

## Fast Fourier Transformation of Vibration Signals Using Qt Creator

**T.Vishali**

(M.Tech Embedded Systems),  
Gokaraju Rangaraju Institute of Engineering &  
Technology, Hyderabad.

**G.L.Sumalatha**

Assistant professor,  
Gokaraju Rangaraju Institute of Engineering &  
Technology, Hyderabad.

### Abstract:

One needs to do signal analysis for a variety of situations. We know that vibrations are the fluctuations of a mechanical system about an equilibrium position [1]. In some practical situations, it might be difficult to develop a mathematical model of the system and predict its vibration characteristics through an analytical study [2]. In such cases, we can use experimental methods to measure the vibration response of the system to a known input. This helps in identifying the system in terms of its mass, stiffness, and damping.

In this proposed project work, the data may need some sophisticated mathematical techniques like FFT. The Fast Fourier transform (FFT), which is an efficient adaptation of Fourier's work, is regularly used to analyze sound waves [4]. The Fast Fourier Transformation is the mainstay of vibration signal analysis as it is the most time saving computational algorithm [3]. The project uses a software tool 'Qt Creator'. FFT and Wave file format are been included in a program thereby getting the random vibration, FFT and vibration data in graphical form as the output.

In the vibration meter if there is any failure occurring, we can find out where the error has been occurring and can sort it by using this project [5]. Hence, a vibration measurement program helps to stop failures by detecting machine problems before any serious problems could occur, as it acts as a troubleshooter and would be used in preventive maintenance programs to help save time and money in eliminating critical downtime [6]. In this project strategy to find out the random vibrations is proposed.

### Keywords:

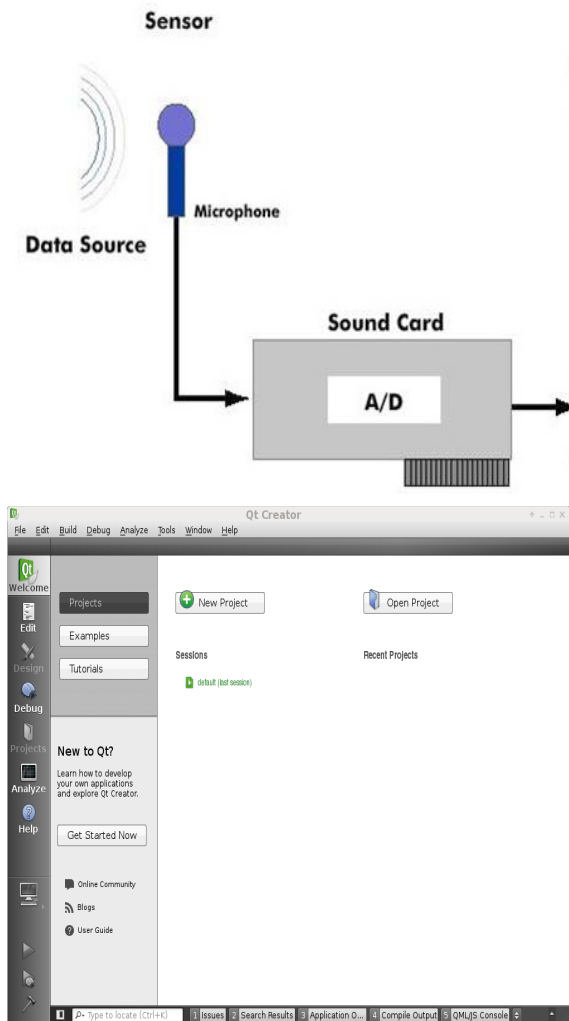
Qt Creator, Wave file format, FFT.

### Aim of the Project:

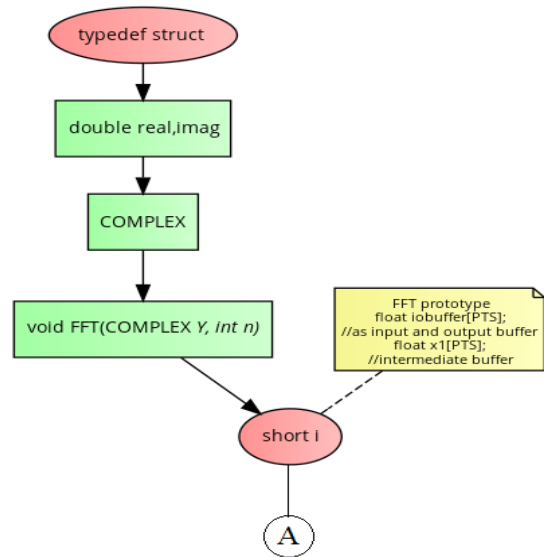
The main aim of this project is to measure the vibrations of any rotating machines. Many factors can cause excessive vibration in rotating machinery including worn out bearings and couplings, foundation damage, broken mounting bolts, or even misalignment and unbalance. The vibration meter is a vital instrument in troubleshooting machinery performance. We know that vibrations are the fluctuations of a mechanical system about an equilibrium position. For example, in a lawn mower if used for few years it would not produce noise, but eventually it may produce some sort of disturbance in the form of noise which not only cause irritation to humans but also create a huge destruction to the internal mechanical parts of a machine. By using the Qt Creator tool, which helps in getting the Graphical User Interface (GUI), a cross-platform C++ integrated development environment. The Fast Fourier transform (FFT), which is an efficient adaptation of Fourier's work, is regularly used to analyze sound waves. One needs to do signal analysis for a variety of situations. The Fast Fourier Transformation is the mainstay of vibration signal analysis as it is the most time saving algorithm.

### Design:

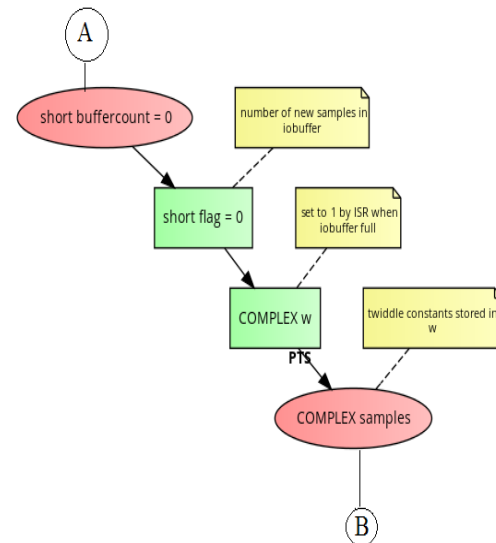
We will use the microphone and sound card, would like to store the resulting sound in windows using c/c++. Direct access to the microphone does not give you "frequency, pitch, all of that good stuff". First, frequency and pitch are the same. Secondly, they are found by processing microphone data. The raw microphone data consists of a sequence of periodically measured voltage samples. A "pure" sound would be a sine function, but of course there are always background noise and harmonics and measurement noise. The wave In Open function is where you start if you want low-level access to data from the microphone on Windows.



**Fig1: Interfacing of microphone to Qt creator**



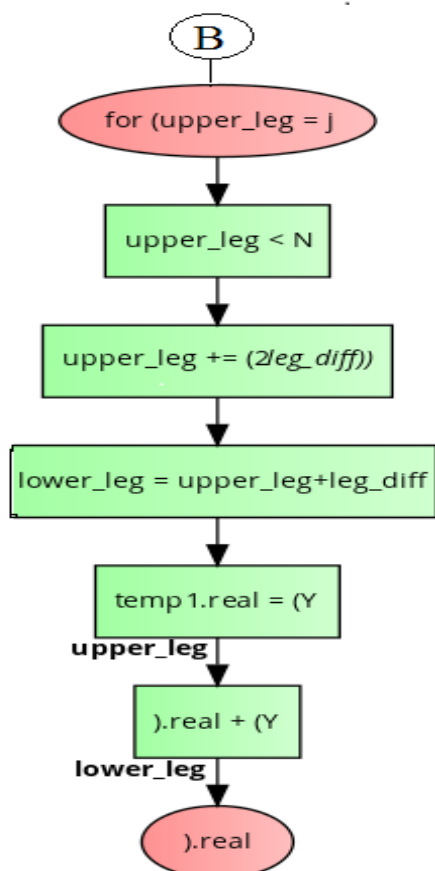
**Fig 2: Flowchart of complex variables**



**Fig 3: Flow chart of buffer count**

**Explanation of code:**

The complex variables are doubled and FFT values are been calculated as shown in fig2. The buffer count is kept to zero and the number of new samples are kept in input output buffer. It is set to '1' by ISR when input output buffer is full. The twiddle constants are now stored in 'w' as shown in fig3. Now the FFT is been calculated using the upper leg and lower legs as shown in fig4.



**Fig 4: Flow chart of FFT legs**

•A FFT is an algorithm to compute the DFT. A Fourier transform converts time (or space) to frequency and vice versa; an FFT rapidly computes such transformations by factorizing the DFT matrix into a product of sparse (mostly zero) factors. As a result, fast Fourier transforms are widely used for many applications in engineering, science, and mathematics.

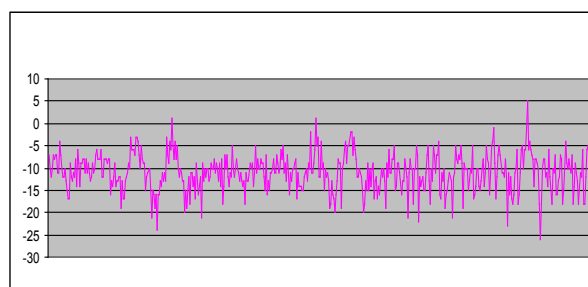
•The important and basic requirement is that the number of points to be transformed must be always 2n. To make exactly with the same number of samples points if needed can add zeros.

•An FFT operation involves complex numbers. In excel, such a number can be represented as -19.085527+.66697647i. The Wave data we are getting is –real number, just complex number with no imaginary parts. The computer sound card captures the data by 22050Hz sampling rate. Here we have got 512 sample points in the table shown in fig 2. By plotting the graph of derive points (FFT values) and Signal Amplitude column the resultant picture shown in fig 3. Actually, there are a lot of things to be borne in mind when doing

this, but this simple example does illustrates the method and shows its potential.

	A	B	C
1	No. of Points	Signal Amplitude	FFT Value
2	1	-7	-5587
3	2	-10	58.0685411624935-14.6546757537388i
4	3	-12	87.4745398130599+46.1567349717454i
5	4	-10	-89.7540525620199-94.0683692940344i
6	5	-7	-19.0833842674565-27.666927272647i
7	6	-8	5.22578277467738-14.8750114959963i
8	7	-7	59.6081188746813-51.6974442818348i
9	8	-7	20.4936266900632+18.3082291926865i
10	9	-11	15.2328818710643+5.21486961414661i
11	10	-11	98.6280029130323+94.9771805161527i
12	11	-4	-46.4795039593747+30.1050557217179i
13	12	-7	-46.1579378166159+71.0875699032463i
14	13	-10	-75.6880078365471+204.573139354099i
15	14	-12	293.634830107905-215.707172139929i
16	15	-12	99.3340629871662-141.613302979236i
17	16	-10	14.0697665129952-377.510842264678i
18	17	-13	-67.4876873117957+376.312327349873i
19	18	-15	-49.5443462472886+210.891465343749i
20	19	-17	-19.1216977768208+238.453698975552i
21	20	-17	47.9737363544092-363.192671547439i
22	21	-9	-37.1924499615108-56.5261201366755i

**Fig 5: Reconstructed signal values from sampled points using MS-EXCEL**



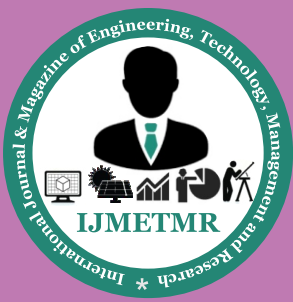
**Fig 6: Fast Fourier transformation using MS-EXCEL**

**Conclusion:**

Qt program provides quite a powerful environment for this type of experiment. We are in process of developing many other experiments using sound card. Being equipped with this knowledge and using the tools discussed in this application note can bring you more success with your individual application.

**References:**

[1] “Using Sound to Analyze Hardware Operation: A Progress Report”, Robert Karns Henry, East Tennessee State University.  
 [2] FFT Spectrum Analyzer Applet, United Kingdom.



[3] “Digital Data Acquisition and Analysis”, Department of Physics, Northeastern University .

[4] “Sound Analysis using FFT”, Dr. Mohammed Iqbal, Department of Physics, India.

[5] Ronald N. Brace well, “The Fourier Transform,” Scientific American, June 1989, pp. 86-95.

[6] “Mechanical Vibrations”, S. Graham Kelly, McGraw-Hill International Editions, Mechanical Engineering Series .

[7] The Zoom FFT by Bores Signal processing, Surrey.