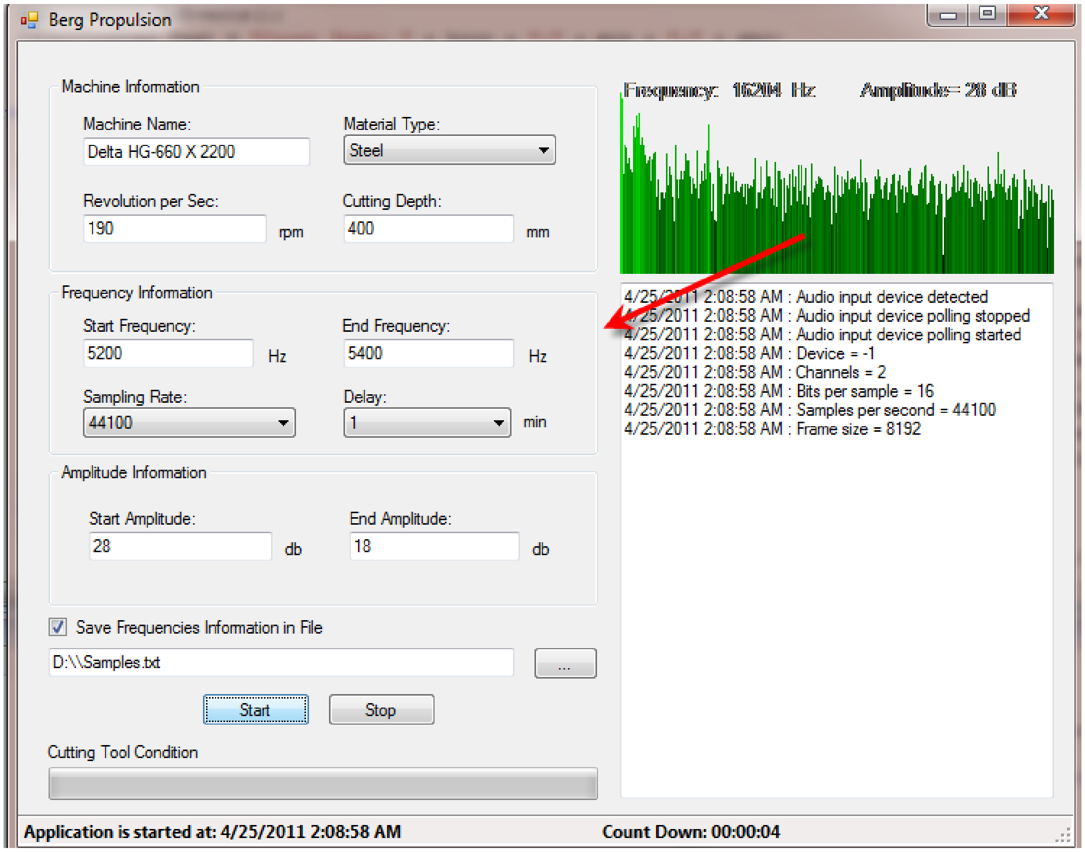
This group is for machine specific information. We have name of the machine, what type of material we are cutting, what revolution per second and what is the cutting depth which cutting tool is cutting.



Picture machine information

7.1.2 Frequency Information

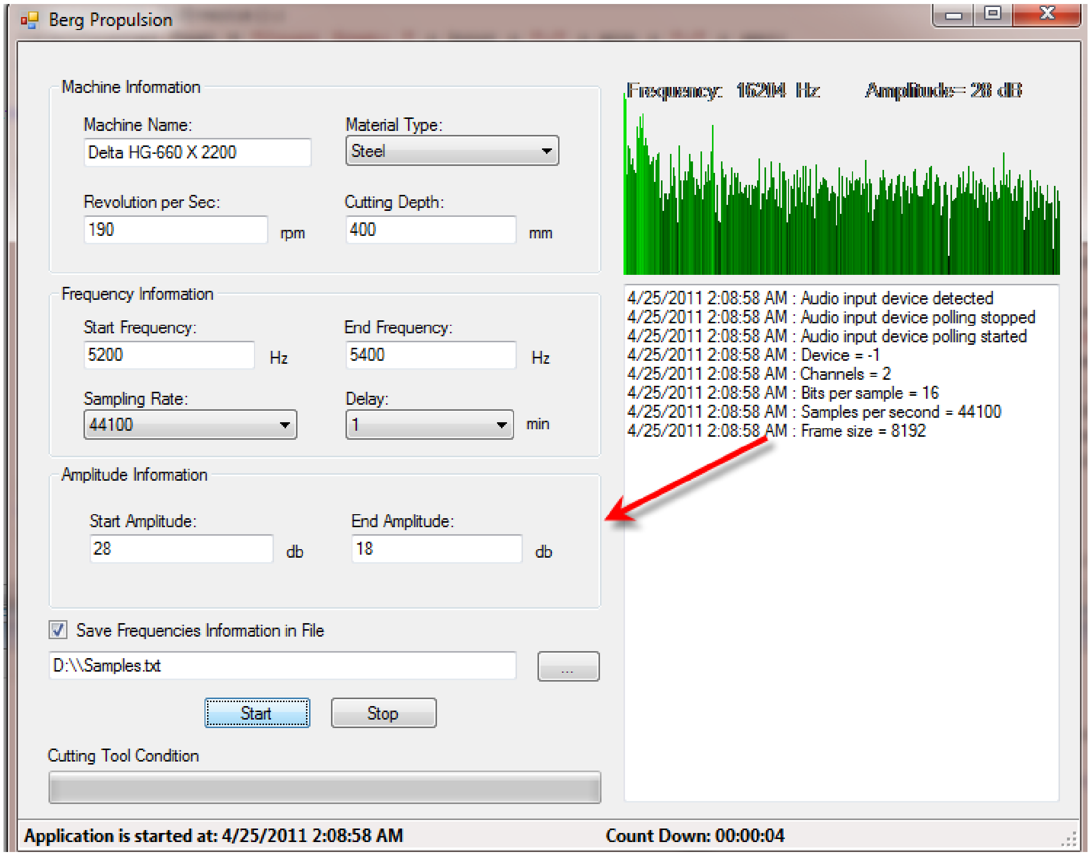
This group has frequency related information. We should enter start frequency and end frequency. Start frequency should always be greater to end frequency. We should set appropriate sample rate which is 44100 Hz in our case. We can set delay. This delay is used to refresh cutting conditioning tool. Like every one minute amplitude will be check and cutting tool progress bar will be updated.



Picture frequency information

7.1.3 Amplitude Information

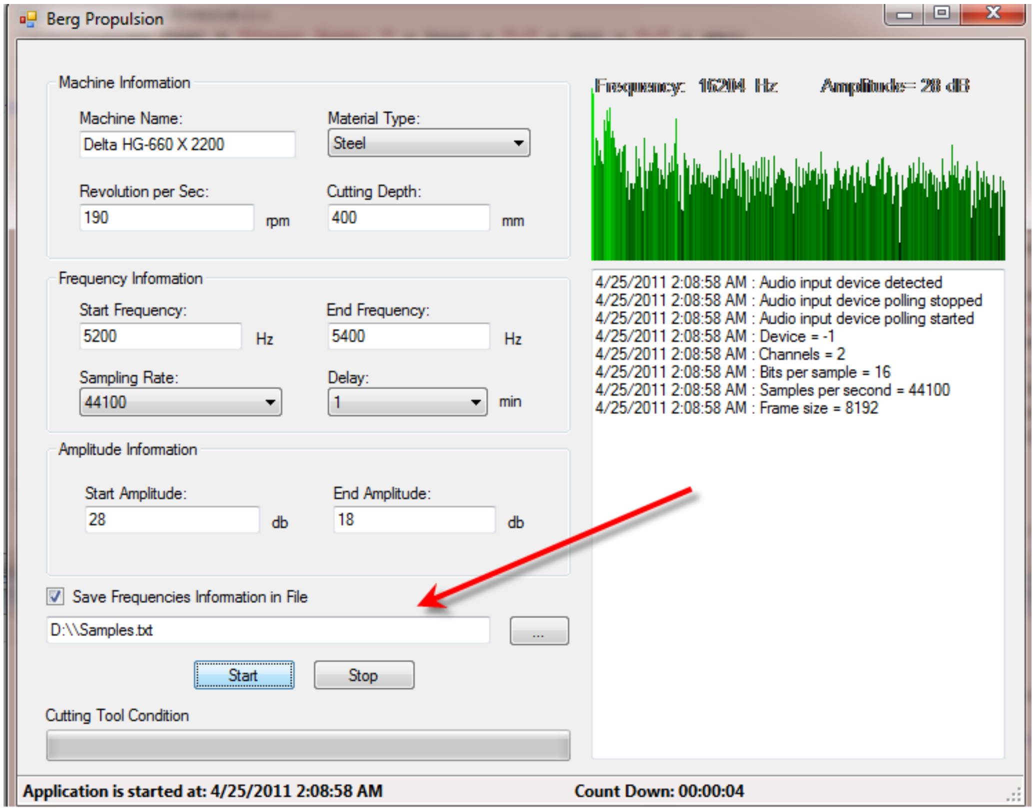
We should give appropriate amplitudes which we have analyzed at section xxxx. Start amplitude should be greater than end amplitude.



Picture amplitude information

7.1.4 Save into file

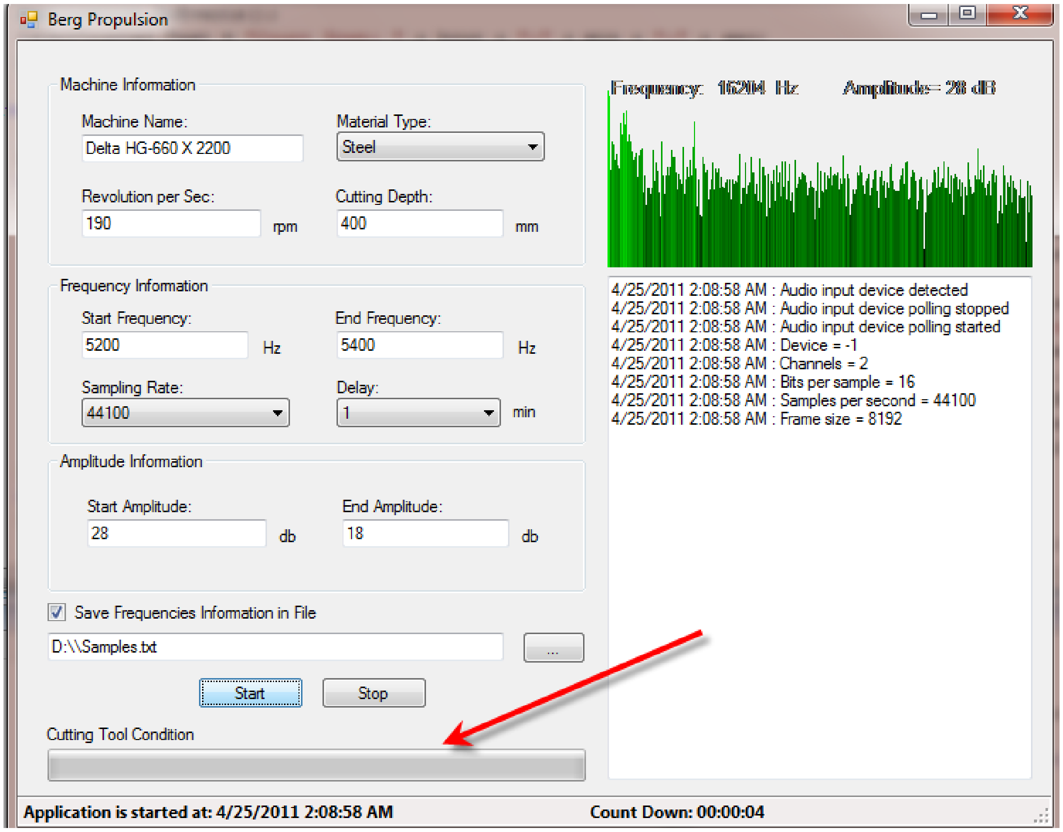
If save frequencies information in file check box is checked then all frequencies and amplitude information will be saved in file sample.txt in specific location. We can change location from a button named ... as shown above.



Picture save file

7.1.5 Cutting tool monitoring

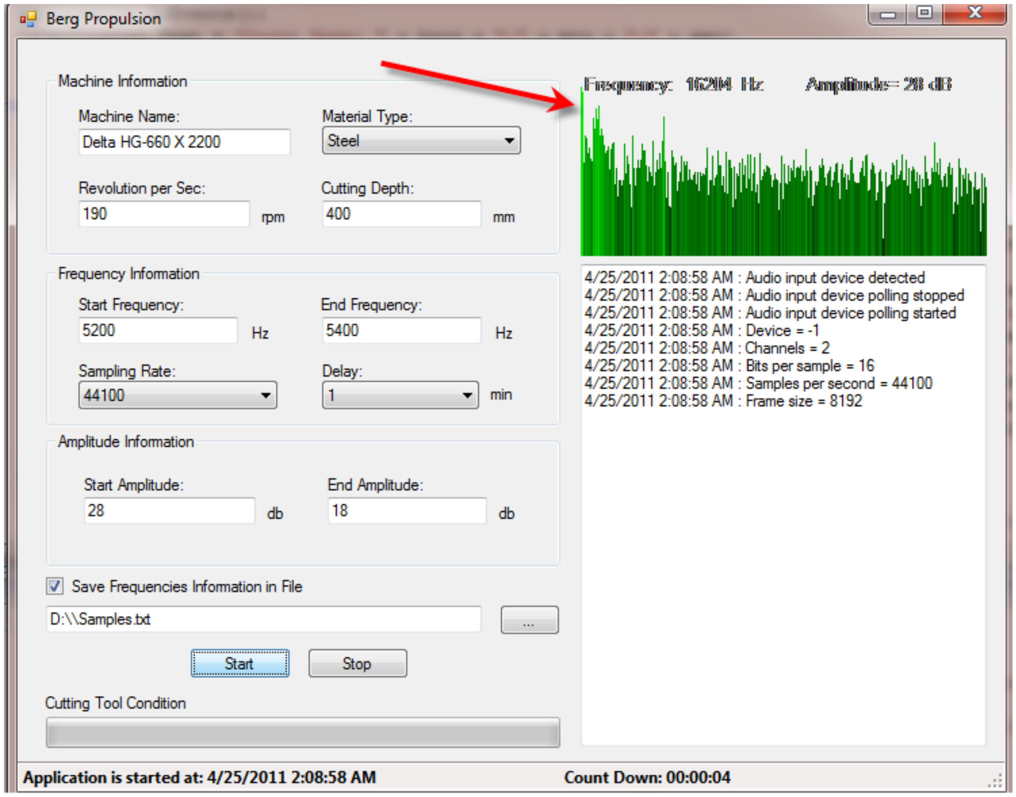
This is main core part of our thesis. This progress bar will show the condition of cutting tool. It depends on values of amplitudes which we set on section 4.3. if cutting tool is worn out then progress bar should be 80%. We set a threshold value because after 80% cutting is already in bad state, so it is better to notify little early.



Picture cutting tool condition

7.1.6 Spectrogram

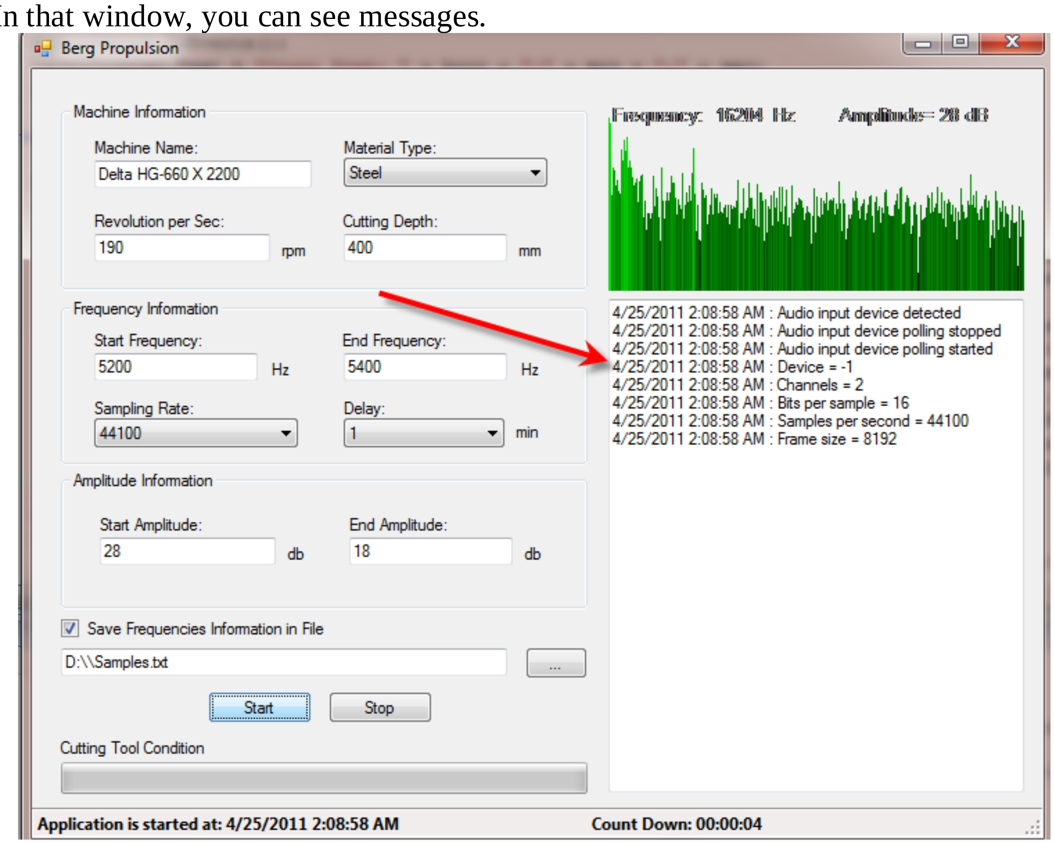
This graph will show the frequency and amplitude of the sound. This will show information directly from microphone.



Picture spectrogram

7.1.7 Console

In that window, you can see messages.



Picture console

7.2 Business Logic Layer

7.2.1 Audio Wrapper

In this layer, all classes are presented which can take input data from the microphone. It has also option for recording. A class named WaveIn.cs is used to get input from microphone. A class named “WaveOut.cs” is used to store sound. A class name “Fast Fourier.cs” is used to get samples in db. There are some other classes also, which is not important in project, rather used to display spectrogram on presentation layer.

7.2.2 DB Wrapper

In this layer, all database related quires are located.

8. Results

We have presented two parts of our thesis, theoretical and implementation part. For theoretical part, we have analyzed sound signals on software named Audacity. After analyzing that part, we are sure our analyzed result are pretty accurate. Because, in section 4.3 and 4.4, you can see a pretty clear pattern, which we really want to implement a prototype system.

For implementation part, we have tried you show a progress bar which shows the condition of cutting tool (see different states of cutting tool in section 4.2.4). The results got from FFT algorithm from implementation part are different from theoretical part. Sometime, our software shows rather correct results but more time results are totally different. In our point of view, there can be some reasons behind this.

- Interference of operating system sounds. There also can be noises from wire itself.
- There can be some difference in Audacity FFT algorithm than implementation FFT algorithm.
- Less testing on site.

9. Conclusions

9.1 Performed Work

We have performed very interesting and knowledgeable tasks in this thesis work.

First of all we have tried to give reader a simple an easy overview of different tasks involved in monitoring the condition of cutting tool. The main tasks are case-based reasoning, state of the art and different types of signal processing. The second and important task was to collect the sound signals and analysis them. We have to get valuable data from collected sounds and remove the noises from the sounds. The next step was to classify the signals into different categories depending on their similar properties. The next step was to research on the different software techniques to analyze the properties of the sound signals. The last task which we have performed is the implementation of a computer based automatic system to monitor the condition of the cutting tool.

9.2 Achievements

I think result which we get from our theoretical part are really interesting and worth watching. It is really very good building block to make a reliable system. Also to some extend our achievement in limited time for implementation is a prototype system that is able to monitor the working condition of the milling tool using acoustic emission signals. The system arise an alarm if the cutting tool is damaged and is in bad condition.

9.3 Critical Discussion / Lesson Learned

In the thesis, we can infer many positives in cutting monitoring condition. An approach which we used to find FFT is reliable approach for monitoring system. As, we have shown in section experiment2 and table 3, there is a case while recording when there is some rough cutting has done by the cutting tool. We cannot extract suitable information from our FFT. So, that part is not go in our way. But, that rough cut happened once within 6 months according to operator of machine. We think that from result we have obtained from FFT, we can build a system which can indicate condition of cutting tool.

10.Future Work

As we know, research is wide growing area. We see a lot of researches are going on every kind of subject. There are room for improvements in our thesis also. There are some described as follows

- There are a lot of room to implement better filters. A lot of work can be done on that area. There are many kind of disturbance which may affect the process. Disturbance like human are talking near machine, there are particles are coming out as cutting tool is cutting, sometime machine can change sound due to some problem. So, filter out important data is really hard and need more research.
- We can use case based reasoning more in this thesis. There are a lot of room for better implementing case based reasoning. Because in case based reasoning, we can predict a situation which has happened before. We can save many solutions in the database. Now, as when there is some problem in current process then we can check previous solutions and if we can find similar situation then we can implement.
- In our application, we can raise an alarm when condition of cutting tool is worn out. It will raise an alarm and operator can check immediately and can change the cutting tool.
- We can enhance our application by attaching a simple mobile SMS application. This application can send SMS to the operator number what is current condition of cutting tool. Now, operator can exactly know the condition and he can prepare himself.

Reference

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[14] http://en.wikipedia.org/wiki/Analogue_filter
[15] http://en.wikipedia.org/wiki/Digital_filter
[16] http://en.wikipedia.org/wiki/Linear_filter
[17] http://en.wikipedia.org/wiki/Non-linear_filter
[18] https://ccrma.stanford.edu/~jos/fp3/Time_Invariant_Filters.html
[19] http://en.wikipedia.org/wiki/Time-variant_system
[20] http://en.wikipedia.org/wiki/Active_filter
[21] http://en.wikipedia.org/wiki/Passive_component
[22] http://en.wikipedia.org/wiki/Low-pass_filter
[23] http://en.wikipedia.org/wiki/High-pass_filter
[24] http://en.wikipedia.org/wiki/Band-pass_filter
[25] http://en.wikipedia.org/wiki/Band-stop_filter
[26] http://en.wikipedia.org/wiki/All-pass_filter